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INTERNET TELEPHONY®

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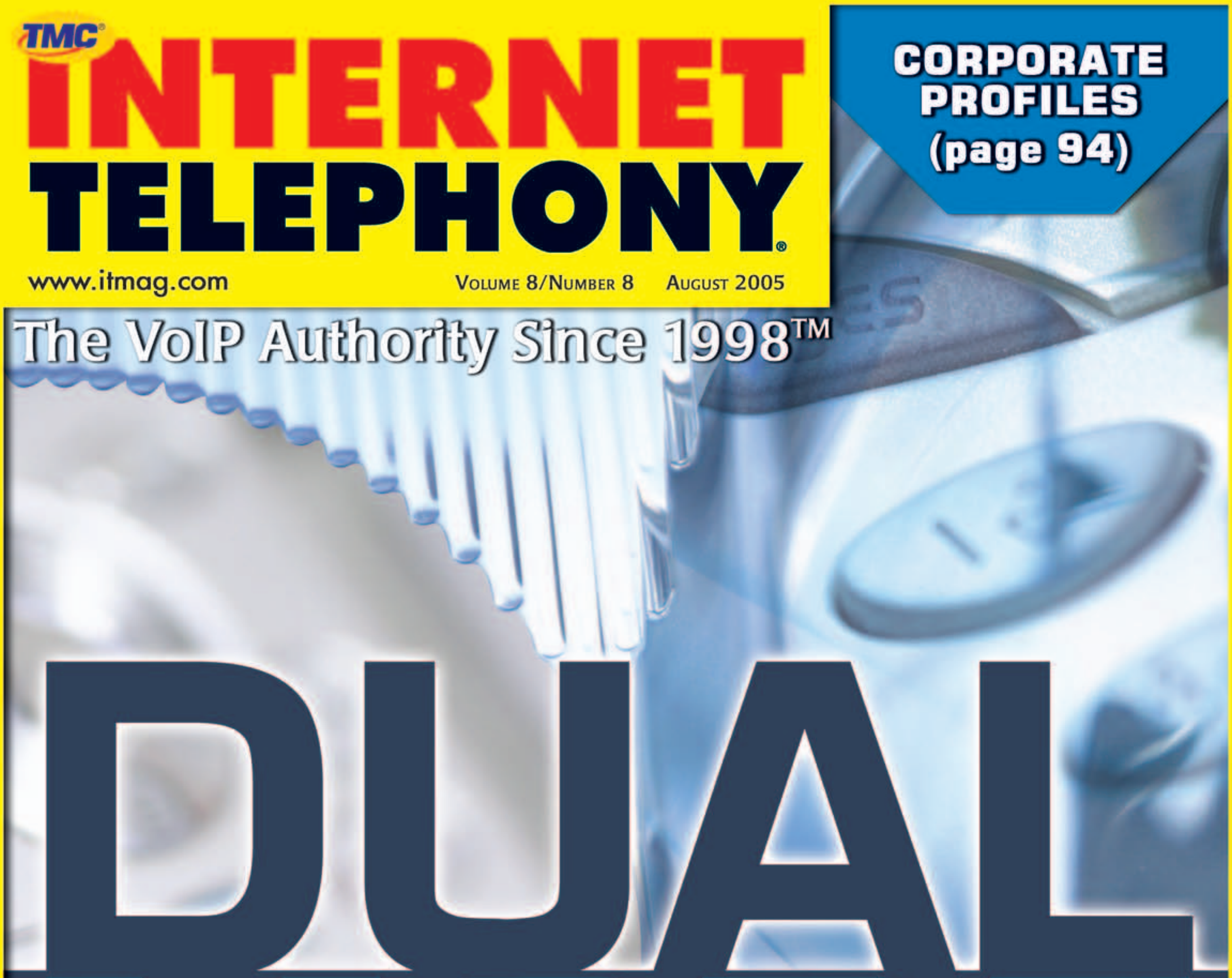
The VoIP Authority Since 1998™

CORPORATE PROFILES (page 94)



DUAL MODE

WiFi & Cellular Collide

Also In This Issue:

- PingTel CEO Bill Rich on the Future of VoIP
- CommuniTech's Neal Shact Speaks Out
- TMC Labs Innovation Awards: Part II

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

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The VoIP Authority

By Greg Galitzine

Who's The Best? You Tell Me!

It's not often we get the chance to let our readers get involved in the process of publishing this magazine, but now, for the first time ever, our readers can help us choose from among the hundreds of VoIP service providers, and help us decide who is the best at what they do.

INTERNET TELEPHONY® Magazine, in conjunction with TMCnet.com, is reaching out to you — our readers — to help us select the best of the best VoIP/IP Telephony Service Providers. This gives you the chance to decide who's making the grade, and grants you the opportunity to reward your favorite service providers!

We're asking you to nominate the best Service Providers in over 20 different categories, including:

- Best Overall
- Best New VoIP Provider (12 months on the market or less)
- Best Enhanced Services
- Best SMB Service
- Most Innovative, and more.

To nominate a company for this honor, we're asking you to submit your favorite Service Provider's name and Web site, along with the category you wish to nominate them in, and send it via e-mail to **SPAward@tmcnet.com**. Please include comments as to why you think your choice deserves to win in their particular category.

Also include your name and place of residence (City/State/Country).

Deadline for nominations is September 15. Go to http://www.tmcnet.com/awards/spa/ for more information, including a complete list of categories in which you can nominate your favorites.

The results will be announced at the annual Service Provider Awards dinner to be held in Los Angeles, CA on October 26th, 2005. We will publish the award winners in a future issue of INTERNET TELEPHONY magazine and online at http://www.tmcnet.com after the winners are announced.

Be a part of the first-ever reader-nominated service provider awards. Don't miss your chance to determine who wins this coveted prize!

I'm curious also to see how some of the better known service providers fare in our contest, especially in light of the recent news from Keynote Systems and their first-of-its-kind VoIP quality study. Vonage (news - alert) emerged as the most reliable provider and AT&T's CallVantage service claimed top honors insofar as voice quality was concerned.

More information regarding the report can be found at the Keynote Systems Web site, http://www.keynote.com.

-Greg Galitzine, ggalitzine@tmcnet.com

INTERNET TELEPHONY®
Volume 8/ Number 8 August 2005

Contents

90

IN EACH ISSUE

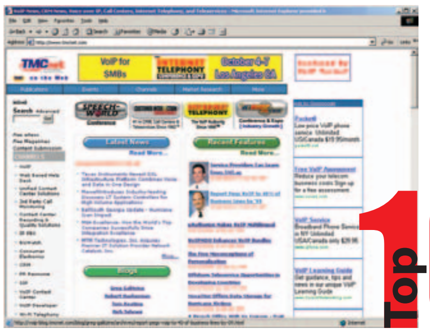
COLUMNS

DEPARTMENTS

FEATURE ARTICLES

Go To Ad Index

Contents



Top 10

Top 10 Visitors to TMCnet.com's Blog Pages (Most Active, By Country)

1.	United States	6.	Germany
2.	Australia	7.	France
3.	Canada	8.	Japan
4.	United Kingdom	9.	Singapore
5.	Netherlands	10.	Taiwan

QUOTE OF THE MONTH:

"It's a pretty safe bet that dual-mode handsets will proliferate in both enterprise and consumer markets. A few details still need to be worked out for dual-mode phones since WiFi telephony is still in its adolescence. Making it all work together — from both a technical and market perspective — will require a lot of cooperation between network operators, telecom infrastructure providers and handset developers. But both WiFi and cellular are here to stay…"

– Ben Guderian (page 54)

WHAT'S ON TMCNET.COM RIGHT NOW

To stay current and to keep up-to-date with all that's happening in the fast-paced world of IP telephony, just point your browser to http://www.tmcnet.com for all the latest news and analysis. With over 3.9 million unique page visits per month, translating into nearly half a million unique visitors, TMCnet.com is where you need to be if you want to know what's happening in VoIP.

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

Security Co. Says It Found Flaws in Cisco's VoIP Infrastructure
Internet Security Systems (ISS) claims to have a patch for critical security flaws in VoIP technology offered by Cisco.
http://tmcnet.com/137.1

Global IP Sound Signs OEM Deal With MSFT
Global IP Sound announces an OEM license agreement with Microsoft.
http://tmcnet.com/138.1

VoIP Phones: A Look At Newly Released Devices
A number of technology providers have announced new VoIP-based devices.
http://tmcnet.com/139.1

Skype To Go: Boingo Powers VoIP Hot Spots
Skype and WiFi aggregator Boingo Wireless announced they teamed up to develop Skype Zones, a set of 18,000 hot spots.
http://tmcnet.com/140.1

France Telecom, Cisco Get Airbus IP Deal
France Telecom and Cisco will install an IP communications system for Airbus.
http://tmcnet.com/141.1

TMC's IP PBX Channel
The IP-PBX Channel on TMCnet.com features the latest news and original bylined articles on IP-PBX. To visit TMCnet.com's IP PBX channel, just point your browser to http://www.tmcnet.com/channels/ip-pbx/. Sponsored by Sphere Communications Inc.

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

By Rich Tehrani

Making Sense Of Broadband

Recently the Supreme Court made a ruling on a case involving Brand X Internet that will have major implications for the future of broadband in the U.S. If you aren't aware, the U.S. lags behind many other countries in broadband adoption and the last research data I saw put us somewhere around 15th in the world.

If you are reading this magazine it is safe to say you understand how important broadband access is. I know I do. If you think about it for a while you realize that recent uptake in broadband connections has been a driver for e-commerce and VoIP (define - news - alert). Dig deeper and you realize broadband is also a driver for companies like Google specializing in search. Keep digging and you realize the proliferation of blogs are in part due to increased broadband penetration and this in turn has enhanced free speech and reduced the plain vanilla news stranglehold a few large media companies have in the U.S.

Think about craigslist, a site that has democratized entire industries. There is Monster for jobs, iTunes for music, etc.

The point is broadband is really changing the way we live work and play in the U.S. and obviously everywhere else in the world. Waves of companies are looking to sites like eBay as a place to sell directly to consumers without the need for a middleman such as a reseller or distributor.

You could probably fill this magazine with industries that are being changed for the better due to broadband proliferation.

President Bush understands this and in April of 2004 he made a speech at the American Association of Community Colleges (http://tmcnet.com/142.1) where he said, "And I want to talk about affordable broadband technology so that America can stay on the leading edge of technological change." On its face, this statement is magical. Not as magical as the latest Harry Potter book mind you but for those of us who rely on broadband technology to make a living, it is exactly what we want our president to say.

This gets us back to Brand X, a company that fought to have access to cable lines to provide Internet service to its customers. The Supreme Court ruled against Brand-X and in so doing backed up the FCC's decision.

Cable companies have used this monopoly position to invest into their broadband networks. Is this fair?

If you think about it, why should the cable companies share anything with anyone? After all the government doesn't have the right to say that TMC has to share its magazine pages with competitors.

The difference is that cable companies enjoy government sanctioned monopoly status in the U.S. There isn't competition for cable service. I either purchase it or I don't. I can't go to a cheaper cable company to get cable service if I would like. Sure there is satellite you might argue but it isn't as good as cable for a variety of reasons.

Cable companies have used this monopoly position to invest into their broadband networks. Is this fair?

It seems obvious that if cable companies were forced to share their lines, consumers will get access to cheaper broadband service. I am unclear of any scenario where decreased choice is good for consumers.

What's happing behind the scenes is that cable companies and ILECs are upgrading their networks and they tell the government that if they are forced to share, they won't upgrade anymore. The government is buying this argument hook, line, and sinker.

While I understand this position, these companies owe us. They owe the government, they owe consumers, and they owe the community… They owe everyone. Having no competition for so long means they were able to gain many advantages in the market. Now they are using these advantages to influence politicians, run TV ads, and do other things to protect their existing monopolies and grow them into other areas such as broadband.

The government has an obligation and of course has tried to increase telecom competition. The Telecom Act which is now almost a decade old didn't work as planned so they have decided that having just the cable and phone companies competing is adequate with the hope that power line and WiMAX (or some variant of broadband wireless) will add extra competition.

But what about the small ISP? Small companies are the backbone of the U.S. and some of the really good ones become big companies. Recently, TMCnet's Ted Glanzer wrote an article about the Brand X case (tmcnet.com/143.1) and I thought there are some key excerpts worth sharing.

"It's a terrible decision," Brand X owner Jim Pickrell told TMCnet. "It's bad for consumers and it is bad business . . . [Brand X is] effectively locked out . . . It's an end to competition in broadband and telephone . . . For us it's a disaster."

Pickrell added that the U.S. will fall further behind other countries in providing broadband access to its citizens.

"We're falling behind," said a frustrated Pickrell, noting that the cost of broadband in Japan is half of what it costs us in the United States. Not surprisingly, Pickrell said, a greater percentage of Japanese citizens have broadband access than those in the U.S.

At the other end of the spectrum are Senator John Ensign (R-Nev.) and Carol Mattey, a former FCC deputy chief of the

Wireline and Competition Bureau.

"It is my hope that Congress can build on the Supreme Court's decision today on Brand X by updating our nation's communications laws," Ensign, who is reportedly redrafting the Telecommunications Act of 1996, said in a statement. "It is time that these laws reflect the sweeping changes technology has brought to this critical sector of our economy. Revising the communications laws will remove barriers to innovation. Consumers will benefit from exciting new technologies that get to the marketplace faster and at lower prices."

Many in the government are applauding this decision but the net effect is that this leaves us with a grand total of two broadband choices today. This is not enough for real competition. This leaves no room for new upstarts who want to gain share by lowering prices or those companies that want to differentiate based on service. I also reiterate that it is unfair to allow entrenched monopolies to grow into new monopolies or duopolies keeping the strong corporate identities they were able to build via their monopolistic position in the market.

As I write this, TMCnet is receiving a staggering amount of traffic to its Web site. We have ordered a much faster connection but have to wait for many weeks before we get it. Perhaps the story is different in Manhattan but here in Norwalk, CT, you can wait months for the phone company to move. I had an installer tell me that he will be on vacation for a week and he needs to string fiber for about 250 feet, which he pointed out to me was a very far distance. Due to one person's vacation schedule and a few hundred feet of geography we need to wait months for service from our LEC. Bear in mind TMC is a paying customer. What other non-monopoly business makes you wait so long to buy something from them? The point I am making is that entrenched service providers in general are a laughing stock when it comes to customer service.

Two competitors does not a competitive market make. Allowing two industries with monopoly positions and terrible service levels to continue their monopolies makes little sense.

Perhaps I am an idealist but having less competition doesn't ever seem like a great way to build competition. Politicians are waxing poetic about how now the market needs to be deregulated resulting in more competition. I just don't get it. It's like an episode of Twilight Zone to me. The Supreme Courts says that cable companies don't have to share their lines with competitors — effectively putting them out of business — and politicians put out press releases telling us it is a great day for telecom competition. I am not sure what to make of it.

Is the issue cut and dry? No — it never is. In fact to make this whole debate more interesting, Cablevision Is slowly rolling out 100 Mbps service in and around the New York area. I am a Cablevision customer and shareholder so this prospect is extremely exciting to me. Furthermore, this is the Internet speed that some Asian countries provide routinely. I have been arguing for a while that we need service providers to give us the same speeds as other countries. Now, at least Cablevision is.

So I am torn by this decision. Half of me thinks it is terrible but the other half sees a potentially bright future if we start to see routine implementations of 100 Mbps connections to the home and office.

Perhaps President Bush was planning to kill the small ISPs all along. In the speech I mentioned above, Bush says, "The

Dear Mr. VC

To be more politically correct: Dear Mr. or Mrs. VC

Many of my readers are in the VoIP community — you know who you are… You were doing VoIP in the late nineties — when it wasn't fashionable and you used to call it IP telephony or Internet telephony. I am running into more and more of these people and they often ask me if this time the IP revolution is for real. The answer is absolutely.

As president of a leading publishing and trade show company having been involved in VoIP since its inception and having launched the first magazine in the space, I think I have a unique perspective on the market. I am seeing more optimism and sales than at any time before. I saw purchasing in 1999–2000 but many of those sales were to CLECs that were new, venture backed, and lacking long-term business plans.

So why is it real this time? Simply stated, because this stuff works and works well and most importantly perhaps, the stigma of VoIP being bleeding edge has worn off and has been replaced by the stigma that anything that isn't VoIP is legacy. That is a powerful argument for the success of any industry — the competition to VoIP is basically outdated.

Barring geopolitical events beyond our control it would seem that the industry is healthier and has a brighter future than ever. The level of optimism I am seeing is unmatched and is exceeding what I saw in 2000.

There are a number of companies looking to take advantage of the VoIP space and many of them are still not funded well and don't market. These companies are going to have a virtually impossible time making it. If you don't have funding to market properly, you better come up with a business plan like Skype or hope that someone swoops in to purchase you.

The fact is the PR and marketing machines in virtually all companies are running on all cylinders or at least most cylinders. The landscape is becoming much more competitive.

I have met a few companies where they tell me they need to sell some products before their investors will give them money for marketing. I hope these investors are reading. Do you really think anyone wants to buy products or services from companies that they haven't heard of? If the product is under $100 maybe, but if you are an investor in a company that produces VoIP equipment costing thousands of dollars — no one will buy these products if they haven't heard of the company. Would you put your job on the line, buying from a company that can't afford to tell the market what they do? Of course not, and neither will anyone else.

I suggest you cut your losses and see if you can find a company with a Skype like model to get involved with.

In my experience — the companies with the best technology hardly ever win any competitive races. It is the companies that understand PR, marketing, and messaging that win. We are coming out of a time when there were more telecom company bankruptcies than in any period in history. Marketing (PR, trade show exhibits, advertising, direct mail, etc) shows strength and stability. Your potential customers will not buy from you if they are not sure you will be around.

In today's ultra-competitive VoIP market you cannot be heard, seen,or gain mindshare if you don't spend. As a rule you should be spending no less that 20 percent of what you spend on R&D on outward marketing. A company that wants to grow more quickly needs to spend 30–40 percent or more. Technology can be built by anyone and the features and functions you think are so novel today will be duplicated

Go To Table of Contents | Go To Ad Index

third goal is to make sure that we have access to the information that is transforming our economy through broadband technology. I'm talking about broadband technology in every part of our country. I was the governor of Texas for a while. I remember talking about access to information and there was always a group of people saying, that's fine, big cities get it but rural people don't. I'm talking about broadband technology to every corner of our country by the year 2007 with competition shortly thereafter."

This last statement seems unusual. Why mention competition unless you are going to make sure to squash it until 2007. It would seem the Brand X case never had a chance based on this statement.

Is our president banking on power line and wireless broadband to be the competition? Does his crystal ball see these technologies being major forces in 2007? One thing's for sure… There is no more exciting time to be in this market as WiMAX and broadband over power line providers come on line. But I have to worry about VoIP. Somehow entrenched providers have found ways to stifle competition and use the political system to their advantage. How long will it take them to come up with a legal way to do the same to VoIP providers? What about video providers (it is a matter of time before a slew of competitive IPTV providers emerge). Stay tuned.

Video Over IP

I received a call from Kevin Shahbazi recently. Kevin used to run a company called Trust Digital that was involved in encrypting information on handheld computers so the data couldn't be used by unauthorized people. He now heads eView Technologies, a company that analyzes and mines real-time video streams. This is important for a few reasons, the first of which is that using this technology you may have been able to foil the London bombing plot. Cameras are great to have but it is impossible for a human to monitor multiple cameras at once and be alert for long periods of time. Technology needs to assist us in these endeavors. The second area is looking at shopping and buying behaviors of customers in retail environments.

There has been a great deal of interest in our Security over IP Summit at Internet Telephony Conference and EXPO this October in Los Angeles and I think companies like eView Technologies will benefit greatly from the interest and growth in this market.

Two competitors does not a competitive market make.

SIP Under Attack

I recently caught up with Rich Medoza who now works for BorderWare in the SIP (define - news - alert) Security division. Rich used to work for Level 3 and as such has great experience in the product line that his new company produces, namely SIP-aware firewalls. Rich tells me that typical service providers are better at transporting packets than blocking them and this is just one of the reasons that session border controllers do well.

But this can be a problem with SIP messages as they can have nasty payloads that do bad things to your network. They could contain SPIT otherwise known as spam over Internet protocol. Using their SIPassure product they can set up the network to proxy the firewall before it hits the softswitch. If the firewall sees 50 messages from one user inside of a minute it knows the user is a spammer and can block subsequent messages.

Currently there aren't too many SIP trunks coming from service providers but that will change soon. In the mean time, SIPassures's products can also protect companies from internal theft of service. In fact Rich cited an example of a company whose softswitch was being used after hours to host a call center without the company's knowledge. The products can also be used to prevent internal denial of service shutdowns, buffer overflows, SIP invite request attacks, and more. You should seriously consider a SIP firewall before deploying VoIP on your network

VoIP Quality Survey

Recently TMCnet teamed with Keynote Systems to sponsor a webinar on VoIP quality. What is exciting to me is that we broke news during the webinar as they announced their key findings to TMCnet's audience. The key points of the study are that VoIP quality still lags behind the PSTN while Vonage and AT&T CallVantage provide the best service. The details are available at (tmcnet.com/144.1).

One Reseller, Many Generations

While INTERNET TELEPHONY magazine often writes about products and the companies that produce them we don't often write about the companies that help get great technology into the hands of the end user. One such company, GBH Communication (formerly GBH Distributing) has been around for well over a decade selling and distributing telecom products. I knew them in 1990 as a major Plantronics reseller and they later they became the first Polycom reseller.

The company has evolved into a full service multimedia reseller and the company's charismatic founder Von Bedikian, who I've known for a decade and a half, tells me he coined a new phrase to represent the products they sell. The term is CommIP and it's short for communications over IP and it's definition is VoIP, video, and data collaboration over IP. GBH resells products from lots of companies today from Masergy, Sphere, Radvision, Interactive Intelligence, and Vonexus. I

tomorrow — and you don't have enough money to sue all the people that may infringe on your patents.

A company is about it's brand, not about it's products. Sure, sometimes a product becomes stronger than a brand (iPod, VW Bug, etc) but there is still branding that must be done in any and every industry. VoIP is growing faster than virtually all other tech markets and real profits are being made. Everyone wants part of this pie and it takes branding and imaging and an overall marketing strategy to claim a piece of it. If you enter a marathon and don't have enough stored up energy for the race, you don't finish. Dear, Mr./Mrs. VC, are you in the race to win or watch your runner die off at the half-way point? IT

Go To Table of Contents | Go To Ad Index

asked Von how big the company is these days and he responded, "We have 70+ people and 40 million in revenue, going on 400 million." Now that's confidence.

SIP Speed Boat

SIPquest (news - alert) is a company that seems to be popping up more and more in the industry as an enabler of a number of technologies from WiFi telephony to advanced mobility applications. Dave Hattey, who used to work at 3Com recently decided to take a leadership role at SIPquest as President, Chief Executive Officer, and member of the Board. So I decided it would be a good opportunity to hear from Dave firsthand why he made the switch.

Dave told me the reason he decided to go to SIPquest was that he saw fun, innovation, and vision as well as a unique relationship with Columbia University giving them the rights to the multimedia applications created at the prestigious university.

Dave tells me that managing a smaller company is great because he can reach out to the entire team at once and act more quickly. He says he feels like he is a speed boat. A large company he said can be like a battleship. Where does he see the future of VoIP going? He says SIP is the VoIP solution. I asked him about the growing industry moaning about large vendors adding so many proprietary extensions to SIP that interoperability suffers. Hattey feels that the market is powerful enough to drive us to the end goal of interoperability. I hope he is right. IT

VeMail Instead Of E-Mail

A while back I switched to Verizon Wireless from T-Mobile as I often had dropped calls on T-Mobile's GSM network. One thing I missed from being on GSM was the great devices the GSM networks supported such as SonyEricsson phones that have e-mail clients built into them. I often used the voice recording function of the phone to then e-mail a .WAV file to others. It was such a great productivity booster.

Alas, when I switched to Verizon I ended up with a phone that supports Bluetooth but is so handicapped in features and has such a terrible user interface that I feel deprived when I use it. Not to point fingers but it is the Motorola V710.

I also miss the simple e-mail client that at least let me keep up with what was happening in my inbox — even though I never really typed long messages via my phone.

Enter VeMail, a service provided by VoiceGenesis that is integrated with Verizon Wireless devices as well as a number of other carriers such as Alltel, Cellular One, and a bunch of smaller providers. I found the company by accident and am surprised that Verizon Wireless keeps this service a secret.

Why you ask? Because it is very useful as you get all the benefits of having access to your e-mail on a mobile phone without the need to type using a mobile phone. The way this works is you record your message and send it as an attachment via e-mail. Unlike the .WAV messages I used to send via my SonyEricsson phone which sometimes could not be opened, the company has employed a clever workaround.

The solution is simple — in the e-mail you receive you get a link to an audio file that streams and you get a phone number with an ID so you can call in and get your messages. This latter addition is perfect for computers or devices that don't have speakers or can't stream.

Other benefits of the service are the ability to upload up to 10,000 contact names into the service and have them accessible via your mobile phone. This number is dependent on the model of device you have.

In order for the service to work, you need to download what Mark Marriott, the CEO of VoiceGenesis jokes is a "fat client" which is really just a 325K file. One of the benefits of this file is that it helps avoid the pitfalls of WAP-based e-mail where you need to have the server download every screen to your cell phone and you need to wait for this download to happen. Once you have the "fat client" on your phone, you no longer have to wait for screens to redraw as many of them are in the phone's memory. You can see 50 e-mails per screen and up to 1,000 messages are cached on your phone (that's a lot of Viagra and mortgage messages) at a time. There is currently no support for other folders besides the Inbox but it seems like the company is considering adding this functionality down the road.

The service costs $5 per month plus air time. Some service providers will charge you a separate data rate while others will just use plain old minutes from your plan. Some upgrades we can look forward to are the ability to make calls from the address book stored in VeMail's database as well as getting attachments downloaded to your device.

The downside to this service is that you can't use voice messaging for large amounts of e-mail. At least most users would probably be annoyed if you kept sending them voice responses to e-mail. This is unless of course they work for you. One day soon when speech recognition gets good enough, this solution will get even better. The other drawback is that speech is a great interface for many situations but doesn't work too well on a train or in a crowded location where you need privacy. For these situations, typing is still preferred and the ultimate killer device would probably allow a blend of voice e-mail with recognition and typing, depending on the situation, privacy level needed and your mood. IT

Industry NEWS

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FAST FACT

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Arel Launches Spotlight II Conferencing Solution

Arel Communications (news - alert) and Software recently launched the newest version of the company's Arel Spotlight conferencing and collaboration solution. Arel Spotlight II is a Web-based conferencing and collaboration solution combining voice, video and data collaboration in a single software solution that is designed to enhance communications, facilitate workgroup collaboration, and drive productivity in the enterprise. Arel Spotlight II offers key features including new quality of service (QoS) mechanisms, dynamic video bandwidth and interoperability with heterogeneous networks and endpoints (SIP and H.323 end-points, multipoint conferencing units, and PSTN and satellite networks).

Arel has implemented a new QoS system that manages the bandwidth utilization of the Spotlight II in order to accommodate more complex application sharing while enabling simultaneous audio and videoconferencing at lower bandwidth thresholds than ever before. The QoS feature enables Spotlight session participants to collaborate on larger, heavier files in more sophisticated applications such as product life-cycle management (PLM) applications that require significant resources; all this with enhanced application response times.

Arel Spotlight's dynamic video bandwidth feature enables participants of varying connection qualities to always receive the maximum quality available on their network without compromising the video quality of others. This capability ensures that broadband users leverage the full bandwidth capacity of their network to receive the highest quality videoconferencing experience possible even if low-bandwidth users participate in the same conference.

Although Arel Spotlight II traditionally requires a PC, Web cam, and an Internet connection to participate in a conference, the software's flexible architecture allows participants to utilize a variety of other endpoints and devices to connect to a Spotlight session. Users can connect to an Arel Spotlight session through a telephone, mobile phone, IP phone, PDA, and legacy videoconferencing system over a hybrid of networks including IP, PSTN, H.323, SIP, and Satellite.

http://www.arelcom.com

Quintum, Unified Communications In SS7 Deal

By Johanne Torres

VoIP (define - news - alert) technology provider Quintum Technologies (news - alert) has reached an agreement with Unified Communications, (news - alert) a telecom signaling provider, for the companies to jointly support Signaling System 7 (SS7) on Quintum Tenor Carrier MultiPath Switch (CMS) and DX product lines.

The agreement between both companies is designed to allow the SS7 Signaling Gateway to support multiple Quintum Tenor VoIP switches over the IP network, enabling service providers to support SS7 in their VoIP networks. Quintum's Tenor DX and CMS products provide the functionality required to support applications such as Wholesale VoIP Termination, Tandem Switching, IP Local Access Services and Calling Card Services, H.323, SIP, intelligent call routing, TDM switching and QoS. The Tenor DX supports up to 4 T1/E1/PRI trunks, whilst the Tenor CMS supports up to 32 T1/E1/PRI trunks.

"We have been working on SS7 signaling for more than six years and have a strong base of technical competency in this area. Our SS7 Signaling Gateway is a flexible, high-performance, and cost-effective solution that can work seamlessly with Quintum's Tenor CMS and DX product lines without any change in hardware. As part of our agreement with Quintum, we will be introducing the SS7 Signaling Gateway to Quintum's global network of resellers and distributors, who will be able to integrate the products to provide a complete VoIP solution that supports SS7 to their customers. This deal not only strengthens our existing relationship with Quintum, but will also boost our existing distribution network," said Wong Tze Leng, CEO of Unified Communications.

"Recognizing the growing popularity of SS7 signaling in the modern international traffic market, this collaboration with Unified Communications will allow us to compete well in the market for VoIP solutions that require SS7 signaling. Together with our own Analog/R2/ISDN offering, we will have most of, if not all, major signaling protocols used in today's telephone network to offer our customers," said Mr. Cheng Chen, CEO of Eatontown, NJ-based Quintum.

Unified Communications' SS7 Signaling Gateway is available now via selected Quintum certified channel partners.

http://www.quintum.com
http://www.unifiedcomms.com

TalkSwitch, Mediatrix Team On VoIP For SMB

Centrepoint Technologies (news - alert) had confirmed SIP interoperability between the TalkSwitch (news - alert) telephone system and Mediatrix 1102, 1104, and 2102-20 gateways in November 2004. Since that time, hundreds of units have been deployed and customer reports indicate they are delighted with the joint solution. As a result, Centrepoint has announced it is adding the 2102-10 and 2102-21 gateways to the suite of Mediatrix products it carries, and has made the solution even more affordable by reducing prices on all Mediatrix products available from Centrepoint.

"Interoperability with industry leaders like Mediatrix (news - alert) allows us to offer the first truly cost-effective, complete solution for companies looking to use VoIP to tie in all their branch offices and teleworkers," said Jan Scheeren, President and CEO, Centrepoint Technologies

http://www.talkswitch.com
http://www.mediatrix.com

Inter-Tel Launches Unified Communicator 3.0

Inter-Tel, Inc., (news - alert) has announced the release of Unified Communicator 3.0, a software application designed to deliver powerful presence management and collaboration tools to users throughout an enterprise. Unified Communicator 3.0 was developed to allow team members to control and prioritize how, when, where, and from whom they receive calls, as well as remotely videoconference with colleagues, share files and documents, deliver presentations, co-browse the Internet, and participate in online chat sessions.

Unified Communicator 3.0 joins two recently released applications, Inter-Tel Web Conferencing and Inter-Tel Remote Support, to form Inter-Tel eCollaborations Solutions, a suite of IP-enabled presence management, mobility and collaboration applications, designed to empower users to streamline and enhance business communications.

"Today's evolving business climate is causing companies to seek new tools that will empower their staff members to increase their productivity, whether they are working in their office, at their home, customer site, an airport terminal, or virtually any other location," said Craig W. Rauchle, president and chief operating officer for Inter-Tel.

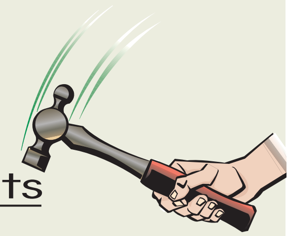
http://www.inter-tel.com

Fonality Releases New Phone System For Small Businesses

Fonality (news - alert) recently announced PBXtra, an IP-PBX designed to deliver enterprise-class capabilities to small businesses for up to 80 percent less than the cost of traditional PBX systems. PBXtra combines the powerful call handling and advanced features companies expect from an enterprise-class PBX system at an affordable price. Configurable from anywhere on the Web, PBXtra is easy to set up and administer, and eliminates the need for specialized phone support.

PBXtra runs on standard PC hardware and routers and uses layers of open-source Linux and Asterisk technology to provide the least expensive option for small businesses deploying new IP-PBX phone systems. Plus, PBXtra includes new features not found in any PBX solution for such an affordable price including: click and call from your PC, four-digit dialing, easy telecommuting, Microsoft Outlook integration, CRM and Web browser integration, MP3-based music on hold, unlimited extensions, unlimited voicemail, hybrid VoIP/PSTN, support for IP and analog phones, and automatic desktop notification of callers. Companies that require call center functionality, such as agent and queue support, can use PBXtra's Call Center Edition.

http://www.fonality.com



Quick Hits

Zultys Announces 24-Port PoE Switch

Zultys Technologies (news - alert) has announced the release of the EPS24. This switch has 24 10/100 Ethernet ports and a 1Gb/s fiber port. It provides industry standard power over Ethernet (PoE) on all copper ports, and fully embraces open standards including IEEE 802.3af.

The EPS24 can be used to power devices such as IP telephones, keyless entry systems, security cameras, and WiFi access points.

http://www.zultys.com

Siemens Deploys Into Financial Co-Ops

Siemens (news - alert) announced today that FIDUCIA IT AG, an IT service provider for the Volksbanken and Raiffeisenbanken banking cooperatives in Germany, has chosen the Siemens HiPath 8000 Real-Time IP system as its telecommunications platform. The SIP-based HiPath 8000 solution will become the central softswitch in FIDUCIA IT's network and act as a hosted, or overlay network, for the company's customers in the banking industry.

http://www.enterprise.usa.siemens.com

Ingate Interoperable With Avaya

Ingate Systems (news - alert) announced that its Firewalls, SIParators, and application modules for remote connectivity and VoIP survival are compliant with key IP telephony solutions from Avaya. All Ingate products have been compliance tested by Avaya for compatibility with the Avaya Converged Communication Server, an application that enables SIP services, and with the Avaya (news - alert) Communication Manager software.

http://www.ingate.com
http://www.avaya.com

Go To Table of Contents | Go To Ad Index

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VoIP Performance Still Doesn't Match PSTN

While cost savings associated with VoIP are undeniable, the industry still has a lot of room for improvement, a new study suggests. From an end-user's perspective, service quality is still lacking in the areas of reliability and audio quality.

By Robert Liu

No matter how you slice it, Voice over IP (VoIP) will eat up more and more of the pie that is the telephone market. For enterprise-class customer to the consumer, the cost savings are undeniable. But if you've ever thought VoIP performance still doesn't match the standards set by the century-old public switched telephone network (PSTN), a pertinent new study confirms that is no misconception.

In fact, service providers will need to address the issues of reliability and audio quality if companies like Vonage or Cablevision hope consumer will buy into the "cost-savings" proposition. Those are the findings of the "VoIP Competitive Intelligence Study," an independent survey examining end-user perceptions of service quality conducted by Keynote Systems, the San Mateo, Calif.-based performance measurement specialists.

And while the findings aren't particularly surprising given the infancy of the burgeoning VoIP industry, analysts agree the quantitative data is significant because it provides validation. "What I get out of it is the third-party verification," said Will Stofega, analyst at IDC's VoIP Services practice.

To be sure, even executives at Keynote don't believe the results will likely hinder VoIP acceptance in the consumer market. As such, analysts like Stofega haven't revised downward their growth forecast for VoIP consumer deployments. The IDC analyst still projects a compound annual growth rate of 94 percent with the consumer VoIP market reaching 27 million by 2009.

"My personal opinion is that the lower quality of VoIP communication relative to PSTN is considerable; but, given how aggressive most VoIP providers have been in the past, this gap will slowly shrink. Effectively, this wide gap should shrink relatively quickly and should not inhibit overall market growth for VoIP providers," said Dharmesh Thakker, Senior Product Manager for Service Level Management Solutions at Keynote.

Still, the valuable insight provided by the Keynote study serves as an extremely useful benchmarking tool for service providers to employ. "It's a reasonable first start," agreed Eric H. Siegel, senior analyst at Burton Group. "The actual way they are doing the measurement seems to be quite good. And the data seems quite reasonable."

The five-week study was conducted from May 21 to June 25 between New York and San Francisco using six of the leading U.S. service providers: Vonage (both standard & softphone), AT&T CallVantage (news - alert), Lingo (news - alert), Packet 8 (news - alert), Verizon VoiceWing (news - alert) and Skype (news - alert). Noticeably absent were cable telephony offerings like Cablevision's Optimum Voice; however, Comcast or Time Warner Cable were among network carriers used to sample the voice packet transmissions. In addition to the cable networks, each of the seven service providers (remember, there were two from Vonage) was also tested on T-1 business class network carriers (AT&T, Sprint and UUNET) and the incumbent DSL carriers (SBC or Verizon), creating a possible 35 different combinations.

Only VoIP-to-PSTN and PSTN-to-PSTN (as a baseline) calls were tested; no VoIP-to-VoIP calls were made during the data collection period. A total of 163,000 calls were placed every 30 minutes by an automated technology agent that transmitted a test audio that was subsequently returned and then compared back to the fidelity of the original file. The clarity was measured based on any amount of delay as well as the mean opinion score (MOS) as calculated by the ITU Standard P.862, a.k.a. Perceptual Evaluation of Speech Quality (PESQ). Reliability was based on the average number of dial attempts and dropped calls. Here's what Keynote found:

"When you average it out, the industry on a whole has a lot to catch up on,"

Go To Table of Contents I Go To Ad Index

Thakker told INTERNET TELEPHONY magazine.

No one VoIP service provider or network carrier dominated in every key performance metric. On a whole, VoIP service availability scored only 96.9 percent, compared with the 99.999 percent of PSTN. Audio clarity of the service providers was also below PSTN (define - news - alert) levels. What's worst the amount of audio delay of VoIP calls approached the threshold where most users become dissatisfied.

Vonage led its peers in reliability while AT&T CallVantage won for Best Audio Clarity.

Keynote noted there was a noticeable gap between the leading companies and the laggards in both reliability and audio clarity but declined to disclose the complete league tables generated by the study.

As for the network carriers, Time Warner Cable was the Most Reliable while UUNET and Time Warner Cable were tied for Best Audio Clarity. Generally, DSL proved more reliable than business class or cable carriers but also created the biggest audio delays. There was little variation across the board for overall audio clarity (delay and MOS score).

"Everyone needs to catch up compared with PSTN," Thakker concluded. "The average VoIP performance is still not up to the standards of PSTN based on reliability and quality."

Even though the VoIP offerings of Comcast and Time Warner Cable weren't included as participants, Thakker interpreted that it wouldn't have made a difference if one company served as both service provider and network operator. For example, that didn't help AT&T or Verizon rise above its peers. That said, as service providers rush to capture more of the consumer market, they mustn't ignore the importance of proper service level agreements (SLAs) with carriers, he added.

Analysts note some providers have already approached carriers to address such concerns. "It seems people are racing into the market," Siegel said.

"You pay for what you get," IDC's Stofega said. "My response would be some of these guys may not even be around to take advantage of the big upswing beyond the early adopter phase."

http://www.keynote.com

Beating The September Deadline: Providers Join VoIP E911 Race

By Johanne Torres

A few VoIP (define- news - alert) technology providers have already announced that they have actually beat the September deadline set by the Federal Communications Commission (FCC) to have E911 access working so consumers can reach emergency personnel when using a VoIP-based phone. Specifically, the agency mandated that all VoIP providers interconnected to the public switched telephone network must provide E911 with their consumer VoIP offerings.

Among the companies that announced access is up and running was Level3 Communications. The company said it expanded its E911 platform for VoIP to assist customers with FCC's compliance. Level3 said it's implementing a new emergency-calling network system for so-called nomadic VoIP users, and plans to extend its E911 coverage footprint to more than 70 percent of the country by the end of 2005. Additionally, Level 3 plans to offer E911 functionality through its (3)VoIP Local Inbound service, which enables customers to purchase inbound local, IP-based calling services in selected areas throughout the United States.

Another company that joined the bunch is Vonage Holdings Corporation. I actually came across some news reports that said the company set 10 million bucks aside to be used solely for its E911 efforts. The VoIP-based calling service provider began its E911 efforts by sending a letter to its subscribers during the last week of June.

The letter explained that the company has been routing 911 calls whether 911 has been manually configured or not (the company had previously provided users with instructions on how to do this manually). What seemed very interesting about Vonage's move then was that the company arranged to play an automated response saying: "Stop. You must dial 911 from another telephone. 911 is not available from this telephone line. No emergency personnel will be dispatched" when users dialed the three digits.

You might also remember reading about Vonage's partnership with Intrado a few months ago. Vonage arranged to team up with Intrado in order to fix the E911 dilemma. The VoIP provider even went as far as to claim that the service, along with E911 access capabilities worked just fine in the state of Rhode Island. Vonage's Jeffrey Citron explained last November that when Vonage customers dial 9-1-1 using Vonage's VoIP connection, Vonage would route the call over to Intrado. Intrado would then route the call to an operator who will determine where the call should go in order to resolve the emergency in question. With this Vonage/Intrado system in place, Rhode Island emergency service dispatchers will be able to obtain callers' call-back phone number (from IP) in case the call gets disconnected, complete name, and location when they dial 9-1-1.

Moving on to Intrado, in addition to its partnership with Vonage, the company also announced a partnership with USA Datanet in November. Intrado partnered with USA Datanet to deploy its Emergency Calling Service (ECS) nationwide. USA Datanet provides long-distance voice; Internet access, and IP network services to 480,000 residential and commercial customers. With this move, USA Datanet will be joining other Incumbent Local Exchange Carriers and wholesale VoIP providers currently dealing with Intrado.

Intrado also made a significant announcement about new infrastructure designed to enable VoIP Service Providers to provide E911 service for their subscribers throughout New York City. The Longmont, CO-based company worked with city officials and local 911 service provider Verizon to complete the deployment. The company describes the deployment as the first of its kind to define a VoIP Positioning Center (VPC) implementation model that can help VSPs meet the FCC's pending deadlines. According to the company's announcement, the V9-1-1

Mobility Service, Intrado's VPC offering, allows VoIP providers to route their subscribers' 9-1-1 calls into the dedicated wireline E911 network.

VoIP provider Nuvio Corporation (news - alert) also made a new E911 announcement towards the end of February. The company launched E911 services for its NuvioCentrex product, Nuvio's hosted IP Voice system. The system is available to broadband providers, cable operators, CLECs and VARs through Nuvio's private-label partner program. The company claimed to currently offer E911 in over 1,500 rate centers servicing over 2,700 cities nationwide and has plans to deploy this feature in additional markets this year.

Nuvio's E911 calls are routed as emergency traffic and provide computer-based caller information to emergency personnel at local Public Safety Answering Points (PSAPs). Through its service, 911 calls are automatically transferred to the PSAP and the operators are presented with the person's telephone number and location, ensuring that callers receive the same response they are accustomed to from traditional 911 services.

Nuvio has been very involved with the E911 portion of the whole VoIP-based services regulation issue. The company even had its CEO testify in opposition to Kansas State Legislature's attempt to tax VoIP providers for 911. In February, Nuvio's Jason Talley testified in opposition to proposed taxation of VoIP providers for 911 service by the State Legislature of Kansas. The VoIP-based service provider contested that in addition to being contrary to the FCC ruling, requirements for traditional 911 service were technologically impossible for VoIP. Nuvio also explained back then that it saw significant barriers to service implementation including call origination traceability and access to public safety answering points (PSAP).

Also joining the group is Dedham, MA-based RNK Telecom. The company announced that it will launch Edison, its E911 system. RNK described Edison as a potentially "life saving" emergency calling system for the company's VoIP resellers. RNK explained that Edison is a GPS-enabled system that will connect in-between a user's phone and VoIP connection, ensuring E911 systems continuously receive precise and updated information about the VoIP caller's location. RNK is currently beta-testing Edison, and plans to formally launch the service within 30 to 60 days.

Richard Koch, co-founder and CEO of RNK had this to say about the company's announcement: "In the first phase, RNK's Edison will relay the nomadic user's precise geographical location, enabling RNK to quickly know if the user's VoIP unit has been moved. This addresses the FCC's concern that customers need not self-report their location when they move. In a subsequent phase of development, Edison will provide updated geographical location to the appropriate E911 systems via a pseudo-ANI. RNK will simultaneously update the emergency ALI database, providing the end user's address and phone number."

As you can see, there has been a significant response from VoIP technology providers regarding the FCC VoIP E911 order. I'm sure we will see many more announcements to come in the next few weeks, as we count down the days until the September deadline.

http://www.level3.com
http://www.vonage.com
http://www.usadatanet.com
http://www.intrado.com
http://www.verizon.com
http://www.nuvio.com
http://www.rnktel.com

Excel Switching Carrier-Grade Media Gateway

Excel Switching Corp. (news - alert) announced the general availability of its Integrated Media Gateway (IMG) 1010, the latest addition to its AnyGen 1010 family of VoIP gateways and open service platforms.

Excel's IMG 1010 offers service providers the performance, reliability, and flexibility to introduce services across fixed and mobile networks worldwide. With its compact 1U package, integrated SS7 and rich media processing capabilities, the IMG 1010 is designed as a carrier-grade VoIP gateway and/or VoIP transcoder that enables service providers to reduce costs while improving service quality.

LatiNode, a Miami-based telecommunications company that provides international transportation of calls using VoIP, is currently using Excel's IMG 1010 for IP-IP transcoding to lower transport costs and provide a higher quality of service.

"We look forward to continuing our work with Excel as we build out our network," said Miguel Tarrau, CTO, LatiNode. "The unique capabilities of the IMG 1010 teamed with Excel's exceptional customer service and support creates a compelling package that we can leverage to quickly introduce new services."

Excel says that deployment of the IMG 1010 provides carriers with immediate benefits, including:

- Reduced capital expenses, resulting in the ability to quickly introduce new revenue-generating services to the marketplace
- Reduced operational expenses, including those associated with training, sparing, OA&M, environmental, and service agreements
- Carrier-grade reliability to ensure better system and network performance
- Investment protection via compatibility between existing systems and new offerings, eliminating the need for "forklift" upgrades.

http://www.excelswitching.com

Telstra Launches Asia Express In Collaboration With tel(x)

Telstra (news - alert) Incorporated announced its Asia Express offering to automate and simplify the global bandwidth procurement process. Asia Express is Telstra's indirect channel solution for connectivity from the U.S. to Hong Kong at aggressively-priced, monthly flat rates with no term commitment. Bandwidths from E1 to STM 4 international private lines are provided. As of July 1, Clear channel E1 to DS3 activation should take less than a week to install.

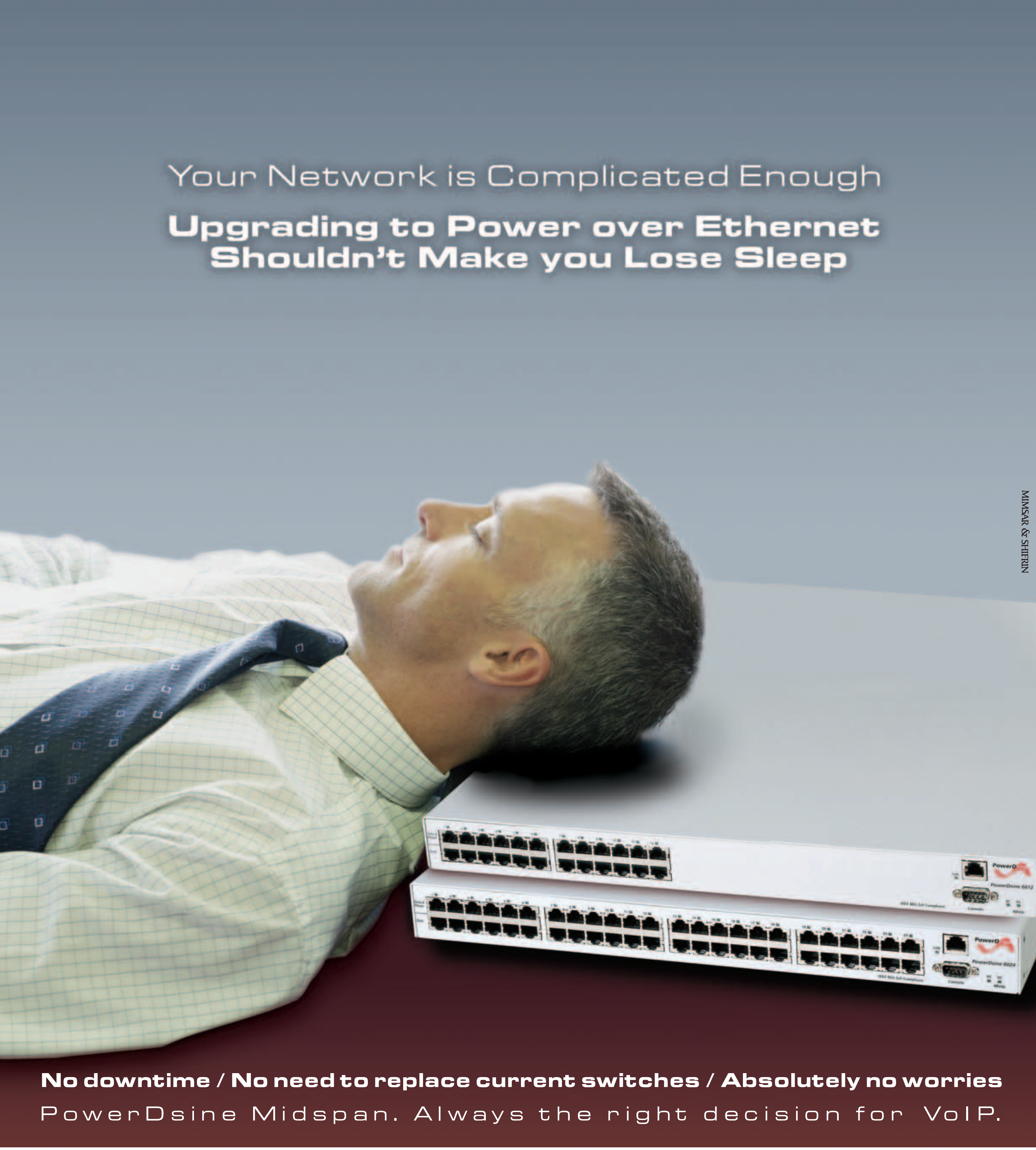
"This offering changes the rules and revolutionizes the way that carriers are going to be able to procure connectivity into Asia," said Arthur Weissman, Director, Channel Development, Telstra Incorporated. "We are introducing a new business and service delivery model that greatly simplifies the procurement process."

tel(x) (news - alert) and Telstra have entered into an arrangement allowing tel(x) customers to obtain a direct connection to Asia by simply cross connecting to the tel(x) Brilliant Platform. This service is designed to provide service providers the flexibility to enter into new relationships and to offer the services that their customers require ranging from voice services to global virtual LANs.

"Asia Express provides the streamlined transactions that carriers have been asking for while saving them from the activation delays and lengthy negotiations that plague today's global bandwidth marketplace," Weissman added. "It gives them the flexibility they need to try out new business relationships and new routes, or to manage risk on behalf of enterprise customers. This is not just a PoP to PoP (point of presence) solution. We can also provide connections directly to a customer's premises in Asia, as well."

http://www.telstra-usa.com

http://www.telx.com



Power over Ethernet (PoE) is a must when installing Wireless LAN Access Points, IP Security Cameras and IP phones. Adding PowerDsine Midspans to your network enables the simple, rapid and affordable convergence of voice, data, video and power on the existing LAN infrastructure. No need to replace current switches. With the plug-and-play PoE Midspan, you have a dream solution… rest assured you've made the best decision.

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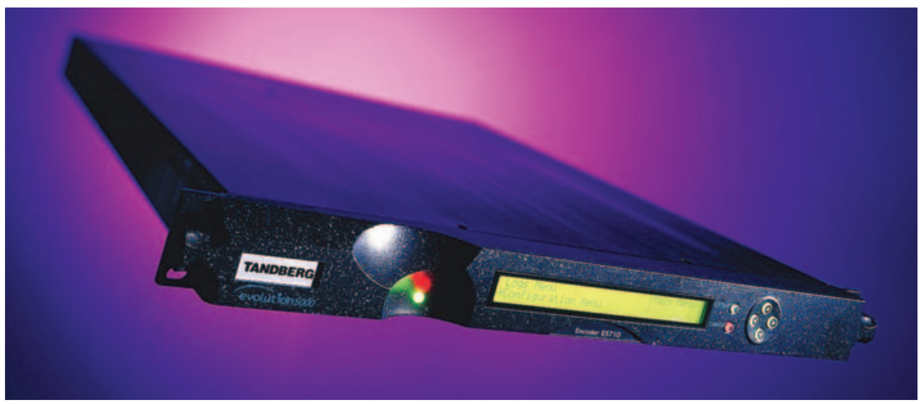
Citizens Selects TANDBERG

TANDBERG (news - alert) Television announced that Citizens Cablevision has selected TANDBERG E5710 MPEG-2 encoders for its high-action and premium channels that require very high compression performance. Citizens Cablevision, based in Floyd, Virginia, deployed a new IPTV headend in early 2005 that delivers digital television to customers in two separate areas. The new IPTV headend creates a triple play scenario of voice, data, and digital television for the company's current customer base, with the potential to reach more than 10,000 customers between the two businesses.

"The delivery of digital television services as part of a triple play package is essential to the long-term growth of Citizens," said Danny Vaughn, Network Manager, Citizens Cablevision. "It became very clear that our traditional business could not survive on legacy services alone. When we made the decision to offer video over DSL and hybrid fiber-coax, we recognized that certain encoders were incapable of delivering pristine-quality images for high-action, high-end channels that offer movies, sports, and other fast motion, highly detailed programming. We identified 12 channels that required the best possible encoding performance and invited eight vendors to participate in side-by-side testing. Each vendor encoded the same signal at the same data rate. TANDBERG Television proved that they could most reliably deliver high-quality video for our most detailed and premium programming."



The compact 1U E5710 with 2 option slots provides low bit rate performance through a unique combination of pre-processing, encoding techniques and enhanced noise reduction circuitry. The E5710 also provides an easy upgrade path to both the Windows Media 9 Series and MPEG-4 AVC platforms should Citizens decide to upgrade to advanced compression for significantly reduced bit rates in the future.

http://www.tandbergtv.com

Pirelli Completes Trial Of Persona FMC Solution

Persona Software, Inc., (news - alert) has announced that Pirelli Broadband Solutions, and Pirelli Labs have completed a successful trial of the Persona OnePhone Personal Mobility Application for Fixed Mobile Convergence (FMC) using Pirelli Broadband Solutions' Discus access gateways.

Persona OnePhone is an application that is designed to enable mobile users to roam seamlessly between WiFi and cellular networks with one phone, one identity, and one phone number. With Persona OnePhone, mobile users can access advanced IP voice features anywhere, anytime, plus a new world of advanced "Personal Mobility Applications" that provide road warriors and consumers with convenience, productivity, control, and savings.

Persona OnePhone uses the SIP standard, which has been embraced by service providers worldwide and is at the heart of the IP Multimedia Subsystem (IMS). Persona OnePhone is in trials with leading carriers and service providers worldwide.

http://www.personasoft.com

Go To Table of Contents I Go To Ad Index

Pyramid: Mobile Infrastructure To Reach $190 Billion

Mobile carrier capital expenditures (CAPEX) on infrastructure will enter a progressive decline beginning in 2006 that will see infrastructure investments decrease from 47percent of total operator CAPEX to 33 percent by 2009. While infrastructure spending will remain the largest slice of the CAPEX pie, Pyramid Research's new report, *Mobile Operator CAPEX: Charting the Transformation of Mobile Carrier Spending*, examines how vendors must adapt their business models to address the evolving mobile operator expenditure patterns to capture new, non-infrastructure investment opportunities.

According to report author Ozgur Aytar, "The rapid growth of non-infrastructure spending is due to the combined effect of factors ranging from demand for additional capacity to convergence and network evolution towards next-generation networks (NGNs). To provide end-to-end solutions, vendors will increasingly rely on partnerships and acquisitions of other sources of expertise. In the NGN world, service providers will evaluate the ecosystem as much as they evaluate the vendor itself."
http://www.pyramidresearch.com

Endavo EnHances Entertainment Product Line

Endavo Media and Communications, Inc., (news - alert) announced the latest addition to the Endavo Entertainment Product Line, the EnHance solution.

EnHance — powered by Red Swoosh — is designed to enable owners of Web-based portals to transmit DVD quality video and audio content to their community of end users with "broadcast economics" allowing for unlimited video delivery at a fixed cost while leveraging the full scale of the Endavo EcoSystem.

The EcoSystem provides content capturing, transcoding, storing, delivery, and transaction processing. The addition of EnHance to the EcoSystem allows content developers to store and retrieve content from both the EcoSystem and from a grid of connected machines within their own network that are idle, leveraging dormant bandwidth and providing twice the download speed of current delivery platforms for content-hungry end users.

Global Entertainment Media Group (GEMG) will be the first to deploy the solution in July.
http://www.redswoosh.com
http://www.endavo.com

Reef Point Systems To Launch UMA Security Gateway

By Ted Glanzer

Reef Point Systems (news - alert) announced that it is launching a massively scalable, multi-service UMA Security Gateway for UMA Wireless LAN or 802.xx networks. The gateway, according a statement from the Burlington, Mass.-based company, "solves the problem of the VoIP handset-to-network security challenge in the converged services infrastructure."

In a recent briefing, Reef Point Systems CEO Stephen Diamond highlighted several features that separate the UMA Security Gateway from other solutions.

For example, Diamond said that other solutions, such as session border controllers, can't support the Reef Point solution's scalability, which can secure a half a million simultaneous connections. Such scalability enables secure WiFi roaming and provides subscribers with confidentiality by encapsulating the call and signaling data in secure IPsec tunnels.

All of this significantly reduces capital and operating costs, Diamond said.

"UMA is a pre-cursor to IMS networks and enables operators to easily expand their coverage and introduce new mobile data services," said Diamond said. "These new services will not be widely adopted if there is a threat to the availability or integrity of the service. Reef Point offers a complete security solution and understands that these new services are critical to operators' profitability."

Diamond quoted AT&T CTO Hossein Eslambolchi, who said, "It's almost impossible to scale by putting all the security at the end of the network . . . We need to go somewhere in between . . . an intelligent network with smart devices."

According to the Reef Point, by deploying UMA technology, mobile operators can enable subscribers to roam and handover between cellular networks and public and private licensed wireless networks using dual-mode mobile handsets – providing subscribers a consistent and transparent user experience as they transition between networks.

WiFi and cellular convergence serves as a cost-effective method to improve residential coverage and keep churn down for operators, Diamond said.

While users are on local wireless LANs, UMA enables "free" VoIP (define - news - alert) when in office or home. The user, equipped with a dual-mode handset, can make called across any generic wireless LAN and IP network, with the call and signaling data encapsulated in secure IP tunnels. These tunnels terminate on an access gateway, which processes and passes call data to the circuit-switched or packet-switched mobile core network.

Additionally, the UMA Security Gateway offers the following:

- **Comprehensive Multiservice Security** — Threat defense, including firewalls with Denial of Service attack prevention, intrusion detection services, custom firewall filtering, dynamic virtual routing with Network Address Translation, and session limiting to protect against external threats.
- **Purpose-built, Carrier-class Design** — Performance and reliability through Reef Point's patented Flow Application Streaming Technology and optimum mix of custom ASICs/FPGAs and network processors.
- **Wireless Standards Support** — Reef Point solutions support Internet Key Exchange v2 and Extensible Authentication Protocol Method for GSM Subscriber Identity Modules (EAP-SIM) to provide scalable mutual authentication, encryption and data integrity safeguards for signaling, voice and data.

http://www.reefpoint.com

Viper Testing, Looking To Deploy WiFi Phone

Viper Networks, Inc. (news - alert) announced it is also joining the group of next-gen developers who are currently testing and will begin marketing a WiFi enabled phone that operates on its global VoIP network. According to Viper, the device, called WiFi vPhone 3100, will replace the 2100 model that the company launched in April of 2004. The WiFi vPhone will allow users to make phone calls over Viper's global VoIP network from any "hotspot" or location with 802.11 wireless Internet access.

The company explained that it hopes that increased availability of WiFi hotspots such as those found in many Starbucks, airports, restaurants, and other public and private locations, in conjunction with a new lower equipment price, will lead to increased sales of the 3100.

Viper Networks CEO Ron Weaver, who spoke about the potential of the product at last year's WiFi Business development Summit in Paris, commented, "I am ecstatic, as this moves Viper Networks far ahead of our competition in the VoIP market. We have been offering a WiFi VoIP device to our customers for over one year and are already offering a new and improved product at a better price."

The company will make the phone available for purchase on its Web site, through its distributors, and in its recently launched retail partner stores.

http://www.vipernetworks.com

Wireless Valley Announces Advanced Capabilities

Wireless Valley Communications (news - alert) announced new capabilities for capacity and traffic planning. Available in its EnterprisePlanner and LANPlanner wireless network design products, this new functionality is designed to enable businesses to tailor the design of wireless networks down to the user and application level. The products will ensure the capacity and coverage needed to handle the ever increasing demands being placed on wireless networks.

With this announcement, the company is expanding the predictive design capabilities of its software to include the number of users and the types of applications they are using.

"Wireless Valley has established itself as the leading supplier of design and management software for companies making the transition to wireless networks that support VoIP and other business critical applications," said Jim Welch, CEO of Wireless Valley. "Given the vital nature of these applications, it is not acceptable to experience dropped calls or delays in information transmission, which is exactly what will happen if the wireless network is not planned to support the proper amount of bandwidth and coverage. Only our software enables companies to consider all the factors impacting wireless network performance to create a highly customized, robust network."

SpectraLink, the leader in workplace Wi-Fi telephony, uses Wireless Valley's LANPlanner software to design wireless networks for customers using their NetLink Wireless Telephones. "To ensure the excellent voice quality our customers require, the wireless infrastructure must be engineered to provide adequate bandwidth and coverage to support call volume and seamless roaming," said Wayne McAllister, senior director of SpectraLink Service. "Wireless Valley's software has been invaluable in designing or upgrading customer networks to meet these requirements."

http://www.wirelessvalley.com

Go To Table of Contents | Go To Ad Index

Voice Call Management For TDM And VoIP

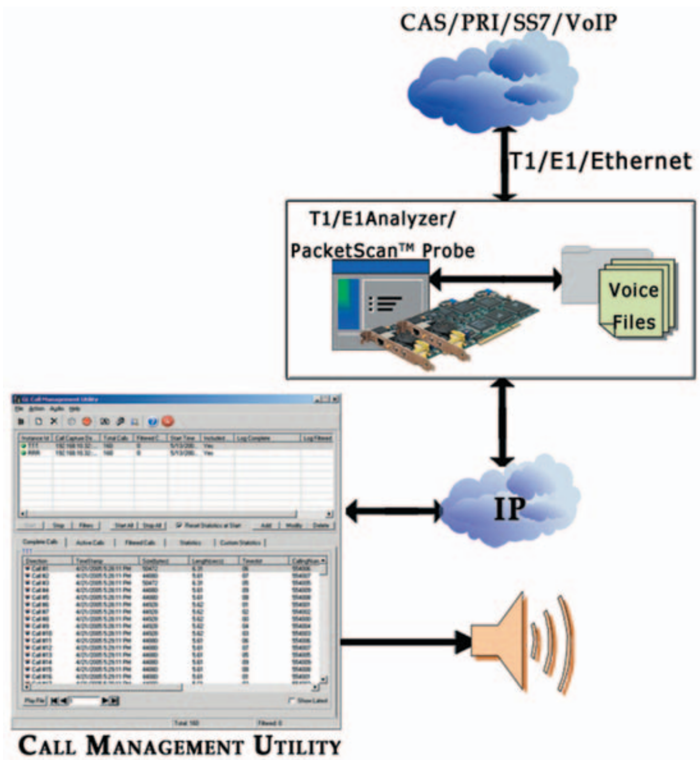
GL (news - alert) recently announced their Voice Call Management System (CMS) — a 'voice call logging' application designed to remotely monitor, record, and log every voice call occurring at any or multiple points in a voice network (TDM or VoIP). Distinguishing features include its ability to remotely listen to active or pre-recorded calls in real-time. In addition, the CMS can access multiple network locations simultaneously, whether VoIP or TDM, or both. The CMS works with associated "probes" such as GL's Call Capture Analysis (CCA) application with T1/E1 Analysis cards, or GL's VoIP PacketScan. The supported calling protocols include: CAS, PRI ISDN, and SS7 for TDM networks and SIP and H.323 for VoIP networks.

Significant features include: remote monitoring and playback of calls in real-time, call filtering and sorting, statistics gathering, recording of unlimited calls, and search for unique calls.

Calls are monitored and recorded in real-time at TDM and IP 'probe' sites. The call records are then transferred via IP to the CMS central location where they are immediately indexed and displayed as either 'active' or 'completed calls'. Filtering of call records can be based on a variety of user-defined parameters, and is available for real-time capture as well as for post processing. Complete statistics and/or user-defined statistics can be displayed and printed to file.

The recorded voice calls can be played to the CMS speakers, regardless of where the CMU is located with respect to the 'probes'. In addition, all recorded calls are stored on the 'probes' and may be searched/filtered based on called or calling number, time of day, or length of call.

http://www.gl.com



Fluke Adds Real-Time VoIP Diagnostics To OptiView

Fluke Networks (news - alert) announced that real-time VoIP diagnostics, including call-by-call Quality of Service tracking, are now part of its OptiView Link Analyzer and Protocol Expert network monitoring and analysis solution. The addition of the VoIP diagnostics is intended to enable network managers to view all levels of VoIP activity on their network, from very broad-based call volume and call quality to individual call channel details, thus allowing for real-time identification, isolation and repair of call quality issues.

"Voice over IP is one of the most mission critical applications being deployed by corporations around the globe," says Robert Finlay, Fluke Networks Product Manager. "Yet many IT managers can't check to see if their existing infrastructure will support VoIP, and even fewer have a plan for managing VoIP once it's deployed. Without useful monitoring and troubleshooting solutions like the new versions of OptiView Link Analyzer and OptiView Protocol Expert, call quality may cause nothing but user complaints and frustration."

The VoIP analysis features in OptiView Link Analyzer provide a real-time view of call volume and Quality of Service (QoS) for instantaneous identification of trends leading to call degradation. This enables troubleshooting before end users are impacted. Additionally, this information can be used to track down misconfigurations in network infrastructure. The real-time capabilities provide IT professionals with continuous vision of VoIP call quality in dynamic, changing network environments.

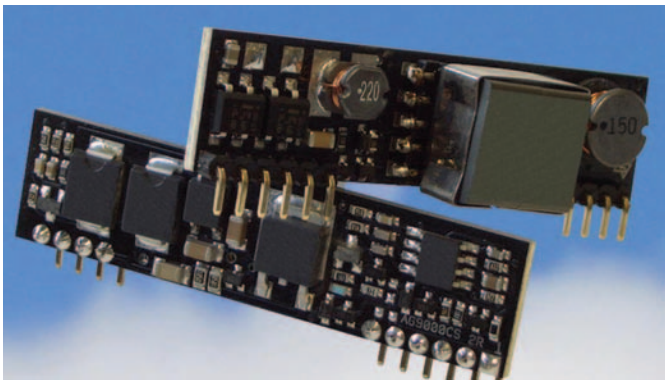
OptiView Link Analyzer and Protocol Expert are available for immediate delivery through Fluke Networks' sales channels worldwide. OptiView Link Analyzer 7.0 has a suggested U.S. list price of $18,995. Protocol Expert 7.0 has a suggested U.S. list price of $3,195.

http://www.flukenetworks.com

Silver Telecom Launch Power over Ethernet Module

Silver Telecom (news - alert) announced recently the launch of the Ag9000 series of Power over Ethernet (PoE) modules. According to the company, the Ag9000 is ideal for any Ethernet peripheral up to 12W, such as IP phones and gateways, wireless access points, Web cameras and security cameras, door entry systems, RFID tag readers, security systems (PIR, fire detectors, etc), network drives, Bluetooth access points and video servers.

The device is a fully self-contained 10-pin SIL (single in line) device, providing an isolated DC-DC converter with signature recognition compliant to IEEE 802.3af. Available with outputs of 3.3V, 5.0V or 12.0V, the Ag9000 requires just one external capacitor. The combination of features and packaging offers extremely efficient use of board area, saving significant system size and cost, minimizing time to market for developers of Ethernet peripherals.



http://www.silvertel.com

Acterna Integrates Telchemy's Technology

Telchemy (news - alert) announced that Acterna (news - alert) has expanded their use of Telchemy's VQmon/SA VoIP analysis and call quality monitoring technology. The technology will be integrated into Acterna's ng DA-3600A and DA-3400 Data Network Analyzers, PVA-1000 VoIP Analyzer, Digital Services Activation Meter (DSAM), and Remote Services Activation Meter (RSAM). For service providers and enterprises, Acterna's VoIP Analysis Software is designed to provide a comprehensive suite of problem-solving capabilities necessary to deploy, troubleshoot, and maintain today's complex VoIP networks.

The DSAM and RSAM are part of Acterna's Application-Aware suite of products that provide cable operators with tools that test complete digital and IP services with simplicity, flexibility, and ease. Adding VoIP support to these leading edge products make them more powerful than ever — VoIP deployment for network operators becomes increasingly easier to manage with the ability to place and listen to a live call, measuring the quality of the user-experience before going live to the customer.

http://www.telchemy.com
http://www.acterna.com

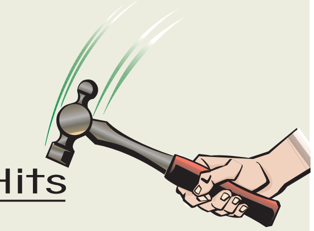
Z-Com Selects TI Embedded WLAN And VoIP Solutions

Texas Instruments (quote - news - alert) has announced that Z-Com (news - alert) has selected TI's WLAN and VoIP products for its universal serial bus (USB) adapter and residential gateway (RG) module product lines, enabling consumers to benefit from high-quality, integrated broadband data, wireless, and voice applications.

Z-Com has developed the XG-750 USB adapter, which plugs into the computer's USB port to enable seamless wireless connectivity. The product is based on TI's TNETW1450 WLAN solution which features a compact design and high security and quality of service (QoS) standards. In addition, Z-Com's RG module incorporates TI's TNETW1350A solution, which features industry-leading output power enabling extended range operation.

Z-Com has also selected TI's TNETV1060 VoIP solution for its P-1050 VoIP gateway, the company's first product targeting the VoIP market. A highly integrated software and silicon system-on-a-chip, TI's expandable and scalable solution is designed to address the requirements of residential and small office/home office (SOHO) gateways.

http://www.ti.com/broadband
http://www.zcom.com



Quick Hits

Spirent Partners With Objective Concepts

Spirent Communications (news - alert) and Objective Concepts announced the availability of the Spirent TradeTest solution powered by Objective Concepts. The new offering is designed to provide the financial securities industry with a highly sophisticated toolset for automated testing of trading systems. Using real-world market and trading data feed interface simulators for order entry and primary exchange feeds, TradeTest creates a virtual market place in the quality assurance lab.

http://www.spirentcom.com

SEI Receives PoE License

SEI (news - alert) has received a non-exclusive license to manufacture and market Power over Ethernet (PoE) solutions under U.S. patent number 6,476,608 from PowerDsine. This patent forms the basis of the IEEE 802.3af powering standard both for the powering algorithms inherent in the standard and also for the delivery of device power over the Ethernet in using the spare pair.

http://www.seipower.com

Intel Turns To Psytechnics

Intel Corporation (quote - news - alert) has selected Psytechnics to develop perceptual video quality management solutions. Psytechnics (news - alert) has dedicated the last five years to developing voice quality solutions, and now the company has applied the same methods to deliver its video solutions. The burgeoning market for video-based services, such as IPTV and video streaming, combined with new technology, is driving the demand for video test and monitoring applications.

http://www.intel.com
http://www.psytechnics.com

Go To Table of Contents | Go To Ad Index

Mindspeed And Netbricks Collaborate On VoIP Integration

Mindspeed Technologies, Inc., (news -alert) has announced an enhanced system-on-chip (SoC) voice processing solution for enterprise PBX, gateway, and voice-enabled router equipment with the integration of Netbricks' (news -alert) SIP-Bricks session initiation protocol (SIP) software into Mindspeed's Comcerto integrated VoIP processor.

"Mindspeed's Comcerto processor provides superior processing power for implementing the features of our SIP-Bricks solution, providing faster time-to-market for our customers" said Didier Raffenoux, president of Netbricks.

"The combination of Netbricks' SIP-Bricks software and Mindspeed's complete VoIP system-on-chip portfolio offers the industry's most compelling solution for OEMs who wish to leverage their development across enterprise and carrier applications," said Steve McClure, executive director of marketing for Mindspeed.

Mindspeed's Comcerto architecture is optimized for VoIP applications with separate signal and packet processors. It integrates high-performance DSP cores and a packet-processing engine with industry-standard streaming packet interfaces to provide a high-density VoIP and voice-over-ATM system on a single chip.

http://www.mindspeed.com
http://www.netbricks.com

Ubiquity, IBM In Service Creation Pact

Ubiquity (news -alert) Software recently announced an agreement with IBM (quote - news - alert) to provide an integrated next generation service creation and delivery solution for service providers. The joint solution is designed to enable service providers in both enhanced VoIP and 3G IP Multimedia Subsystem (IMS) networks to design and streamline the creation and deployment of new services in fixed, mobile and broadband networks.

The integration of Ubiquity's SIP Application Server into IBM WebSphere software provides service providers faster, easier runtime deployment and management while orchestrating the high availability strengths of J2EE and SIP Servers. Integrated with IBM Rational software, the Ubiquity SIP Toolkit offers service providers a unified development environment, managing the lifecycle of service creation. Ubiquity solutions are enabled with Linux, IBM DB2 Universal Database, IBM eServer BladeCenter and IBM eServer xSeries. The IBM BladeCenter integrates servers, storage, networking and software to create an environment that is reliable, secure and easy to deploy and manage.

"Formally integrating the Ubiquity SIP Application Server with the unified service creation, deployment and management capabilities in IBM's middleware portfolio provides carriers with accelerated time to market for new SIP services, while reducing the complexity." said Tim Greisinger, director, Worldwide Communications Sector, IBM Software Group.

As service providers continue their network transformation to next generation networks, the IBM and Ubiquity integrated software solutions enable the enhancement and blending of services that become available with IMS adoption.

http://www.ubiquitysoftware.com
http://www.ibm.com

Go To Table of Contents | Go To Ad Index

Net-2Com Develops Softswitch, Phones On RADVISION SIP Kit

RADVISION (news -alert) has announced that Net-2Com Corporation has used RADVISION's SIP Toolkit to develop its new WiPCom1000 wireless IP telephone as well as its SubCentrex IP-Centrex carrier-class softswitch.

Net-2Com (news -alert) uses a proprietary VoIP technology to develop and distribute innovative VoIP and networking-related products for enterprise environments. Its WiPCom1000 is a wireless/IP phone that features built-in wireless LAN (WLAN) functionality, allowing it to be used automatically in public WLANs.

The SubCentrex carrier-class softswitch is an IP-Centrex enterprise solution that supports the conversion between SIP and H.323, enabling end-to-end IP telephony between PSTN users.

"By selecting the RADVISION SIP Toolkit, we realized immediate benefits, ranging from direct access to an up-to-date communication protocol to unparalleled development flexibility made possible by the multiple API layers," commented Kiyoshi Oh, CEO of Net-2Com. "With RADVISION's SIP Toolkit, our development team was able to focus on service functions rather than protocol and interoperability details, which directly resulted in faster service rollout and lower costs."

http://www.radvision.com
http://www.net-2com.com

ATTENTION VENDORS!

Send your **News** and **Product Releases** via e-mail to **itpress@tmcnet.com.** Whenever possible, please include high-resolution (minimum 300 dpi) color graphics (.BMP, .EPS, .TIF, or .JPG).

Contact Center Companies Concerto And Aspect To Merge

By Johanne Torres

Concerto Software (news - alert) and Aspect Communications Corporation (news - alert) announced a definitive agreement to merge operations. The new agreement will assign Aspect shareholders $11.60 in cash for each share of common stock, which represents an approximate 15 percent premium to the average closing price over the last 30 trading days prior to the announcement. The merger also asks for holders of Aspect Series B Preferred Stock to receive an equivalent amount of cash per share on an as-converted basis. Based on the number of shares of Aspect common stock, common stock options and Series B Preferred Stock outstanding on July 4, 2005, the transaction is valued at approximately $1.0 billion.

The companies involved in this transaction claim that the merger will form one of the largest companies solely focused on contact center products and services in the space. The deal will combine Concerto's predictive dialing and unified contact center system products with Aspect's contact center workforce management applications and performance analytics. Its also key to mention that the merger will combine both companies' traditional voice and VoIP automatic call distributors (ACD), enabling the new entity to offer multi-channel routing, self-service interactive voice response (IVR), Internet contact and virtual contact center capabilities, reporting, monitoring and recording.

"With greater resources and scale, we will offer customers a more robust product suite, an expanded services and support infrastructure, and increased investment in research and development, making the new company uniquely positioned to be the contact center solutions provider of choice," said James D. Foy, Concerto's president and CEO. "Concerto has a proven track record of integrating people and products following acquisitions. Given the customer- centric philosophy that Concerto and Aspect share, we anticipate a rapid, successful transition."

"The merger with Concerto is an exciting opportunity to create a new standard of excellence in customer interaction. The contact center market is evolving with the advent of new technologies, like voice over IP, and growing customer demand for reduced complexity and increased capabilities. Together, we will focus on continuing to support our customers today and bringing converged solutions to them as their business needs dictate," said Gary E. Barnett, president and CEO of San Jose, CA-based Aspect. "We are pleased with this outcome. I and other members of the Aspect executive team look forward to being a part of the combined company."

Aspect had recently announced the latest release of its contact center operating environment, the Aspect Uniphi Suite 6.1. The product offers businesses ACD, CTI and IVR, on a single, centrally managed, switch-agnostic platform. The company said the new version 6.1 simplifies application development, system administration and management, while doubling the number of agents supported by last year's release. Uniphi Suite v6.1 is currently available and it includes support for up to 500 agents (blended and voice-only), Microsoft .NET and Windows, Aspect Call Center, Cisco CallManager, Microsoft Exchange and eGain Service. The software is built on core standards, such as VXML (define - news - alert), SOAP/Web (define - news - alert) Services and SIP (define - news - alert).

"Aspect Uniphi Suite helps companies greatly simplify contact center infrastructure and manage contacts and resources easily, so agents can deliver superior customer service experiences," said Brian Gentile, senior vice president and chief marketing officer at Aspect. "The quality of customer care suffers when contact centers are stuck with mismatched enterprise technologies. Even making simple changes in business processes becomes a complex and frustrating task, adding to implementation time and cost, as well as maintenance. We designed Uniphi Suite to address these challenges and to provide excellent support for the migration to VoIP networks and for replacing traditional PBXs with IP-PBXs."

http://www.concerto.com
http://www.aspect.com

Go To Table of Contents | Go To Ad Index

Genesys Announces IP-Enabled Voice Platform

By David Sims

Genesys Telecommunications Laboratories Inc., (news - alert) has announced an IP-enabled version of the Genesys Voice Platform "to serve enterprise customers of all sizes," according to Genesys officials.

The introduction of the new IP-enabled enterprise edition of GVP is part of Genesys' strategy to expand IP capabilities throughout the Genesys Voice Platform product portfolio. One feature that's key in marketing the GVP to enterprises actively considering migrating to IP is that the product has been designed to help them do so without an extensive system overhaul. The new IP-enabled version of GVP allows for time-division multiplexing (TDM) and IP to coexist within the same environment.

Existing TDM-based environments limit the return on investment on contact center software due to the inherent complexities and high cost of ownership at the infrastructure level. Genesys is hoping their adoption of an Open IP approach will convince enterprise customers that they can increase their ROI.

"Enterprise customers that are actively moving forward with IP initiatives will find value in the self-service and ROI deliverables of the IP-enabled Genesys Voice Platform," said Elliot Danziger, chief technology officer, Genesys.

The GVP supports open standards, such as VoiceXML 2.0 and Session Initiation Protocol, and migration isn't rip and replace, but a phased approach. "The transition to IP will not require a drastic cutover as the GVP TDM and IP versions can coexist within the same environment, allowing for customer choice for voice-over-IP infrastructures," officials say.

http://www.genesyslabs.com

Avaya Tweaks Contact Center Express

By David Sims

Avaya Inc., (quote- news - alert) has announced enhancements to Avaya Contact Center Express that "will enable medium-sized businesses to use intelligent communications to better serve their customers," according to company officials.

"We know that medium-sized contact centers want to benefit from the same IP telephony, multi-channel and agent empowerment capabilities as larger enterprises — but need solutions that are much easier and more cost effective to deploy," said Eileen Rudden, vice president and general manager, Enterprise Communications Applications Division, Avaya, explaining the targeting of the mid-market.

Rudden added that the features added to Contact Center Express are designed to lower contact center costs for their mid-size business target.

The software enhancements to Avaya Contact Center Express, which is tightly integrated with Microsoft applications, include a new Microsoft CRM connector that identifies incoming calls received by the contact center and provides an appropriate "pop up" with customer history on an agent's screen.

The Microsoft CRM connector also enables agents to click a button on their Microsoft CRM screen to place outgoing calls. Contact Center Express now also handles and routes instant messages from customers using Microsoft Messenger for another channel of contact.

Other enhancements include simplified management tools, easier e-mail handling and spell-checking, greater customization capabilities and more detailed reporting.

Contact Center Express has always been aimed squarely at the mid-size business market. Its multi-channel features of inbound and outbound voice, e-mail, Web chat and Microsoft Messenger for instant messaging, packaged with Avaya's IP telephony and contact center routing and reporting and priced from $425 per concurrent agent for voice to $1,000 per concurrent agent for all channels is a classic mid-market offering.

It's where the action's going to be over the next few years, too. The global market for contact center technology is expected to grow to $5.1 billion by 2008, with the most growth over the next five years in the mid-market in contact centers with up to 250 agents, according to Datamonitor.

http://www.avaya.com

Go To Table of Contents | Go To Ad Index

Powerdsine Joins Cisco Technology Developer Program

PowerDsine Ltd. (news - alert) has recently been granted membership of the Cisco Developer Program. Program membership is reserved for companies offering complementary and compatible solutions that hold a market leadership position in their respective market sector. It offers customers and resellers the reassurance of timely resolution to support enquiries involving both companies' solutions.

PowerDsine produces IEEE 802.3af-compliant PoE Midspan solutions that supply operating power via Ethernet cables to powered devices such as Voice over IP (VoIP) phones, wireless LAN access points and IP-based cameras. PowerDsine's PoE products can reduce the time and costs associated with installation of such end-devices because PoE technology eliminates the need to wire additional AC outlets.

PowerDsine joined the Cisco (quote - news - alert) Technology Developer Partner Program as a Cisco Compatible Wireless Partner in April 2005.

http://www.powerdsine.com
http://www.cisco.com

SNET DG Partners With Stealth Communications

SNET DG (Diversified Group) (news - alert) and Stealth Communications (news - alert) announced that SNET DG will interconnect with Stealth's Voice Peering Fabric (VPF) as an application services provider to offer VPF members SIP access to Intelligent Network Database Information such as, CNAM, 8XX, and LNP.

"VoIP providers need access to legacy telco database information for calls that interact with the PSTN. By joining the VPF family, SNET DG is offering VPF members quick and easy access to the information they need without the complexity and expense of an SS7 network, or handing off database queries to a TDM-based carrier," remarked Suzette Taylor, Business Development Manager at SNET DG. "Our service supports the spirit of the VPF, which is to extend the reach of VoIP and help VoIP providers operate more efficiently and therefore, more profitably."

http://www.snetdg.com
http://www.stealth.net

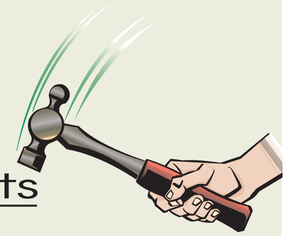
Equant Announces Communications Outsourcing Contract With STMicroelectronics

Equant (news - alert) has signed a multi-year contract with STMicroelectronics (news - alert), for more than $100 million, to provide a complete outsourcing service that includes data, fixed and mobile voice and service management to more than 100 sites in 34 countries.

As part of this communications outsourcing deal, Equant will provide a full set of communications services, including data services based on IP VPN, fixed and mobile voice services including VoIP and IP Telephony, as well as service management. Equant has already installed 40 VoIP sites in 22 countries. This contract also includes the transfer of several hundred third party contracts to Equant.

"This new contract demonstrates the success and development of Equant's services strategy," said Michel Picaud, senior vice president and head of Equant Solutions & Services. "We are extremely pleased to provide the worldwide communications infrastructure for STMicroelectronics, a true industry innovator.

http://www.equant.com
http://www.st.com



Quick Hits

Uniden, 3Com Work Out Cordless Phone Deal

Uniden America Corporation (news - alert) and 3Com Corporation (quote - news - alert) announced an agreement to deliver VoIP cordless multi-line phones for enterprises. Available immediately, the 3Com 3106C cordless phone and 3Com 3107C cordless phone can seamlessly be used with the 3Com NBX IP telephony system to deliver complete and cost-effective IP offerings to business owners.

http://www.uniden.com
http://www.3com.com

Earthlink And Covad Announce Market Trial

Covad (news - alert)and EarthLink (quote - news - alert) have announced a new VoIP joint market trial that will give EarthLink the ability to offer its customers low-cost phone services along with high-speed Internet access. The three-market trial is slated to begin in October in Dallas, TX, San Francisco and San Jose, CA, and Seattle, WA.

http://www.earthlink.com
http://www.covad.com

Cisco Adds D&H As Authorized U.S. Distributor

Cisco (quote - news - alert) Systems announced it has added D&H (news - alert) Distributing to its list of authorized distributors in the United States. This distribution relationship focuses on accelerating SMB customer adoption of networking technology, combining Cisco SMB-tailored technology with D&H's commitment to provide superior service and support to the value-added resellers that serve this rapidly growing customer segment.

http://www.cisco.com
http://www.dandh.com

Go To Table of Contents | Go To Ad Index

Telemedicine: The ER & Beyond

No longer relegated to just back office roles or for secondary education, telemedicine has increasingly become an integral part of a healthcare provider's practice even reaching into the field of emergency medicine.

By Robert Liu

TORONTO — In the remote town of Marathon, Ontario, a three-year-old boy named Callum sat in his bed with a fever, coughing and suffering badly from symptoms of pneumonia. In the dead of winter, the small child was brought in and admitted to the local hospital of his rural town, which is located over 600 miles (1,000 km) to the north of this bustling metropolis. Callum's parents and his family practitioner knew he desperately needed the medical attention of a pediatrician but the closest specialist was over 180 miles (300 km) to the west in the city of Thunder Bay. Even on a warm summer day, it was a four-hour ambulance ride. In the unrelenting snow of January, it was a hazardous race against time that exposed Callum not only to the elements but also to the very real dangers of slamming into a moose.

Today, Callum is a healthy boy still living in Marathon. He never needed that ambulance ride in January 2004; he was able to stay in his local community. Instead, the pediatrician came to him through an IP-based video link that tapped into a tele-health network operated by the Northern Ontario Remote Telecommunications Health (NORTH) Network, a membership-based program that receives funding from the Ontario and Canadian governments.

"There's a huge shortage of health professionals in Ontario and this technology is dramatically improving access to care for our rural population," said Dr. Ed Brown, an emergency room physician who was the founder and now is executive director at NORTH Network.

What started as an experimental partnership among four hospitals in 1998 has now blossomed into a wide-area network of 170 sites (hospitals, universities, long-term care facilities, nursing stations, public health and ambulance units) in 105 communities that spans across the 460,000 square miles of the province of Ontario. Callum's case was NORTH Network's 10,000th consult. By May, 31, 2005, the program already hit the 25,000 mark and is expected to reach 50,000 consults by the end of next year.

The concept of telemedicine (or telehealth — the terms are synonymous) may lend itself well for the Canadian market, where healthcare is public and land is vast. Of the 12 million people that reside in Ontario, nearly 30 percent live like Callum in rural areas where healthcare professionals are less likely to set up practice. But the growing importance of telemedicine on the healthcare world isn't just isolated north of the U.S. border. Whereas in previous years, two-way IP-based communications only served healthcare professionals with basic corporate video conferencing needs or secondary education (i.e., teaching sessions), companies like Polycom or Tandberg have refocused their efforts on the actual practices of healthcare providers.

In its simplest form, telemedicine is essentially video conferencing with a few extra bells and whistles. Those bells and whistles may include digital stethoscopes, sphygmomanometer (blood pressure), handheld patient examination camera, pulse oximeter, etc., in addition to two-way television transmission. With a combination of tools to examine vital signs, eyes, ears, nose and throat, a doctor or nurse could gather enough information to diagnose the patient.

"If there's a growth area, it's in the practice of telemedicine itself," said Tim O'Neil, Acting Global Director of Polycom's Healthcare practice. "The doctor has a certain amount of budget to treat you and those dollars aren't increasing."

In practice, if a patient can avoid hospitalization, then those added resources could be reinvested back into the entire healthcare system to everyone's benefit. That's exactly what has happened in the field of palliative medicine. After establishing a tele-homecare practice to monitor patients suffering congestive heart failure, NORTH realized reduced hospitalization rates of 30 to 80 percent. "Very, very dramatic numbers," Dr.

Go To Table of Contents | Go To Ad Index

Brown said during a recent on-site tour sponsored by Cisco Systems Canada Co. (NORTH has built its network infrastructure entirely on Cisco 7200 series routers with dual-power supplies and other redundancies.)

The reliability of its network as well as its own success has led NORTH to even roll out telemedicine initiatives in the areas of emergency care. For example, if a stroke is detected early, a victim can be administered clot-busting drugs to reverse the process. In addition, NORTH received funding from its local utility for an emergency telemedicine program for electrical burn victims. Other areas include clinical care, dermatology, allied health professions like speech therapy, and rehab specialists.

Telemedicine-enabled Communities Partnering with NORTH Network

Of course, not everyone is convinced that telemedicine has reached the threshold of critical mass. Earlier this summer, the Canadian Supreme Court in a ground-breaking 4-3 decision ruled that denying a citizen the right to purchase private health insurance plans was unconstitutional and jeopardized the well-being of Canadians. But that alone isn't likely to thrust telemedicine into the spotlight.

"The remote solutions make great press but my guess is the numbers are not that big," said Tom Astle, senior vice president at National Bank Financial's Technology Group.

While the NBF analyst currently stands correct, a confluence of other factors has changed the face of telemedicine. For example, ten years ago in the field of homecare, Medicare only reimbursed healthcare professionals if they actually paid the patient an on-site visit. But since the enactment of the Home Health Prospective Payment System (PPS) on October 1, 2000, Medicare began reimbursing healthcare providers between $2,000 and $6,000 per 60-day term depending on the level of care that the patient required.

"For the first time, that allowed us to consider employing telehealth as an adjunct to face-to-face visits. Up until October of 2000, there was no economic imperative," said Marcia Reissig, president of Boston-based Partners Home Care, which cares for some 3,000 patients on any given day. Of its total base, approximately 150 patients are receiving telemedicine monitoring services.

In June, lawmakers in Congress introduced the Medicare Tele-health Enhancement Act of 2005. If enacted into law, the legislation would provide $30 million in funds for tele-health programs.

"The majority of U.S.-based telemedicine initiatives were cost-reducing. That's turned into revenue-generating," Polycom's O'Neil added. IT

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By Tony Rybczynski

Site-By-Site Versus Centralized Telephony

Centralization of IT resources has traditionally been the path to lowest TCO, whether we are talking about application processing, or data storage. Centralization provides higher utilization of servers and reduces server costs, reduces operations costs including power, staffing, and management, and provides increased agility to accommodate network changes and new sites. Do these concepts equally apply to IP telephony?

IP telephony partitions telephony system functionality into a number of elements, which themselves can be centralized or distributed as required across a telephony-grade IP network: clients, communications servers, media gateways, and application servers (e.g., for unified messaging and conferencing). Whether centralization or distribution of this functionality is right for you depends on how well either approach meets your business, reliability, functionality, and price/performance requirements.

Factor 1: Remote Site Business Model

If remote offices are employee-centric, economics rather than customer service may be the primary deciding factor for a centralized architecture. On the other hand, if remote offices are run as separate businesses and traditionally look after their own telephony needs (e.g., in some franchise environments), then a distributed model may best fit your environment. In addition, if service functionality and consistency for telephone customers needs to be delivered at your branches under various failure conditions, this requires careful attention to calling scenarios and is easier to achieve with a site-based IP telephony approach.

Factor 2: Business Continuity/Disaster Recovery and WAN Reliability

The key questions are how reliable is your WAN and at what cost, and what is acceptable business impact in case of WAN failures.

There are three general approaches to meeting these requirements. Firstly, you can centralize communications and application servers, with PSTN (define - news - alert) gateways deployed on a city basis to minimize cost of handling public voice calls. Secondly, you can centralize communications and application servers, but deploy survivable gateways at each site to provide key telephony features and PSTN access in case of network failures. In these cases, the cost benefits of centralizing telephony functionality need to be weighed against the costs of upgrading the IP network to provide the required level of reliability and availability. Thirdly, you can add IP telephony on a site by site basis, with less reliance on the network.

In all cases, you should consider business continuity and disaster recovery requirements for any IP telephony communications server that is handling a large number of users, whether at larger sites or in data centers. Solutions range from cold standby to active-active system designs.

You should consider business continuity and disaster recovery requirements for any IP telephony communications server.

Factor 3: WAN Bandwidth and QoS

Another factor to be considered is the obvious need to provide adequate bandwidth for voice and associated QoS capabilities, remembering that a highly compressed IP telephony call uses 24Kbps. The bandwidth to each site must be able to handle all inter-site data and voice traffic. Under failure conditions, there has to be enough bandwidth to handle priority traffic, including enough CO trunk capacity to handle calls to be offloaded to the phone network. Intra-site calls don't use up WAN bandwidth, except for a low level of signaling traffic for centralized deployments. Centralization results in increased traffic to access application servers, such as conference bridges and voicemail.

Factor 4: Client Mix

Clients are a major cost of any telephony system. In a new remote office or branch, client choices include wired and wireless telephone sets, and IP telephony and multimedia soft clients. For existing sites, there may be an economically driven desire to retain current digital sets, complementing these with IP telephony clients where required. Digital sets and incidental analog sets (and fax machines) can both be handled by line side media gateways.

Factor 5: Business Economics and Migration Risk

Centralization puts increased demand on the network in terms of bandwidth and reliability, the cost of which needs to

Go To Table of Contents | Go To Ad Index

AD

be included in any cost-benefit analysis. In addition, building in adequate levels of redundancy in the central site requires additional investments. For some enterprises, it may be more attractive to take a nodal approach to IP telephony, and thus limit the impact of possible failures to a single site. Specifically, in many customer service branch environments, service continuity is paramount and may not be economically achievable with a centralized approach. Finally, leveraging installed base investments in an IP telephony environment can be a very positive contributor to a business case.

> **In deploying large scale IP telephony systems, you should carefully assess your business needs.**

Centralize If You Can; Distribute If You Must

In deploying large scale IP telephony systems, you should carefully assess your business needs and the degree of centralization that best meets these needs. Distributed architectures more closely follow today's approaches and are the lower risk path particularly during the transition period and if you are running a customer-centric network. In many cases, you will opt for a hybrid approach with some degree of centralization (e.g., around the metropolitan area where your head office is located) and distribution (e.g., for off-shore sites where network costs may be considerably higher). Alternatively, you can decide to distribute your IP telephony systems while centralizing your unified messaging and multimedia capabilities. IT

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By Hunter Newby

The Transformation Continues…

VoIP Peering has its segments which are commonly known as carrier and user. Within the user segment there are two groups, end user and enterprise. Aside from the international, wholesale carrier version of VoIP Peering, which might be better described as SIP Peering, its impact on every aspect of the domestic carrier networks is substantial and has an effect on the user side of VoIP Peering as well. This massive network transformation is occurring simultaneously across the country and world and it is causing most people (end users, businesses, and carriers) to completely change the way they think about networks and the business of voice communications.

As if the retail side of the telecom business knew what direction it was heading in with broadband voice and flat rate offerings, it seems it might get broadsided by gaming services that allow multiple users to play each other and speak in real time as they do it. Now gaming services, such as Xbox Live, will be in a position to enable VoIP (define - news - alert) calls, a.k.a. end user VoIP peering, as a separate function not tied to any game being played. So, the users of this service are now all on the same network platform and can call each other for free. It may seem small and niche today, but those 13–18 year olds are tomorrow's 25+ crowd and they won't know their telephonic lives any other way. The one downside to this is (and other Internet VoIP offerings such as Skype) is that they ride the public cloud and therefore quality and security are major concerns, but that's nothing a private VoIP network can't fix.

Other than that the MSOs and VoBBs (define - news - alert) seem to being doing quite well getting new subscribers. They are also further down the line with ENUM VoIP Peering than any other service provider group. As slow as the ILEC/RBOCs may seem to be, did anyone notice the Verizon announcement at SuperComm calling for video competition? The MSOs might need to get some Windex on their rear-view mirrors. Imagine the ILECs beating the MSOs to video peering!

The corporate side of things in VoIP Peering is moving along nicely as more and more small, medium, and large enterprises are in various stages of learning, evaluating, and deploying VoIP in to their own private WANs. The earliest adopters are beginning to realize the benefits of peering these purpose built VoIP networks. This segment continues to show strong growth and buzz in the industry, which is driving many service providers to try and meet the demand for VoIP services.

Right now businesses are trying to figure out what to buy that will lower their costs. There is a lot to learn and analyze and many are not reaching the ultimate correct conclusion because they are seeking knowledge from service providers (and some equipment vendors) many of which don't have the right services to sell in the first place. Some of those providers are not VoIP enabled at the edge, or customer interface, so that keeps the TDM (define - news - alert)for provisioning purposes and its associated costs and delays in place. No gain there.

Many are also still trying to sell voice billed per minute. This model will carry on for a while, but its days are numbered. The reason is that the corporate IT director, who is now the voice buyer and who also took over the telecommunications managers' job, has Vonage, or some other VoBB service at home and is already paying a flat rate for voice. They know nothing of the old way, 1+ contracts with loops and terms and traffic commitment levels. To them voice is a flat rate service. It's just data.

So, the old school service providers push what they have and it doesn't seem to fit the technology needs, or financial requirements and it is back to the drawing board. Now, this isn't the case with them all, but most of the legacy voice carriers that are trying to morph are stuck in-between here and there and they're looking like it. Trying to protect the old revenue model with a VoIP marketing make-over is like putting a $5,000 paint job on a rusted out car. The end result is full of holes.

The carrier side is almost a mirror image of the buyers. They're all learning as they go feeling out where the demand pockets are while trying to pick the right solution at the right price. It's a juggling act. Many of the providers have launched VoIP services in response to the trends and media attention, but many are not fully cooked yet and they are only available in limited locations. This is very problematic when the sales reps get a multi-site opportunity, but can only handle a small piece. Many have VoIP enabled their core, but not the edge. It is the edge that enables the provisioning and trunking benefits out to the customer. That needs to be IP (Ethernet really), or else we're still dealing with yesterday's problems to the customer premise. That said there are many who are now offering flat-rate domestic calling via SIP and an Ethernet cross connect at major carrier hotels.

Don't forget, this isn't an Internet delivery model for the mid to large size business, this is a broadband access (loop) model. Most businesses don't want to use the Internet for corporate voice traffic. In addition, the way in which the IT departments are building their VoIP WANs is a huge opportunity for the Ethernet transport providers who don't even have a VoIP service to offer. As a testament to that I want to

Go To Table of Contents | Go To Ad Index

share with you an e-mail I received from an Ethernet transport provider in response to my June column entitled *The Only Constant is Change*.

Hello Hunter,

It was a pleasant surprise to see your picture while flipping through Internet Telephony Magazine. Your article was very relevant and pointed. Here at American Fiber Systems we are selling quite a bit of data pipes that our customers are using for VoIP and MPLS. The tide is indeed changing and business' challenge is to create new products and services on the new technology while still making money!

Hope to see you when you come back to Atlanta.

Sincerely,
Mary Ann Galvin
American Fiber Systems

So true, and obviously AFS has got it right and they are benefiting from that. Thanks Mary Ann!

What the old school voice service providers, mainly the CLECs (define - news - alert), should be doing is interconnecting their core networks to peer their respective VoIP traffic and eliminate the costs of hopping on and off the PSTN between and amongst each other. This is beginning to happen, but many are focused on getting services to market for the buyers and don't see what may be an even easier first step to bottom line improvement.

Carrier VoIP peering is developing from within their own internal network core out to their network edge as well as from the network interconnection points (core of the edge's — carrier hotels) out to the customer premise. These are two different pieces of the carrier network moving in two different directions. One is for how the carriers' switches speak to each other and the other component is for how those switches speak to other carriers' and enterprise VoIP networks. The users are seeing, hearing, and reading about new services being created and enabled, but brought to different markets at different times and they are reacting to it as it comes to them. If it seems like it is a bit confusing that's because it is.

These are growing pains and ones that everyone will get through, but it seems that the enterprise buyers are in certain ways more prepared than the carrier sellers at this point. They know what they want, but just can't get it across every site yet. Plus, they don't make money from voice, so they have nothing to lose in an implementation and therefore are motivated. Across the board we all need more access from the premise to the edge to the core, awareness, and a little bit of interoperability (OK, maybe more than a little) and we should be on a fast track to an on-net voice nation! IT

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

By Ben Guderian

The Two Faces Of Dual-Mode Handsets

Dual-mode handsets — mobile telephones that work on both cellular and WiFi networks — are a hot topic in wireless telephony today. The proliferation of WiFi access, both in homes and public hotspots, has driven interest in using WiFi to augment or substitute for cellular networks. WiFi can't totally replace cellular, mostly because cellular technologies were designed specifically for covering large geographic areas while WiFi is more suited to in-building coverage or dense urban deployments. So don't plan on trading in your cell phone for a WiFi-only telephone just yet, unless you don't need to be accessible in the car or walking between hotspots. To really take advantage of what both cellular and WiFi technologies have to offer, you'll need a handset that supports both kinds of networks.

The idea of a dual-mode handset isn't new. Almost ten years ago, Ericsson offered a handset supporting both GSM (define - news - alert) cellular and DECT cordless technologies. BT offered their Onephone service, which tied their cellular service with DECT (define - news - alert), but it never took off partially due to the lack of support from handset manufacturers. So what's changed? Most importantly, WiFi is an IP-based packet data technology, so unlike DECT it can be used for broadband data applications like Internet access. Also, DECT didn't have the global availability and widespread industry support WiFi has today.

Several handset manufacturers are already shipping or have announced dual-mode products, and it's likely that others are on the drawing board or in development. Most of the dual-mode devices available today are high-end cell phones or PDAs with WiFi (define - news - alert)primarily for broadband data access, although some are incorporating WiFi specifically for VoIP applications. It isn't very complicated to add a WiFi radio module to a cell phone, although as with any WiFi client device, there are still the challenges of power consumption, quality of service, and security to deal with. The component manufacturers are coming up with chip sets with better integration of cellular and WiFi, which will reduce cost and improve battery life.

All in all, there's a lot of development activity, but what's really driving demand for dual-mode handsets? That depends on who you ask.

What Users Want

The opportunities and requirements for dual-mode handsets differ significantly between enterprise and consumer markets. Cellular phones have reached saturation in both of these markets, and cellular services look pretty much the same for both business and personal usage. WiFi adoption is growing in both enterprise and residential markets, also with similar applications, but with substantial differences in equipment cost and functionality.

ABI Research recently predicted that seven percent of mobile handsets shipping in 2009 will have WiFi capabilities. That might not sound like a high percentage, but it translates to an installed base of more than 50 million handsets. ABI expects that demand for dual-mode handsets will be driven primarily by enterprise users that need broadband access for data-intensive applications, and Wi-Fi is better suited for this than 3G cellular data access.

Another view comes from a recent consumer research report from In-Stat, which says that more than 80 percent of respondents were at least somewhat interested in the prospect of using a WiFi enabled cellular phone, with 50 percent either very interested or extremely interested in a dual-mode phone. The study identified better in-building coverage and unlimited calling plans as motivations for using WiFi at home.

So on one hand we have enterprise demand being driven by the need for speed (for data applications), while on the other hand we have consumers wanting better network access, and of course, saving money. Even though fundamentally it's the same device, applications and services differ significantly between consumer and enterprise markets.

Seven percent of mobile handsets shipping in 2009 will have WiFi capabilities.

Putting Dual-Mode To Work

In the enterprise context, a dual-mode handset doesn't make sense for every employee. Anyone who is not given a cell phone by their employer is not a candidate for a dual-mode handset. Depending on their job function, they either don't need a wireless phone at work, or they are better served by a WiFi-only handset for use in the workplace. Employees with company-furnished cell phones who are always on the road might not need WiFi access either, since they spend little or no time in company facilities, so a single-mode cell phone meets their needs. The ideal

Go To Table of Contents | Go To Ad Index

enterprise users for dual-mode handsets are the employees that are given cell phones because they travel on business or need to be contacted during off-hours, and they need to be mobile and still stay in touch when at work. Sure, they might be able to use a standard single-mode cell phone in the office (assuming cell coverage is good enough), but they won't be connected to the corporate telephone system so they can't access business features like conference and transfer, and they need to deal with two separate voicemail systems. And then there's the additional cellular airtime expense that might be racked up if employees are using cell phones as their primary business phones.

This is where a dual-mode handset really makes sense for the business user. The enterprise WiFi network gives them the coverage and reliability they might not get with the cellular network, plus the WiFi network provides access to the corporate telephone switch. Connecting back to the telephone switch could be as simple as using a VoIP protocol such as SIP (define - news - alert) for an end-to-end IP telephony connection, or it may require a VoIP gateway to interface with a TDM PBX or Centrex service. Either way, the dual-mode handset can operate like the wired business telephones with extension dialing and access to PBX features, and there's no usage or airtime expense since it's using the WiFi network.

Most of the utility of a dual-mode handset for an enterprise user comes from having one less device to carry around. Additional capabilities such as broadband Internet access over WiFi in the office, home, or at a hotspot, and extending corporate telephone switch feature access also add value.

Taking Dual-Mode Home

Demand for dual-mode handsets in the residential market is tied more to service offerings than to features. The challenges for implementing WiFi telephony in the enterprise are primarily voice quality, capacity, and feature access; whereas consumers are more interested in cost, ease of use and availability of service. As consumers have embraced cell phones, demand for traditional residential telephone service has diminished. But consumers still want reliable, fixed-cost telephone service at home. A dual-mode handset can meet that need and give wireless and wireline carriers the opportunity to offer new services.

For example, BT recently launched their BT Fusion service in the UK, which was previously known as Project Bluephone. BT Fusion is a service that brings together home cordless and public cellular in a dual-mode handset. The first handset available from BT uses Bluetooth for the home network, but BT has also thrown support behind using WiFi as well. Recognizing that consumers are abandoning traditional home landlines, carriers like BT can offer an all-in-one residential and cellular service plan that gives the consumer what they want — one phone that has the same telephone number whether at home or in the car.

The ability to seamlessly handoff calls between the cellular and WiFi networks is probably more important for consumers than for business users, mostly because consumers will expect the same experience no matter which network they are connected to, and ideally they won't even be aware of the fact that they use two very different kinds of wireless networks. Business users will have access to PBX features when using the enterprise WiFi network which may not be available through the cellular network, so it is likely that they will be aware of the different networks and not necessarily expect to be able to roam seamlessly back and forth. Nevertheless, there is a lot of work going into developing inter-network roaming solutions, and these will have implications on both the network and device implementations.

Bringing It All Together

It's a pretty safe bet that dual-mode handsets will proliferate in both enterprise and consumer markets. A few details still need to be worked out for dual-mode phones since WiFi telephony is still in its adolescence. Making it all work together — from both a technical and market perspective — will require a lot of cooperation between network operators, telecom infrastructure providers and handset developers. But both WiFi and cellular are here to stay, and bringing them together in one device will make mobile communication better for a lot of people in their work and personal lives. In the end however, dual-mode handsets won't be for everyone. There will still be WiFi-only solutions for workplace applications, and likewise, cellular-only handsets will probably never go away. **IT**

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

 

By Jack Jachner & Chris Vuillaume

The Advent Of Multimedia, Multi-Party Collaboration

In this second of a series of articles which address the Enterprise, we focus on the advent of collaboration, a key component of user-centric communication. The increasing number of geographically distributed teams working together, and the increasing range of available technology tools, has created interesting new business opportunities. Today, on-line collaboration is the fastest growing sector of communication in both numbers of users and numbers of minutes.

In essence, collaboration is a multimedia communication: users need to share, edit, and create information, and must communicate with other users simultaneously in order to do so, while being successful in their jobs. Numerous media are available (voice, video, text IM, and others), as well as many types of information to be shared. It is this very diversity that creates usability problems.

There are five dimensions of usability problems, which include:

- Variety of Devices (Blackberry, PDA, cell, desk phone, POTS, IP/POTS, PC…) and Network Access (security, QoS, policy, bandwidth), which cause connection issues;
- Variety of Media (voice, video, text IM…);
- Variety of Information (text, graphics, presentations, spreadsheet, computer desktops…);
- Variety of Actions (sharing, editing, creating, recording…);
- Variety of Interactions (scheduled, non-scheduled, ad hoc).

The essential element in successful collaboration is to make these usability problems go away. People don't want to go to meetings or use tools — they simply want to get their work done, and need to communicate with others to achieve this.

The technology of Presence is a key facilitator to enabling a clear path to communication and making sense of available means. Tracking down a colleague to discuss a presentation can be avoided using presence, if the interaction will be pointless — for example his calls are routed to a mobile phone, where no tools are available for collaborating.

People don't want to go to meetings or use tools — they simply want to get their work done.

To complicate matters, collaboration might be required at any time. If users are required to start up a special "meeting tool" each time to access this type of communication, collaboration will always be relegated to a niche, scheduled existence, within the enterprise workflow. It's unrealistic to rely on users to update their presence, location, or devices manually using a "meeting tool."

Two technology components are needed to resolve this dilemma: a simple and intuitive graphical user interface (GUI), and a signaling protocol that can transport rich presence information along with the communication and collaboration signaling. The "rich GUI" for communication must be available at all times, but not be intrusive. It must let users see quickly and simply what modes of communication and what multimedia tools are available to them and to other users, both before and once a communication is in session. Is my colleague available via IM or telephone? This GUI should be consistent across all user devices whether users are collaborating from their desk or in their home office.

The signaling protocol needs to be lightweight enough to scale up as densely as phone call signaling, in speed and volume, while being rich enough to handle a dynamically changing set of people and terminals. Fortunately, both problems have field-proven solutions.

Numerous user interface metaphors have been tried. The "meeting-table" (with avatar-like users sitting around a table), and a "tools" approach (which presents a dashboard of tools from which a user can choose) are among the most common options. However, both are poor choices. The idea of a "Web meeting" is flawed as most knowledge workers list meetings as the least productive part of their work. Consequently, the idea of reproducing a meeting room on-line is likely to fail. Also, most people do not think about what tools they need; they simply use them!

A promising user interface emerged in instant messaging (IM), already a popular way to communicate with others that made its way mainstream through various providers. The IM interface is a list of "buddies," or contacts, with an indication of their presence. This is a simple and natural way to implement all real-time communication. This user interface metaphor is immediately understandable to anyone who might already be using IM, and is easily expanded to richer media.

Users have a rich set of communication choices at their disposal: they can phone, exchange instant messages, participate in conferences, share applications, share video, and more. These choices are made clear in the GUI by the simple presentation of rich presence information. Communication and collaboration can take place without wondering how the participants will be involved, using which device, or where they are located. The user will focus on the active collaboration and not the tools enabling the communication.

Go To Table of Contents | Go To Ad Index

In a sense, IM becomes a sort of "interactive Caller ID", providing more then the just the identity of the user, making it possible to work with remote on-line colleagues, rather than just scheduling a Web meeting.

On the signaling front, the solution is provided by the IETF SIP standard. SIP (define - news - alert) was designed with the needs of multimedia multi-party sessions in mind. The use of SIP for telephony calls is a trivial case of collaboration. Similarly, the use of SIP for IM and presence signaling stemmed from the fact that communication/ collaboration has the greatest need for presence notifications.

SIP is naturally multi-party, multimedia, and mobility aware. Most importantly, it is an Internet-friendly protocol, which integrates with all the other IETF protocols. Consequently, a SIP-based infrastructure can ensure that collaboration applications no longer have a "niche" status, by integrating them naturally with other Internet user applications.

The business impact of collaboration is already apparent.

The business impact of collaboration is already apparent and will continue to grow over the next five years. For the enterprise user, it represents efficient real-time teamwork and a more effective use of high-value staff, and better responsiveness to customers, translating to a strong competitive advantage. For the IT manager, it means the integration of enterprise communication tools into a cohesive whole. For the vendor community, it represents a shift from stand-alone products to integrated collaboration solutions. IT

This editorial column series is a collaborative effort between Jack Jachner (Senior Director with the CTO office) and Chris Vuillaume (VP Product Marketing with enterprise products) at Alcatel. For more information, please visit http://www.alcatel.com.

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By Michael Marchioni

With Telephony Features Becoming Commodity, Applications = Profit

Since the FCC approval of the Carterphone decision in 1968, customer premises telephony equipment providers and resellers have earned billions of dollars in profits selling closed, proprietary telephone systems to small and medium sized businesses (SMB). Profitability was assured as the manufacturers and resellers controlled end user access to even the most basic telephony features.

Over time a "good old boys" network developed where the typical sales process went something like, "Look at my new whiz bang [*insert proprietary phone feature name here*] feature!!!! How many phones? How many lines? Press hard three copies. Let's go to lunch."

Of course this is an oversimplification, but the point is all phone systems provided the same basic functionality. The sale was won or lost based on the sales person's ability to sell their product, not on their specific understanding of the customer's needs. This model remained secure and basically unchallenged for thirty years. Regardless of the line size and application, selling basic voice features was a profitable business opportunity. Established manufactures and resellers in this space flourished.

Starting around 1998 everything began to change. First, inexpensive products manufactured in places like China and Taiwan finally overcame quality issues that had prevented their acceptability. As quality improved, these products began to steal market share from the incumbents. Second, Voice over Internet Protocol (VoIP) (define - news - alert) matured from a way for technogeeks to make free calls over the Internet into a legitimate business application. Thus the traditional providers no longer "owned" a closed network, but shared an open network with data applications. Finally, new open standards for voice communications such as SIP and H.323 emerged that promised to break the stranglehold the proprietary telephone systems had on the market. The promise of SIP and H.323 was to deliver the basic calling features most people used 95 percent of the time, that previously were only available in an expensive, proprietary solution through an open standard accessible by any developer.

There is a great opportunity in this market space to actually increase profits and recurring revenue opportunities.

Initially these changes were below the surface as the foundation was being built. Now in 2005 we've reached the point of acceleration. At risk is the survivability of the legacy solution providers and manufacturers.

Why?

Well for starters, there is more competition. In 1998 a certain company with zero percent market share in CPE voice equipment was making billions selling routers and switches. Today, over 20 percent of IP lines sold belong to them. That is one example of a data company stealing market share from a traditional voice provider. There are many more examples that we do not have time to get into here.

Second is the commoditization of the proprietary voice features and hardware. In the past, basic voice features such as conference, call forward, voice mail/ automated attendant, and call record were proprietary features that could only be had for a high per station price accessible only from a proprietary piece of hardware. Today, with an open standard such as SIP (define - news - alert), and the price competition at the low end of the market, these features are available to the end user at a much lower per station cost. This trend continues to accelerate as SIP-based applications are now embedded in products ranging from television sets to cell phones to laptop computers. Today a small business owner can literally create a phone system by downloading a SIP-based call control feature set from various developers, buy a PSTN (define - news - alert)gateway or IP trunk service, and purchase a few $50 phones and have access to all of the features mentioned above.

So it looks like doom and gloom for companies selling voice processing equipment to SMBs. But the picture is not quite as bleak as what I have painted above. While it's fact that the profits are slowly being squeezed out of commodity phone features and hardware, there is a great opportunity in this market space to actually increase profits and recurring revenue opportunities. Fact is, the market is growing in this space in both the number of business and total revenues.

We estimate price is the driving factor in about 80 percent of telephone systems sold to companies with fewer than 100 employees. These companies are looking for basic voice processing features at the lowest price possible. It is this segment of the market where profits are shrinking for manufactures and resellers alike. But what about the other 20 percent?

It is there that the opportunity lies. These 20 percent of

Go To Table of Contents | Go To Ad Index

companies are looking for something more than basic voice features. They are looking for help solving problems greater than replacing worn out telephone equipment. With convergence, the bridging of voice and data onto a single network, come integration issues and new application opportunities that extend to every area of the customer's business. In this new environment, to successfully sell terminal equipment and make a healthy profit requires first listening to the customer and understanding their business processes. Then, a company must have the skill set to integrate voice and data products from multiple vendors into a custom solution that solves the specific customer needs. Most importantly, the voice products they deploy must inherently support the generally accepted standards under the hood: SIP, H.323, VoIP, and TDM (define - news - alert). This new generation of voice products must be scalable, customizable, and configurable to operate seamlessly with data processing programs, off-the-shelf terminal devices and endpoints.

Customers are looking for a single source provider willing to take ownership of all the applications running on their converged network.

The successful technology provider's relationship with the customer only begins at the initial sale. Customers are looking for a single source provider willing to take ownership of all the applications running on their converged network. It's the companies that can step up today, address the customer's total need by leveraging the power of convergence, perform the integration services, and provide the long term application support that will survive and flourish.

For many companies in our industry the time has come to make a critical decision — change or die. **IT**

Michael Marchioni is director of product marketing for Iwatsu Voice Networks, a Founding Member of the Enterprise Communications Association (ECA). Marchioni maintains a seat on the Board of Directors. For more information on the ECA, please visit http://www.encomm.org.

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In our efforts to provide a wide variety of viewpoints, **INTERNET TELEPHONY**® *presents the following case study, showing an alternate approach to solving our readers' telecommunications challenges.*

The VoIP Alternative:

Experience The Gain Without The Pain

By Larry Jacobs

Ethernet services are the rage in business today. Enterprises are attracted to Ethernet access due to its low cost per megabit, high speeds, inherent multipoint connectivity, affordability of access equipment, and their overall familiarity with managing Ethernet switches. Service providers also see numerous benefits in Ethernet services including standards-based multivendor interoperability, low cost of deployment, and low provisioning costs.

The rapid growth in Ethernet services is sure to provide additional incentives for enterprises to use VoIP (define - news- alert). By converging all voice and data onto a common Ethernet services connection, multi-location enterprises will reduce costs and increase operating efficiencies.

One obvious way to do this is to replace traditional TDM-based PBXs with native IP telephony. This is clearly the path many companies are taking. Within the next 10 to 20 years, most PBXs will likely have been replaced or upgraded to support VoIP. However, today's installed base of TDM (define - news- alert) PBXs remains huge. Billions of dollars have been invested in this equipment, much of which is still highly reliable and functional.

Telecom/networking managers face a challenging dilemma: On the one hand, they would love to get the "gains" associated with VoIP — lower inter-location communication costs and the efficiencies of a converged WAN infrastructure. On the other hand, many are properly concerned about the "pain" frequently inherent in a move to VoIP — stranding their investment in existing PBX equipment, the high costs associated with the preparation and rollout and elevated risk of reliability and quality issues. This requires a difficult choice, but fortunately, standards activity has recently gained the critical mass needed to provide a real alternative.

Bridging Technologies

After several years in gestation, a new technology, called circuit emulation or pseudowires, has recently emerged that allows the convergence of TDM communications onto low-cost, Ethernet or IP infrastructures. In fact, Dr. Robert Metcalfe, the inventor of Ethernet, has identified TDM support as a critical attribute of carrier-class Ethernet. Pseudowires work by packetizing a TDM stream, transporting it across an Ethernet/IP network and then recreating the TDM stream at the other end. Importantly, the critical TDM timing must be regenerated with great accuracy — a challenging task over a packet network (Figure 1).

Standards bodies have approved a number of different pseudowire/circuit emulation variations and vendor solutions are available. They fall into two main categories: those that provide pure circuit emulation (without regard to the type of voice or data being transported) and those that add specialized voice processing and protocol optimization to gain additional efficiencies for voice and data transport. RAD Data Communications uses an approach called TDMoIP that works in both modes. Similar solutions are available from other vendors as well.

In addition to lowering communications costs, use of a TDMoIP gateway brings the following benefits:

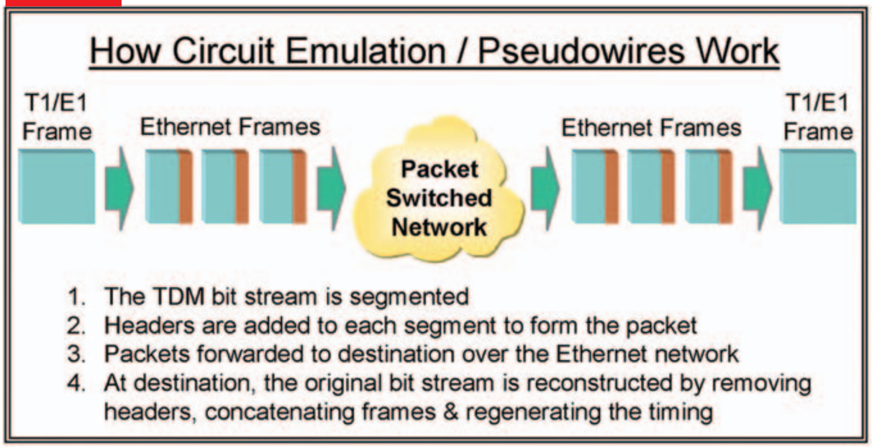
- **Investment Protection**: By interfacing with standard T1 ports, TDMoIP requires no need to change anything on PBX systems. As far as the PBX knows, it is connected to a normal T1 circuit.
- **Transparency**: TDMoIP supports any PBX protocol, whether standard or proprietary, and permits transparent communication across the network. Enterprises can retain all the features and functions of existing PBX networks, e.g., message waiting indications, busy-lamp fields, etc.
- **Efficiency**: TDMoIP provides greater efficiency and higher quality of service than VoIP. VoIP functions as a "switched virtual circuit" in that each packet sent by a VoIP system has its own address. On the other hand, circuit emulation or pseudowire technology functions more like a "permanent virtu-

Go To Table of Contents | Go To Ad Index

"VoIP is a promising technology which we may acquire in the future."

– Rob Carley

Figure 1.



al circuit." Packets always traveling from point A to point B are aggregated into a bundle and sent as a group to the same destination, such as a branch office. This results in TDMoIP being up to 60 percent more efficient than VoIP. TDMoIP also provides higher quality of service since each bundle contains fewer samples of any one conversation making it more resilient to packet loss.

• Voice Compression: Using the same codecs as in VoIP, TDMoIP compresses voice streams for maximum efficiency. Combined with the bundling efficiencies as just discussed, TDMoIP can achieve a better than 16:1 compression ratio for voice calls. For example, one can transmit the full 24 voice channels of a T1 line over as little as 128K of Ethernet bandwidth. Alternatively, the equivalent of 16 T1s can be transmitted over a single T1.

• Security: TDMoIP is more secure because it is of little interest to hackers and because encryption protocols can be applied more easily than with VoIP.

Pseudowire/circuit emulation technology is an alternative to VoIP that lets companies take advantage of a converged packet service for both voice and data transmission while protecting their investment in existing communications equipment and ensuring uninterrupted support of existing mission-critical services.

Applications: TDMoIP Gateways in Practice

Austin Independent School District, which serves about 77,000 students, decided to replace its ATM communications network with a pure IP environment designed to link all of its schools through a metropolitan area network. Rob Corley, the District's Network Analyst, explains, "When our ATM equipment was beginning to reach the end of its life, we conducted substantial research and decided that our needs would be best met with WAN technology that was for the most part pure IP. After researching various options, we identified TDM over IP gateway technology as the most cost-effective, scalable solution for transporting T1 circuits between PBXs at each of our campuses."

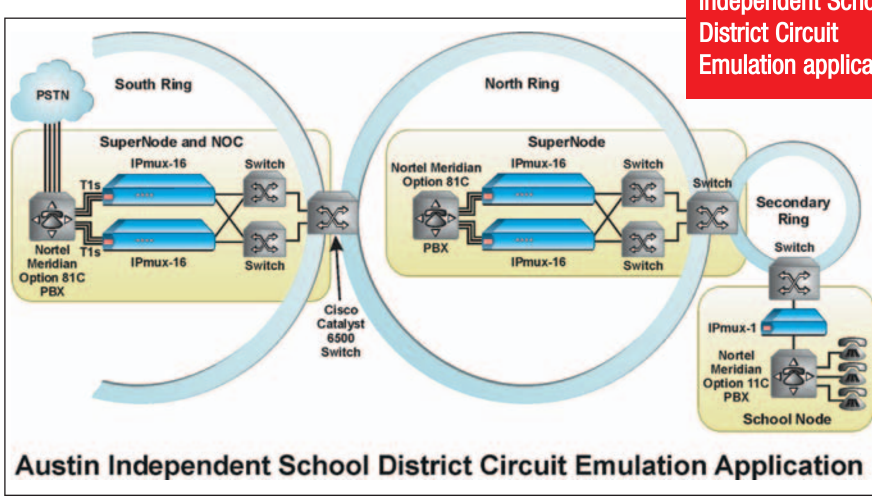
About two years ago, the District deployed TDMoIP gateways at 13 super-node locations to provide tie trunks between their major PBXs. "We have a tree and node structure with a main PBX at the Network Operations Center, distribution PBXs at five sites and access PBXs at each campus," Corley explains. Plans include rolling out the solution to additional locations as well (Figure 2).

Why not just switch to VoIP? "While VoIP is a promising technology which we may acquire in the future," says Corley, "it was not a near-term solution for us because we had a very large investment in traditional PBXs at each of our 125 sites." John Kohlmorgan, the District's WAN Manager, agrees. "The ability to transport the T1 voice circuits over IP allows the District to leverage its current investment in the existing PBX hardware," he says.

"We were hoping to achieve the same functionality with IP as we did with ATM, but have now surpassed that," states Corley. "We've had no failures in the two years since we acquired the technology. In fact, our PBXs appear to work better with this equipment than they did with the ATM circuits. If a fiber is cut between campuses, for example, the PBX is not affected, whereas before, there used to be a little 'hiccup.' The technology has proven very reliable," he adds.

Another example of success comes from

Figure 2. Austin Independent School District Circuit Emulation application.



TeleperformanceUSA, a contact center solutions provider based in Salt Lake City, UT. The company uses RAD's Vmux circuit emulation/ pseudowire solution enhanced with voice optimization technology to slash their cost of transporting hundreds of simultaneous calls to their remote call center locations in Manila and Buenos Aires. With the cost of a single International Private Line E1 circuit between PBX/ACDs easily costing $8,000 per month or more, TeleperformanceUSA had great incentive to use state-of-the-art technology to minimize their costs. While using end-to-end VoIP could have helped reduce their operational costs, TeleperformanceUSA realized greater savings by using circuit emulation and voice compression. Fortunately, the company avoided the large capital outlay that would have been required to upgrade or change out their PBX/ACD systems.

"RAD's Vmux is ideal for customers like Teleperformance USA," explained Michael Schmidlen, President of Advanced Datacom Solutions, a RAD VAR based in Highlands Ranch, Colorado who helped TeleperformanceUSA design the solution. "RAD is leading the industry with 16:1 compression without sacrificing voice quality. The Vmux also provides a robust set of backup and failover options to ensure critical network uptime," he continued.

Evolution

Ethernet has been one of the most successful technologies in history in terms of its speed and breadth of acceptance. It is used universally today for local area networks and is well on its way to global ubiquity for WAN application. While VoIP may be the way of the future, pseudowire/circuit emulation technology is today's solution for enterprises seeking to evolve in the direction of converged packet transport — without the expense of replacing perfectly functional PBX equipment. It's an evolutionary step that brings the best of both worlds: the quality of service and rich feature set of TDM technology combined with the cost-effectiveness and efficiencies of Ethernet/IP backbones. Who says you can't get the gain without the pain? IT

Larry Jacobs is vice president of marketing of RAD Data Communications, Inc. For more information on this company, please visit http://www.radusa.com.

TMC Labs Internet Telephony Innovation Awards 2005: Part II

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

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

TMC Labs proudly bestows 25 Innovation Awards, published in two parts in order to accommodate our in-depth write-ups for the winners. Last month we ran descriptions from AnchorPoint to Go2Call. This month we begin with Inter-Tel and conclude with Zultys.

2005 TMC Labs Innovation Award Winners (Full List)

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

**companies appearing in italics appeared in our July 2005 issue with a full description.*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TMC LABS INNOVATION 2005 AWARD

Inter-Tel, Incorporated

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

The Inter-Tel (news- alert) 5000 Network Communications solutions delivers VoIP communications, including connectivity, presence, collaboration, messaging, and other rich IP applications, including desktop video, and targets small- and mid-size companies. One innovative aspect of the Inter-Tel 5000 system is that it is perhaps the first IP-PBX system to utilize industrial-strength Compact Flash (CF) technology in its call processing units. This significantly increases reliability by eliminating spinning disks and other movable parts from the core system. The Inter-Tel 5000 is also unique in that it allows customers to leverage their existing digital endpoints, further enhancing customer value. Through the use of optional expansion units, customers can add up to 96 digital and analog endpoints, while still leveraging the benefits of VoIP.

Its presence management is one of its most powerful attributes. One of Inter-Tel's presence tools is their Unified Communicator software, a communications management tool with powerful features such as routing rules, presence management options, personalized call handling, customized greeting, call history, consolidated contacts, and more. Another presence application is Connection Assistant, a workgroup tool that provides enhanced call handling, flexible programming of extensions, and screen-pops of frequently used applications. Another interesting presence tool is Telephony Manager, a Microsoft CRM tool integrated with Inter-Tel's voice platforms via the OAI interface that allows you to initiate, manage, and track telephone conversations directly from Microsoft CRM; view contact and account information; and share information across departments.

Another unique feature of the Inter-Tel 5000 is its WAN failover solution to ensure continuity in the event of WAN or LAN disruption. In the event of a failure, the Inter-Tel 5000 still delivers its full feature set to users, in comparison to most competing products, which are reduced to just delivering dial tone. The 5000 is compatible with Inter-Tel's Axxess platform and scalability is very good — up to 63 systems can be transparently networked. Finally, the Inter-Tel 5000 supports several industry standards including SIP, MGCP, 802.11b, 802.3af (Power over Ethernet), TAPI, CT Connect, and more.

LignUp Corporation

LignUp Communications Solution
http://www.lignup.com

Voice is becoming commoditized and the price of dial-tone is rapidly approaching "free" with the help of companies such as Vonage (news - alert) and Skype (news- alert) leading the way. To survive, service providers must offer differentiated services and evolve from telephony providers to telephony integrated Application Service Providers (ASPs). LignUp (news- alert)enables service providers to offer hosted telephony services for small, medium, and distributed enterprises, as well as offer converged application services that integrate voice and presence into business applications and accelerate business processes.

LignUp Corporation delivers a powerful Web services-based VoIP communications platform deployed with hosted IP-PBX, IP-Centrex, Voice Mail, Unified Messaging, and IVR Web applications. The platform supports industry standard VoiceXML, but it also supports LignUp's proprietary Media Control XML (MCTRL), and LignUp Call Control XML (CCTRL) languages. These are more powerful XML-based languages that Lignup has developed since the VoiceXML spec is still immature and incomplete. Using MCTRL or CCTRL developers can create custom voice solutions, voice-enable enterprise applications, voice-accelerate business processes, and more. LignUp Web Services enable Web developers to easily access powerful call control and media processing capabilities via XML. In fact, LignUp's hosted PBX, VoiceMail, Unified Messaging, and other applications have been built using LignUp Web Services.

Essentially, LignUp allows service providers to launch and operate VoIP services with lower capital expenses, as well as allowing these players to scale operations as the business scales. One key aspect of the LignUp solution is that it offers a comprehensive solution as opposed to their competitors which only offer pieces of the solution. This eliminates the costs to integrate solutions from a variety of vendors.

Another key advantage of the LignUp solution is that it is uses open standards including "native" SIP with proven interoperability with a variety of SIP devices. Also, since it supports SIP, the solution can connect directly to VoIP networks such as Level3. Scalability and reliability are excellent with their modular, clustered architecture. Each LignUp component — LignUp Call Directors for call control, LignUp Media Servers for media processing, LignUp Web Applications (Hosted PBX, VoiceMail, Unified Messaging, IVR and others) — can be deployed on a single server or distributed on separate servers. In addition, each component can be clustered separately for more flexible deployments. Redundancy is implemented simply by deploying two of each component. If any one server fails, the other continues providing the function.

The traditional process of telephony application development would require the work of scarce and expensive CTI developers, working on proprietary or platform specific TAPI, or other APIs. Applications would have to be customized to each platform and only one or two significant applications would be able to get to market per year. According to Lignup, "These challenges raise significant barriers to entry for most aspiring service providers, and made even large providers (all of whom are margin sensitive) tentative to embark on fully pursuing the development of high-value applications." Furthermore Lignup stat-

Go To Table of Contents | Go To Ad Index

Getting products ready for market is not easy. To do it right, you need a Q/A staff with up-to-date skills and the latest test equipment. That's where CT Labs comes in. A respected testing authority, CT Labs specializes in real-world testing, test automation, and Q/A services for a wide range of converged communications products.

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ed, "Developers do not need to understand the details of SIP, and can use familiar Web development tools such as IBM Websphere, BEA Weblogic, MSN.net, JSP, ASP, Perl, CGI, etc. This enables service providers to deliver dozens of innovative applications per month." TMC Labs was impressed with the comprehensive LignUp Communications Platform, which allows for rapid application development and deployment thus enabling service providers to quickly incur incoming revenue.

Millenigence, Inc.

DashPhone CXP

http://www.millenigence.com

DashPhone CXP is a DAS module that makes applications written for Cisco 7940/7960/7970 series of IP phones available on any IP phone and IP endpoint that is supported by DAS. CXP can sit between any Cisco XML application server and any supported IP phone or IP endpoint, and deliver the application transparently to the device. Millenigence (news- alert) claims that DashPhone CXP is the first and only product available in the market that delivers Cisco XML IP phone applications to non-Cisco IP phones.

Using CXP, various phones such as Avaya 4620 or Siemens optiPoint 600 can take advantage of a host of Cisco XML applications that are already developed by software vendors for Cisco IP phones. Since Cisco was the first company to popularize a data interface for IP phones, many more applications are available for IP phones made by Cisco compared with any other vendor. CXP makes these applications available on other IP endpoints.

DashPhone CXP allows data-centric applications and services written for Cisco IP phones to not only run transparently on non-Cisco IP phones, but also PDAs, and Wi-Fi or WAP enabled cell phones with zero programming. An employee can use his or her IP phone, PDA, or cell phone as a terminal to interact with the same enterprise data.

Essentially, CXP is middleware that sits between a Cisco application server and non-Cisco phones to translate the XML pages intended for a Cisco IP phone into whatever format the non-Cisco phone is able to understand, display, and respond to. Dashboard CXP is fully J2EE based and therefore capable of running on virtually any operating system and server environment.

Using DashPhone CXP, popular Cisco XML applications (which was just intended for Cisco gear) can now be run on a vast range of IP phones. Developers can easily use Cisco XML for writing their applications and need not worry about the target device.

Thus, if you are developing or have developed Cisco specific applications and services for the 7940 and 7960 IP phones, CXP allows your applications to reach new markets unmodified, without any re-coding. Further, with CXP, you can even run and test the application on your PC browser, making development much more convenient.

Nortel

BCM50

http://www.nortel.com

When you think of Nortel (news- alert) you think of "big iron" PBXs: reliable and extensible through many third-party applications, targeting medium-to-large businesses — but you certainly don't associate Nortel with the SOHO (small office/home office) or SMB market, now do you? Well, times have changed. Nortel is targeting their new Business Communications Manager 50 (BCM50) product directly at small single sites, franchises, or branch offices. The Business Communications Manager 50 platform, which just launched in May, is designed for 20 stations with room to grow to 40+ stations. This 20–40 station size range is certainly a new benchmark for Nortel. We bet if you called Nortel five years ago for a 20–40 station phone system they'd probably laugh at you. Unlike larger businesses which tend to stay with their existing phone system hardware for a 10–20 year lifecycle, small-to-medium businesses are always looking for a competitive advantage and tend to swap out their phone system hardware much more quickly. Further, millions of new small businesses are created each year requiring new phone systems to be purchased.

Well, the Business Communications Manager 50 platform is ideal for businesses that require advanced capabilities, such as robust telephony features, voice messaging and unified messaging, IP networking, Internet/intranet access, skills-based routing, IP telephony to users' desktops, and an integrated router option for Ethernet or ADSL broadband access.

According to Nortel, "Our goal is to make 'convergence' affordable and available to the smallest business sites." They also stated, "Through open standards and an "evergreen" development strategy, Business Communications Manager 50 platforms fit well in hybrid environments that contain a mix of analog, digital, or IP services. And since it interworks with other Nortel key/PBX systems, larger Business Communications Manager systems, and with our portfolio of convergence call servers, you have a smooth migration path."

Nortel told us, "The Nortel Business Communications Manager 50 system combines the best elements of high-end digital PBX phone systems, cutting-edge convergence solutions and robust data networking in one affordable package. By integrating advanced data networking and comprehensive telephony features in a single device, Business Communications Manager 50 delivers a level of system integration and flexibility rarely seen in the industry — and certainly uncommon for small business locations."

Installation and configuration are very easy to do using their included software, which is important in small-to-medium businesses that often do not have their own dedicated telecom department and instead rely on their IT staff to configure various phone system parameters. One innovative aspect of the system, which

Nortel has been doing for years in their other phone products, is that you can program functions using any connected telephone set. One final innovative feature of note is that the BCM50 sports more than 200 telephony features which you usually only see in high-end PBXs.

Pandora Networks

Worksmart

http://www.pandoranetworks.com

We all know the story of Pandora's (news- alert) Box which unleashed evil upon the world. Well, Pandora Network's appliance "box" unleashes a converged platform that includes IP Centrex, ACD, Web Contact Center, video, Instant Messaging services (AOL, Yahoo, MSN compatible), and collaboration. We certainly like their corporate name but we took pause when reading their application for this award when we read that the product name was called Worksmart. A converged VoIP platform was certainly not the first thing to come to mind when we read "Worksmart." Our 'critiquing' of their product name aside, this is one very impressive product.

First off, Worksmart is a plug-and-play IP-enabled enterprise communications appliance that features IP telephony, call center functionality, and more. One obvious benefit of this converged solution is that instead of purchasing and integrating multiple products from many vendors, Pandora Networks packs them all into an easy to use and operate appliance. Using a single Worksmart appliance means you don't have to purchase an IP-PBX, video conferencing server, instant messaging server, collaboration server, and an online Web contact server separately. It connects to either your traditional carrier or to Pandora's own SIP-powered IP telephone network that allows inbound and outbound phone service. One unique aspect of this solution is that unlike IP Centrex services, Pandora can deploy their on-demand services from their Central Offices or allow the customer to operate it on their premise and to blend with TDM interfaces when desired.

Worksmart enables your employees to engage in voice, video, and collaborative conversations from anywhere, using one simple desktop application. According to Pandora, cost savings are up to 80 percent less than single-point solutions. Pandora Networks stated, "There's no complex administration, no third-party hardware integration, no reconfiguring your network, no weeks lost on integration and training — just plug it in and turn it on. The powerful Web-based management interface allows you to start working in just a few minutes. Users will love having just one desktop application that provides all of their communication needs and allows them to use any phone as an extension — including a cell phone."

RingCentral

RingCentral

http://www.ringcentral.com

RingCentral (news- alert) provides customers with a personal toll-free and/or local area telephone number with an integrated auto-attendant that can screen and forward calls and take messages and faxes — all with no customer hardware or software required. When callers phone or fax a user's RingCentral number, RingCentral automatically handles the call routing. You have the option to route the call to a single number or simultaneously to multiple numbers where you can be reached, or you can send the call to voice mail. You can also define schedule preferences for when certain rules are in effect, such as only dial your cell phone during the evening.

If you are online, a pop-up screen notifies you of the call that includes caller audio preview and CallerID. Users of RingCentral can screen who is calling and see their caller ID before deciding whether to accept the call, send it to voice mail, or reject it outright. They can also direct the call to any phone including their cell phone, land line or VoIP phone or they can direct the call to voice mail. For missed or ignored calls, RingCentral can notify users of their messages by phone, pager, Web, or e-mail.

RingCentral also includes full-featured, computer-based faxing capability. RingCentral delivers incoming faxes as e-mail attachments. It can also store them on RingCentral's servers for Web-based access. Users can also send faxes from within any Windows application.

Using this service you have the capability to manage messages, faxes, and call records over the Web, and over the phone, and on a PC using their software. The software includes detailed logs of all incoming and outgoing calls, and it presents calls, messages, and faxes in a Web-based user portal. The service comes with a unique Windows Call Controller application that can be used to manage calls, messages, and call logs locally on a user's PC. There's also an online address book that integrates with Outlook and Outlook Express and can be used for caller identification and easy dialing.

A business plan offered by RingCentral grants you a virtual PBX with auto attendant and extensions for individual employees, who get personalized dashboards to manage inbound calls. Each extension can be programmed with its individual call-forwarding rule. RingCentral also supports direct dial and direct fax numbers, making it possible for a person or department to be reached through both the main company number via an extension, as well as by a direct phone number.

One unique feature of RingCentral's Call Controller is the option of typing a quick message that goes through a text-to-speech engine; the message is immediately played back to the caller as a customized greeting, composed on-the-fly. This is useful when on another line or unable to take a call. The custom message that users compose can be based on their availability and the CallerID of the caller, for example: "Hi, John, I'm on an important call, but I'll call you back in 15 minutes." One final innovative feature is the integrated click-to-call RingOut function. This feature allows you to quickly call-back numbers stored in call logs, voice mail, and directories.

SER Solutions, Inc.

TSP500 Outbound Dialer

http://www.ser.com

SER's (news- alert) TSP500 is not your ordinary predictive dialing switch. Sure, it has full compliance with all state and federal do not call regulations. Sure, it has excellent scalability supporting up to 384 agents and sure, it leverages SER's SmartPace VI dialing algorithm, which SER claims generates more connects as compared to other predictive dialers. But the real innovation is that the TSP500 enables contact centers to consolidate their operations through the use of integrated VoIP. Their VoIP support allows remote agents located either in the U.S. or abroad to connect to the predictive dialer switch using H.323 or SIP. The switch also delivers enhanced Caller ID functionality that allows contact center operators to display both the telephone number and designated name of the caller or client for effective campaign management and compliance with FCC and FTC mandates.

The TSP500 reduces total ownership and communications costs by allowing contact centers to utilize VoIP to distribute information, agents, and technology across multiple centers. According to SER, "We believe the TSP500 is the first predictive dialing switch that can seamlessly scale to 384 agents, provide native support for VoIP connectivity, enhanced Caller ID capabilities, and generate more connects than other predictive dialers-all on a single system."

The TSP500 ensures peak agent efficiency by leveraging SER's SmartPace VI dialing algorithm which maximizes live connects while eliminating unwanted busy signals, answering machines, fax machines, and ring no answers. SER's voice recognition detects a human voice within milliseconds, as well as busy signals, ring no answers, and telephone company SIT tri-tone intercepts. This feature boasts productivity and ensures a natural call flow by allowing agents to hear the first "hello" instead of a silent pause.

Since the TSP500 supports the industry standard SIP protocol and is interoperable with several SIP proxy solutions, the TSP easily supports work at home agents through VoIP and SIP without requiring the agents to use a soft phone or IP phone solution. Remote sites can utilize a VoIP gateway or ATA (analog telephony adaptor), which converts the IP audio stream back to an analog TDM stream, with agents connected through

Go To Table of Contents | Go To Ad Index

simple analog phones. The gateways can communicate with each other using either the H.323 or the SIP protocol. In this configuration, as long as the gateways at the remote site have power, the conversations will not be interrupted, even if the desktops lose power.

The other possible configuration is to connect the TSP500 directly to either IP phones or an IP soft phone on the agent's desktop computer using SIP. While this can be used when the remote agents are located together, this provides the greatest value with agents that are dispersed, because no remote gateway is required. This configuration works with several commercially available off-the-shelf SIP proxy servers.

Toshiba America Information Systems, Digital Solutions Division

Toshiba Strata CIX

http://www.toshiba.com

Toshiba (news- alert) is well known for its relatively inexpensive, feature-rich phone systems targeting the SMB market. Toshiba's Strata CIX is no exception as it is designed for small- to medium-sized enterprises or larger corporate users with multiple sites with support up to 672 ports. It can be run as a pure IP system or can be TDM enabled, allowing enterprises to migrate to IP at their own pace.

One of the most innovative aspects of the Toshiba CIX is its support for several advanced applications on a single system utilizing the embedded Strata Media Application Server (MAS). MAS is one of the first devices to combine voice applications from multiple vendors onto a single platform using Intel's Host-based Media Processing (HMP) technology. This lowers the entry cost to many applications for the small to medium enterprises because it eliminates the need for multiple hardware platforms to support each application separately. MAS applications include Auto Attendant, Voice Mail, Automated Speech Recognition, Text to Speech, Unified Messaging, Interactive Voice Response, Automatic Call Distribution and Reporting, Web-based Personal and System Administration, Web-based Telephone Applications, ACD/MIS, and other third-party applications. If there is an application not included, you can build it yourself using an easy to use script editor that works with MAS to interpret code, process functions, follow custom routing, and more.

Strata CIX supports the SIP standard and it has been designed to deliver virtually every feature to every user, regardless of the type of device they are using, whether they are static or mobile. The system supports IP phones, IP wireless handsets, both analog and digital telephones, IP softphones on laptops, and tablet PCs. Of course, Toshiba hasn't forgotten their TDM roots so they support both pure IP and TDM. This allows users to choose how they will maximize their systems and migrate existing equipment. One final feature of note is Toshiba's new personal administration tool, "My Phone Manager," which enables individual users to easily program the LCD on the telephone and program speed dial buttons and feature buttons via their PC's Web browser. This helps alleviate administrator support.

Tripp Lite

SmartOnline Expandable Rack/Tower UPS System

http://www.Tripp Lite.com

Released back in November, Tripp Lite's (news- alert) SmartOnline Expandable Rack/Tower UPS battery protection system is perfectly suited for mission-critical VoIP equipment. In fact, this is no ordinary battery-backup system. This product is the first to be certified by Cisco Systems as capable of shutting down connected Call Manager servers in VoIP applications.

SmartOnline features a compact tower/rackmount housing and protects systems from the economic damage associated with the disruption and downtime caused by power blackouts, voltage fluctuations, and transient surges. Tripp Lite claims a continuous sine wave output with zero transfer time. In the event of extreme power events, the unit transfers the equipment load to battery power until stable power is restored. If the power disruption outlasts the system's battery (configurable) capacity, connected applications and operating systems are signaled to shutdown in an orderly manner before all power is lost. This includes the popular Cisco Call Manager VoIP application.

This product offers a unique combination of runtime scalability, simple plug-in installation, outlet customization, and hot-swap serviceability for both power electronics and battery systems. The battery system and receptacle panels are modular, making this unit ideal for later expansion as budgets or application-related hardware needs evolve. In the event of a maintenance need, systems can remain connected and available during service, ensuring application uptime.

Another distinctive feature is the informative alphanumeric LCD front-panel display. Even many network managers need assistance in understanding their operating conditions. With power being such a critical component, there should be no doubt. The LCD display greatly enhances all users' understanding of operating conditions, unlike traditional LED indicators. Finally, Tripp Lite backs up their products with $250,000 worth of Ultimate Lifetime Insurance.

VegaStream

Vega 400

http://www.vegastream.com

The Vegastream (news- alert)400 supports both SIP and H.323, has very good scalability and most importantly it has excellent voice quality with minimal latency. TMC Labs knows this since we recently tested the Vega 400 in the labs. The Vega 400 supports four E1/T1 interfaces and can be configured to support one to 120 VoIP channels (four E1s).

The Vega 400 has a very distinguishing feature that differentiates it from many of its competitors. That is, many competing gateways can also do 120 channels, but only using the G.711

codec and not other codecs. On the other hand, the Vega 400 can do 120 channels using G.711 but also 120 channels using, say G.729 or G.723.1. Due to the DSP or processor requirements many competing VoIP gateways can only do, for example, 90 out of 120 channels using the G.729 codec. Similarly, it supports T.38 faxing and it supports the full 30/24 channels per E1/T1 of fax simultaneously where as some other gateways only do a few.

Another unique feature is that the DSPs are on PCMCIA cards, which slide in the back of the chassis. Upgrades (i.e., from one T1 to four T1s) can be done without opening the unit. This flexibility allows you to start with a low-density gateway at a low cost, and add capacity as needed. One final feature of note is that when you first boot the Vega 400 you can select either a SIP or H.323 stack, which is a really nice feature. Many other solutions require that you manually download a separate firmware to install a different VoIP stack.

Witness Systems

eQuality ContactStore for IP Solution
http://www.witness.com

ContactStore for IP captures voice conversations and corresponding computer desktop screen activities in VoIP environments. Providing converged voice/data networks with a sophisticated and robust feature set, it includes the functionality of ContactStore, and enables large organizations and small- to medium-sized businesses (SMBs) to record VoIP calls while simultaneously meeting industry compliance requirements.

VoIP (define - news- alert)is fast becoming a mainstream technology for contact centers. Further, because of the shift toward remote agents and virtual contact centers and the move to centralizing systems like call recording, the ability to record VoIP calls becomes imperative. Witness Systems' ContactStore for IP solution utilizes the Cisco IP telephony infrastructure to capture valuable business intelligence — in the form of both customer and competitive insight — by simply recording, evaluating, and analyzing customer interactions.

Witness Systems (news- alert) claims that ContactStore for IP was the first "software only" solution introduced to the market and they claim more VoIP recording deployments (650) than any competitor. ContactStore for IP leverages a Web-based architecture that scales from a single seat system to a distributed multi-site enterprise with thousands of channels, providing a single view of all customer contacts.

The product features the ability to "record all calls" as well as "record on demand." One unique feature is that it performs stereo recording to optimize the clarity of recorded speech and to aid in automated conversation analysis. This enables both sides to be recorded separately, which is important for dispute management. Also, word spotting and speaking authentication products can be used to only analyze either the caller, the agent, or both depending on your needs.

Each of the recorded interactions are assigned contact attributes, making it easy to retrieve, review, report, and analyze customer experiences for compliance requirements and quality purposes. Another interesting feature is ContactStore IP E-mail Call, which lets you e-mail the entire interaction at any point during call. The system can also be set up to trigger a recording based upon a CTI event. Another nice feature is that you can press the record button on the IP phone at any point during the call. An agent can capture and save the entire interaction and not just the portion from where the agent pushed the button. Also, you can start/stop and pause a recording to omit confidential information such as security passwords.

Xten Networks, Inc.

Xten Pocket PC SIP Softphone

http://www.xten.com

Xten (news- alert) is well renowned for their SIP softphone clients, which typically have excellent voice quality with good codec support, and a minimal software footprint. Xten's Pocket PC SIP Softphone is no exception. It supports open standards, including SIP obviously, but also optional G.729a, wideband codecs, AEC, VAD, and a superior audio mixer provide for excellent call quality and user comfort. According to Xten, "This product can be combined with open source open standards PBX offerings not unlike Asterisk to offer a wireless PBX solution. Compliant with Sylantro and Broadsoft this software is a good fit for IP Centrex as well." Everyone talks about when Skype will be embedded onto smart phones. Well, Xten's Pocket PC SIP softphone client today can run on the new Windows Mobile 2005 operating system allowing you to connect to any SIP-based ITSP or SIP gateway.

Xten has over 1 million endpoints deployed and that number is growing rapidly. They also offer training for those who implement their SDK. Xten told us, "We listen to our customers and closely monitor the market, we have people in the IETF that monitor industry movement and give us guidance as to where we should be focusing our efforts, this really helps us get a head start on any perceived competition."

The Pocket PC-based softphone supports Automated Echo Cancellation (AEC) thus negating the need for a headset. You can simply use your Pocket PC device just like a phone. The softphone client supports all the various call controls including transfer, blind and supervised, as well as three-way conferencing.

Zultys Technologies

MX30 Enterprise Media Exchange

http://www.zultys.com

Zultsys (news- alert) is an interesting affordable "converged" solution targeting the SMB (up to 30 users). The MX30 integrates voice, data, video, and fax, and provides the functions of an IP PBX, firewall, NAT, and VPN — a complete solution for a small business or branch office, all in a single CPE appliance. The MX30 is designed specifically to connect businesses to Internet Telephony Service Providers (ITSPs) using SIP. The MX30 utilizes any broadband connection for its trunking to receive and route all calls to destinations external to the company. This allows customers to realize the full benefits of VoIP and save money through the lower rates offered by ITSPs, such as Level3, Deltathree, etc. Analog and ISDN basic rate connections are only provided as a backup in case the ITSP connection fails.

According to Zultys, "The MX30 follows the MX1200 and MX250 in providing a completely open standards platform for integrating the functions of an IP PBX, Internet gateway, network server, and application server. The MX30 is not a "stripped down" version of its larger siblings, but contains all of the features and functionality a smaller business will need in order to deploy a modern VoIP communications system."

The MX30 is capable of networking with multiple MX systems over a WAN. This provides toll bypass savings and allows a small business to preserve its investment as it expands. Zultys' MXgroup software allows companies to network all sites, creating a single communications platform, enabling telephony, video, fax, voice mail, instant messaging, and presence, regardless of location.

The MX30 is unique in combining the features of an IP-PBX, fax server, application server, VPN, as well as connections to ITSPs, all in a single box. Further, their target market of 30 seats and under brings a new level of advanced VoIP functionality to the SMB market. Finally, every Zultys product including the MX30 is based entirely on open standards, such as SIP and VXML ensuring you are not locked into a single vendor when it comes to IP phones or VXML applications such as IVR, voice mail, etc. IT

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Evaluating IP Contact Center Solutions:

The "Essentials"

Increasing demand for 24x7x365 service, workforce globalization, open standards, and rapid technological innovation have all converged to set the stage for the IP contact center. And while IP solutions offer great potential when it comes to extending the contact center across the enterprise — linking remote agents, integrating distributed call centers, and providing architectural flexibility — the marketing hype surrounding the advent of VoIP *(No cost communications! ROI in weeks!)* can set organizations up for disappointment if they jump in head first without truly considering the practical implications. A well thought out adoption strategy is essential in order to make the most of existing investments, mitigate risk, and ensure that your organization leverages VoIP successfully. How should today's companies go about making the move to VoIP in their contact centers? And what are the five "essentials" that a truly effective IP solution should have?

Migrating To VoIP

Build a strong team. At industry conferences and events, I hear over and over again that one of the leading issues organizations face in migrating their contact centers to VoIP (define - news - alert) is a lack of alignment between the business managers running the contact center and the IT experts that need to make the technology work. This partly springs from the fact that, in the past, the contact center has operated as its own island — separate from the IT underpinnings of the enterprise as a whole. But in a converged world, this is changing rapidly. If the IP contact center is to succeed, business and IT managers need to share the responsibility, by exchanging information early and often about goals, expectations, and how resources and budget will be allocated. From the contact center leader's perspective, this means explicitly specifying requirements related to business rules, reporting, service level agreements, and other key areas. And because downtime is such a critical issue for the contact center, managers also need to provide accurate information on call volumes and flow so that the IT team can make sure there is adequate bandwidth to support IP initiatives. IT must also clearly understand the demands VoIP will put on the corporate network, as well as be prepared to address issues like quality of service, security, piloting, and more.

Plan deployment models. Is your organization planning to expand the reach of its contact center to remote sites? Do you want to network branch offices together to create a "virtual" contact center? Are you interested in leveraging at home agents? All of these IP-enabled scenarios require some planning related to quality of service, security and application choice. For example, connecting remote agents to the main contact center site is relatively simple, but quality of service is a consideration if agents are connected via broadband rather than through a VPN (define - news -alert), and security considerations will vary depending on whether the agent uses a "hard" or "soft" IP phone. A careful assessment of what your organization hopes to achieve from its deployment is essential in order to make the best choices based on both infrastructure requirements and business need.

Get the kinks worked out. When piloting an IP solution, organizations need to take a number of considerations into account. Most important is to have

Go To Table of Contents | Go To Ad Index

By Karen Auman

as accurate an understanding as possible of call volume as this will determine the amount of bandwidth needed. Bandwidth calculators are helpful in this regard and widely available on the Web (see http://www.voipcalculator.com). In addition, the IT team will need to understand how to reduce latency, manage network congestion, and control quality of service. Once the stage is set for VoIP, the next and most important step is choosing the solutions that will leverage your organization's IP infrastructure to its fullest potential.

Choosing IP Solutions: Five Essentials

Practicality. With all the attention surrounding VoIP in the last several years, it can be difficult for today's businesses to separate hype from reality. Resist the temptation to jump on the IP bandwagon without first evaluating what really makes sense from a business perspective: (i.e., How can we improve customer satisfaction? Boost agent performance and efficiency? How can line of business managers obtain a more integrated view of contact center activity?) A company can invest millions in an IP network, but few true benefits will be realized unless application choices leverage the technology infrastructure — whether it's stand-alone IP or a combination of IP and PSTN — to drive real-world improvements. Make the wrong application decision and your organization could be looking at an increase in expense and inefficiencies, regardless of the underlying VoIP network.

In many ways, today's software-based contact center solutions are the "killer applications" for VoIP because they can take advantage of it to add new capabilities, as well as improve existing functionality. For example, IP-enabled applications can support sophisticated call routing, consolidated reporting for an enterprise-level view of contact center activity, and help organizations seamlessly link remote agents and offices to present a single face to the customer. In addition, these solutions don't overlook simple enhancements to things like agent screen pops and IVR (define - news -alert)features that can lead to performance improvements independent of network infrastructure change. Networking devices such as gateways and routers will quickly become standard industry commodities. The solutions that make a difference will be those that help organizations stand apart from the competition to become more streamlined, cost effective and efficient — no matter how much investment is made in the underlying network.

Leverage existing investments. Taking advantage of IP doesn't mean your organization needs to throw out existing (and expensive) voice investments. And for most contact center managers keeping costs in check is just as important as advancing capabilities — no one likes to suffer through "forklift" upgrades. Look for solutions that can incrementally add important IP functionality, and at the same time extend and enhance infrastructure already in place. Operating in an IP or PSTN (define - news -alert) environment doesn't have to be an either/or proposition, especially for contact centers with years of investment, planning, and development linked to existing switching equipment such as ACDs (define - news -alert) and PBXs (define - news -alert). In addition to hardware infrastructure, consider your organization's investment in existing business rules, configurations, and user interfaces. Hybrid architectures make it possible to deploy both PSTN and IP-based agents on a single platform — a good choice for businesses that want to get the most out of the investments they've already made, take advantage of all that IP applications have to offer today, and lay the foundation for future IP migration that minimizes risk.

Simplicity. In a perfect world, technology is supposed to make business simpler, but all too often the opposite is true. Poor choices can result in unnecessary complexity for developers, administrators, managers, as well as users. It is critical to look for IP solutions that enable a single, integrated environment to manage contact center activity and workforce performance, create and deploy business rules, generate reports, and more — regardless of underlying technology. Complex infrastructure creates functional silos that require expensive administration and redundant staffing. Solutions that foster simplicity

VoIP – The Voice Of Choice

By Mark Blankenau

Telenational Marketing made the VoIP choice nearly two years ago and it has proven to be one of our best business decisions. Telenational, located in Omaha, Nebraska, is a Direct Response Center offering 24/7 Call Handling Services for standard or custom DR medium-sized campaigns and test projects. The addition of VoIP has made a positive influence on our profit margin. It certainly gave us the opportunity to remain competitive while offering quality teleservices to our clients.

Communications and labor are two of the primary costs for any Call Center. In a continuous effort to control and minimize these expenses, we researched any option that we felt would decrease expenses and increase profits. The typical local solutions such as constant renegotiations with long distance vendors and relocating facilities to areas of higher unemployment did not appear to offer any long-term effect. Moving or adding offices anywhere else in the U.S.A. was a budget nightmare. Staffing issues continued to create the need for an increase of benefits and incentives but did not effectively control the cost of employee turnover. All solutions resulted in a constant drain of profits and the cost had to be passed on to our clients.

It was at this point Telenational Marketing began its research for business alternatives. Realizing the high-quality labor market that exists in the Philippines, the next step was to tap into this labor pool of well-educated and reliable workforce.

Like many new ventures, our initial attempt in setting up operations in the Philippines was a failure. There were a few minor issues we overlooked and a couple of major issues. The first was the dial-up cost was prohibitive for keeping the line connected even when an operator was not on a call. The other issue was that our first partner had no "American teleservice' expertise. We ultimately had to sever the relationship.

The challenge of tapping the Philippine labor market hinged upon finding a redundant fail-safe method of connecting to Telenational's U.S. operations and the Philippines. The solution was VoIP. It not only allowed connectivity and the ability to retain U.S. call handling projects, but gave Telenational the ability to develop, train, and monitor from our Omaha office.

Our present U.S. partner was operating in the Philippines at a local level and had the technical ability and teleservice expertise we required. PRI circuits were set up between the switches on each side and the Cisco routers. Then VoIP dialing patterns were set up to allow dialing from our switch through the Cisco routers and to the Cisco router in the Philippines and to their switch. The voice quality vastly improved at 16K compression; at 32K compression the quality was even better.

As we continued to develop the processes and systems, we found that not only was there a lower labor cost but there was a higher educational level of employees. Most operators have a minimum of a Bachelors Degree or equivalent, speak fluent English, and are trainable and anxious to work. Each new operator receives a minimum of two weeks of call handling training plus an additional two weeks of 'Americanisms' to assure understanding and accuracy of the information given by the customer. Operators learn the system and are given knowledge and details of each product or service, with role playing as an integral part of the training. All this is accomplished prior to activation of any account, thus assuring understanding, accuracy, and continuity of the program.

Go To Table of Contents | Go To Ad Index

Flexibility has also been one of the keys of success. All administrative functions are performed from our Omaha office including Operations, Accounting, IT/Programming, Sales and Client Services. We design the screens and script, train, monitor, and define acceptable levels of performance for all campaigns. Our partner also implemented their own control systems and the combination offers a dual quality control effort.

VoIP allowed Telenational Marketing to remain competitive in today's global market. It has proven to be so successful that we urge you to consider this option for similar communication changes in your organization.

Mark Blankenau is President of Telenational Marketing. For more information, please visit http://www.telenational.com.

are cost effective and efficient. For example, operations staff, charged with keeping mission-critical contact center systems running around the clock, will find it much easier to monitor and administer resources if they have a single interface and single set of system logs and alerts. Likewise, agents should be able to operate from a single user interface, whether they are handling customer contacts via the phone, e-mail, or the Web. A comprehensive, fully-integrated contact center solution can deliver an environment that is less costly to maintain, whether an organization chooses to run on a 100 percent pure IP or hybrid environment.

Capture the "Big Picture." As the contact center scales outwards, it becomes increasingly important for organizations to capture an end-to-end view of customer service and agent performance across multiple channels, business units and geographies. Consolidated, holistic reporting (as opposed to piecemeal reports on specific functions or locations in the contact center) is the only way that everyone in the organization — from contact center managers and agent supervisors, to system administrators, executive management and agents themselves — can get access to the information they need to continually improve customer service, efficiency, and performance. Look for solutions that offer pre-built and pre-integrated reporting capabilities that collect and analyze contact center data from across the enterprise and present it to users in a timely, actionable, user-specific format — regardless of what system generated the data.

Open standards compliant. As standards such as VXML, SOAP, Web services, and SIP become more and more ubiquitous, the contact center industry must shift towards the use of "open systems." Solutions supporting a common set of technology standards enable faster integration, simpler administration and greater flexibility and scalability. Because of all the advantages afforded by open standards, IP solutions built upon this foundation are a must.

When it comes to leveraging IP solutions in your organization's contact center, don't get lost in the hype. VoIP has many cost and efficiency advantages, but will yield minimal results for the contact center unless paired with the right business applications. Clearly articulate contact center goals, carefully evaluate solutions, and make sure that business management is aligned with IT as your organization moves forward. With proper planning and a realistic approach, companies can avoid costly mistakes while evolving the contact center to take advantage of all that VoIP has to offer, both today and in the future.

Karen Auman is Senior Product Marketing Manager at Aspect Communications. For more information, please visit the company online at http://www.aspect.com.

A Cellular And WiFi Rendezvous

With cellular phones in virtually everyone's pocket and wireless LAN deployments progressing rapidly across the residential and commercial landscapes, there are obvious benefits to be gained by enabling roaming between the two types of networks with the same device. Cellular networks can deliver mobile connectivity when users are away from the home or office, while 802.11-based wireless LANs can carry those connections indoors when they reach their residence or desk, effectively and inexpensively extending the service coverage area. With such convergence just over the horizon, users can realize the dream of single-number access — being able to reach each individual through the same phone number, regardless of his or her location — and corporations can leverage IP voice within their facilities to reduce cellular phone charges and improve productivity.

There is tremendous momentum behind the convergence of cellular and WiFi networks. Recent studies indicate that over 30 percent of subscribers are already using their mobile minutes within the home and office. Unfortunately, users often experience difficulty maintaining call quality and consistent connection in these areas. Installing additional base stations for all indoor coverage quickly becomes cost prohibitive for mobile carriers. Convergence of the cellular network would provide advantages for both these mobile carriers and end users. With convergence, end users could consistently use a single phone number for all their communications — business, personal, desk-based, voice mail and more. In addition, end users would gain a very-high-speed data connection right to their mobile device. At the same time, cellular carriers benefit from inexpensive coverage and capacity expansion to their GSM (define - news - alert) networks, and increased customer satisfaction through better service coverage.

Because of these advantages, the momentum for a converged solution is steadily building. Already, handset vendors have announced telephones that can support GSM mobile networks and 802.11 wireless LANs. In February of 2005, a specification for Unlicensed Mobile Access (UMA) technology which details how a dual-mode phone roams from a licensed spectrum mobile network to the unlicensed spectrum (or WiFi) network, was formally approved by the 3GPP (3rd Generation Partnership Program) and will be included in 3GPP Release 6, the next major update of the standards for GERAN (GSM/EDGE Radio Access Network) TSG (Technical Specifications Group) — the international standards body.

With all of this market momentum and the drive to have single-number GSM/VoIP access, one important industry requirement has yet to be met: the need for wireless LANs that can support IP voice that not only meets, but improves, on the quality of service and coverage that users have come to expect from their current cell phones.

Traditional Access Points Can't Deliver Quality VoIP

The current generation of WLAN access points (APs) are a key obstacle to

Go To Table of Contents | Go To Ad Index

By Joel Vincent

Fortunately, the 802.11 standard is specifically written to enable innovation.

delivering quality voice communications over wireless LANs. The problems encountered a few years ago when designing WLAN equipment are not the same problems that must be solved to use a WLAN as an extension to the mobile voice network. Previously, the design focus for WLANs in the enterprise was to ensure that they were manageable and secure. That design focus drove innovation in the deployment architecture of the WLAN — the trend moved from distributing independent APs that needed to be individually configured and maintained, to distributing the wireless access but centralizing the management and security of those APs into a control unit or switch. This type of centralized architecture was a tremendous leap forward for controlling and managing a network of many access points and moved the WLAN industry in an important direction: out of the coffee shop and into the enterprise to support improved productivity. However, this still did little to solve the problems created by asking the same network to support VoIP. The technology transporting the VoIP packets from the network to the client and back is still the same technology being used in the coffee shops. In some instances, vendors have placed fancy management software wrappers on the access points and some have even gone so far as to distribute their code to these coffee shop hotspot providers so that their hotspots can be managed by the vendors' centralized switches. Unfortunately, software is not enough.

There are two significant limitations of these software-wrapping approaches that make them unsuitable for carrying voice: the inability to prioritize and reliably deliver data packets both to and from the client, and the inability to maintain predictable performance between the client and an AP as the client roams from one AP's coverage area to another's. The problems experienced by hotspot technology can be traced to their Media Access Control (MAC) technologies. APs based on these technologies have access to some general wireless information such as signal strength, station identification, link status, and other asynchronous information, allowing the WLAN switch or controller to set the aggregate transmit powers for the cells, for example, and to crudely shuttle clients onto different APs on the basis of some optimization criteria. But the WLAN has no control over the details of the RF transmission (the latency, jitter, and error rate) because this information isn't being presented to the APs' software and therefore isn't being used to make any real-time, synchronous decisions. It is precisely these details — the latency, jitter, and error rate — that must be understood and coordinated to ensure predictable quality of service for voice.

Without having detailed information of the transmission over the air, APs designed with these simplistic technologies are unable to differentiate voice clients from other clients, leaving the client to work in a best-effort method to get their chance to transmit. Since data traffic is much faster and more aggressive, it will crowd out the slow and steady voice calls because the collision-avoidance algorithm in WLANs (CMSA/CA) responds to the most aggressive incoming traffic. It's this inherent randomness in the last 100 feet of the communications that make these hotspot APs — and even their enterprise big brothers — unsuitable for use in a VoIP network.

Just as these APs make predictable packet delivery impossible, they also prevent reliable handoffs when the user roams from one AP's coverage area to another's. The hotspot APs act as an independent entity in terms of managing their connections, so in this architecture the client collects information about the network on its own and makes the decision of when and where to hand off. A client that switches from one AP to another is forced to perform a handshake each time it re-associates. This handshake has many components, and if any part is lost due to congestion or interference, the entire handoff is delayed. According to David Newman, writing in Network World, recent industry tests have shown that these commodity-based systems experience delays in their handoff processes, consuming half a second to multiple seconds during which no data is being transmitted and call quality is severely impacted.

A Better Solution Is Required

Fortunately, the 802.11 standard is specifically written to enable innovation and encourages methods by which vendors can design enterprise-grade access points that actually do provide precision control over the quality of client connections, and which thereby solve the problems of voice quality.

As it turns out, the real solution to these problems isn't coming up with ever-more-clever software to force legacy and commodity-based independent APs to work together. It lies in re-thinking the design of WLANs and APs to create a global system that meets the requirements of large-scale deployments that are intended to be voice networks. This is where the concept of a Cellular WLAN becomes the next logical evolution for WLAN technology. In a cellular network, each AP coordinates with the others to optimize the connection with each client as those clients roam. These APs use MAC technologies that have fully controlled, over-the-air interface with each client, and can control the ordering of both transmitted and received packets to adjust precisely for the quality of the connection required by each caller. This precise control over the timing and transmission and packet-by-packet scheduling, delivers traffic management over-the-air not only for each AP, but also for coordination across APs. Such a WLAN system creates awareness across

Go To Table of Contents | Go To Ad Index

APs so they can coordinate transmissions to provide clear voice communications for every client without fear of interference from APs that are acting independently on the network.

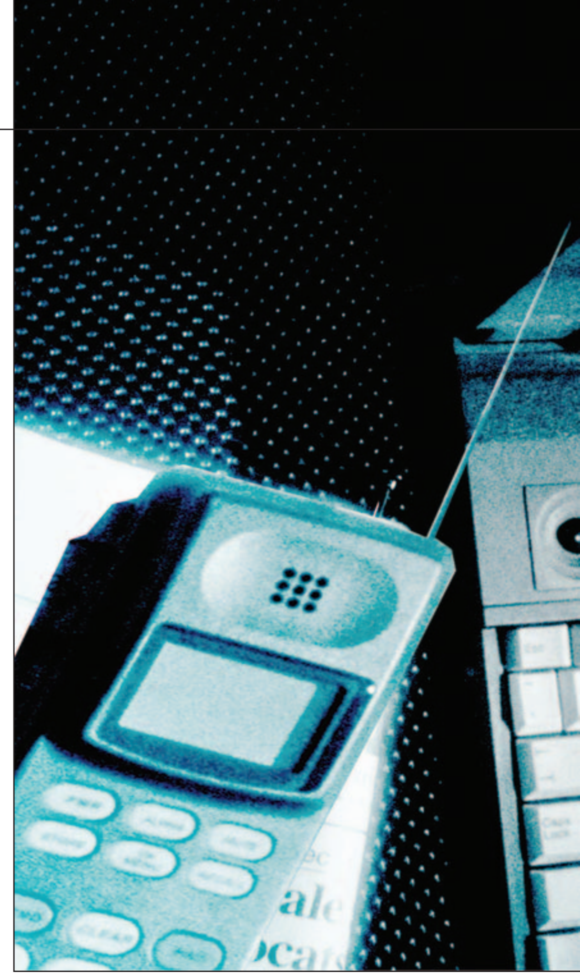
In addition, by asserting control over the MAC, access points can cooperate — without knowledge or involvement of the client — to seamlessly transfer ownership of a connection in exactly the same way as the cellular telephone network, i.e., replicating the connection at the next AP and transferring the transmission to the new AP while simultaneously tearing down the old connection. When a dynamically controlled MAC is used, maintaining voice quality during handoff becomes a simple matter of having the central controller choose which AP is best. In other words, the wireless infrastructure decides which AP is best for the client at any given time, rather than the client deciding which AP is best.

Because the coordinated APs and wireless controller are maintaining awareness and control of all the transmissions and the quality of service to each client, the Cellular WLAN approach brings order and predictable service quality to an otherwise random communication medium. As such, this approach resolves the voice quality and WiFi handoff problems in a similar fashion to the proven method used in cellular telephone networks.

As the momentum continues in the convergence between Cellular and WiFi networks and WLANs move from convenient data networks to critical, revenue generating extensions of the voice network, it will be essential for WLAN vendors to use hardware architectures that provide precise control over the quality of each AP-to-client connection. While hotspot AP technology may have been sufficient for coffee shop coverage and even for data coverage in an enterprise with centralized management and security, it is ineffective for IP voice applications or for deploying a reliable extension to the mobile voice network. New problems require new solutions and the Cellular WLAN approach represents the ideal solution for the emerging, converged wireless networks. IT

Joel Vincent is director of product marketing at Meru Networks, Inc., a VoIP mobility infrastructure solutions specialist. For more information, please visit the company online at http://www.merunetworks.com.

Go To Table of Contents | Go To Ad Index



Is VoIP Ready For WiFi?

The talk has already started. As VoIP gains widespread adoption, technology providers have already started looking out to the future for the next "killer app" by combining VoIP phones with WiFi technology and cellular technology. However, the time to reach consumers and SOHO business users with life-changing VoIP technology is now. With the current infrastructure for VoIP phones already in place, the market is ready to provide the VoIP experience to a whole new range of customers. Proven systems have been set up to provide first-time users with a high-quality experience, at a tremendous value. For so-called WiFi phones that combine VoIP and cellular technologies, there are a number of infrastructure questions that need to be answered before this technology can show its full potential.

The tech industry often puts value on innovative, disruptive technologies, but it's important to remember that the most beneficial applications are often times the ones right in front of you.

At a recent industry conference, a quick survey of all of the service providers revealed that they are eager to deploy VoIP (define - news - alert) phones with Wireless LAN 802.11a/b/g technology (VoIP with WiFi). However, these providers don't see this technology becoming a mainstream play for consumers for at least two or three more years. Some of the main factors holding back the widespread deployment of VoIP WiFi (define - news - alert) phones include price, a lack of access point coverage in the home, interference, battery life, and quality of service.

Are We There Yet?

There is no doubt that VoIP WiFi phones will become a lucrative market in the future, but the question is, "Are we there yet?" These phones typically carry a price tag of $300, breaking the pain point that many consumers are willing to bear. Until WiFi phones get to the $100 price range, the market is not likely to see mass deployment. In addition, WiFi is still a technology that is generally suited solely for PC applications. Voice technology is still prone to packet drop over WiFi, which can disrupt a phone conversation and cause frustration among first-time users.

There are a number of working groups, such as 802.11r and 802.11s, that are actively working on technology that will enhance the WiFi VoIP experience. Although the current 802.11 standards on the market are evolving rapidly, the standards will most likely be completed in late 2005, early 2006. Products would not reach the market until mid to late 2006.

WiFi technology is still limited by the number of access points in the home or in other environments. In some cases, WiFi technology can require additional 802.11 access points around the average-sized home, which increases the cost of the experience. WiFi technology is also still prone to interference issues from cordless phones, microwave ovens, garage door openers and a myriad of other wireless applications found in a typical household. As such, WiFi-enabled VoIP phones are still relegated to the science experiment stage.

It could also be said that WiFi phones have the convenience of operating wherever there are hot spots, but the infrastructure is not yet to the point where WiFi phones can operate seamlessly everywhere. Perhaps WiFi-enabled communities may allow complete coverage within their own boundaries, but substantial gaps in coverage still exist between coffee shops and hotels where 802.11 connectivity is offered. Even in the cases where there have been developments of handsets that offer interoperability between WiFi access points and GSM or other cellular networks, battery life and physical size issues have limited their

Go To Table of Contents | Go To Ad Index

By John Kasha

practicality. Under the most ideal situations, these phones can offer only about an hour of talk time, an experience not likely to promote widespread adoption.

The Answer Is Closer Than You May Think

Until VoIP with WiFi makes its big market entrance, it's important not to forget that just about everybody's cordless wireline phone can be made into a VoIP phone with the addition of a media terminal adapter (MTA). Consumers with broadband connections can today purchase a small telephony adapter box that plugs into their home network and connects to a simple "black phone," providing immediate VoIP service. Such MTAs are attractive as they preserve investment in legacy phones, too. Service providers are estimating adoption rates in the tens of millions worldwide at the end of 2005 for this type of application.

It is estimated that most households are already equipped with a cordless phone, given they have become reliable and ubiquitous. The wireless technology used in these phones, whether they are operating at 900MHz, 2.4GHz, or 5.8GHz, is inexpensive and users can easily roam their entire homes without dropping a call.

VoIP phones are at the stage in the industry's development where the Apple iPod or digital video recorders were two years ago and an area where TAs with cordless phones can excel. Consumers know about them and have a vague understanding of what their capabilities are but, until they get them into the home and into operation, they may not fully comprehend all of the benefits.

For example, when a consumer's VoIP phone rings, it can also ring his/her cell phone or send the voice message to the user's e-mail. That e-mail message can be played back over Windows Media Player or another audio playback software providing consumers with a convenient way of staying connected. Other services, like Internet applications, are readily available in existing VoIP-enabled phones. Certain buttons on a keypad can be configured to retrieve information from the Internet, like stock quotes, and display it on cordless handsets. In this sense, it is imperative that end-users have a flawless experience so that they view VoIP as a great, new, useful technology.

The most important aspect of VoIP technology, at this critical point in the technology's growth stage, is for users to have a positive experience with their first VoIP phone and service provider.

In order to realize the full potential of this growth rate, first-time VoIP consumers need to receive a seamless, trouble free experience that is very similar to the wireline phones they are now using on the public switched telephone network. Once end-users have become accustomed to these phones, they will start utilizing advanced data convergence features that VoIP enables. If, however, consumers are forced to deal with spotty coverage, reduced talk time, security leaks and high retail prices — all of the ills currently besetting WiFi phones — the VoIP rollout that we've been anticipating will get pushed back much further than we expected. WiFi phones for VoIP will be a technology to embrace, eventually. But VoIP phones using existing, successful cordless technology are ready now for consumers to enjoy without the current entanglements with WiFi. IT

John Kasha is director of VoIP marketing at Centillium Communications a broadband access solutions provider. For more information, please visit the company online at http://www.centillium.com.

All Aboard The VoIP Train

Five Criteria For Selecting A VoIP Development Platform

After years of anticipation, the 'VoIP train' has finally arrived at the station and you and your company desperately need a boarding pass before it leaves for its next destination. But, be warned — this train is crowded and you must make sure your pass for the VoIP train is legitimate.

Your company's "ticket" is actually a VoIP (define - news - alert) development platform. Selecting the right VoIP development platform can put your organization on the fast track to VoIP success and increased revenue generation via the introduction of new services. Selecting the wrong VoIP development platform can result in endless delays that will ultimately result in customer churn, affecting your organization's reputation, and, most importantly, bottom line and chances of future VoIP success.

To this end, below is a checklist of the top five attributes you need to evaluate when selecting a VoIP development platform.

Support Of Open Standards

Successful VoIP application development is driven by open standards that have been adopted by the industry. It is imperative that the VoIP application development platform you select fully supports these open standards — not only current standards, but emerging standards that will play a major part in defining the VoIP application development process well into the future. Remember: in the telecommunications world there is one constant — change. Be prepared.

Session Initiation Protocol (SIP) and H.323, the first widely adopted VoIP protocols, are key standards that must be supported by a carrier's VoIP development platform. SIP is a signaling protocol used to create media sessions, such as telephone calls, video conferencing, and real-time data streaming in an IP network, and was created by the International Engineering Task Force (IETF) as an open and flexible protocol that could accommodate new, rapidly emerging applications. Recently, SIP has replaced H.323 as the signaling protocol of choice for the majority of VoIP applications.

A popular emerging standard that should be on the shopping list for all those purchasing a VoIP development platform is VoiceXML (define - news - alert). According to the VoiceXML Forum, an industry organization of more than 325 member companies whose mission it is to accelerate the worldwide adoption of VoiceXML-based applications, "Voice XML is a markup language for creating voice user interfaces. It uses speech recognition and/or touchtone (DTMF keypad) for input, and pre-recorded audio and text-to-speech synthesis (TTS) for output. Callers interact with VoiceXML application via a VoiceXML interpreter (also known as a 'browser') running on a telephony server in an analogous way to Web surfers interacting with HTML applications via graphical browsers on PCs."

Flexibility to support the 3GPP IP

By Harold Klett

The VoIP market is akin to the gold rush — everybody wants a share of the treasure.

Multimedia Subsystem (IMS) architecture is also a "must-have." IMS has been defined as a common platform/subsystem for providing innovative services to mobile and fixed networks. IMS is the architecture that merges the Internet with the cellular world, and makes Internet technologies, such as the Web, e-mail, instant messaging, presence, XoIP, and videoconferencing available nearly everywhere.

Support for open standards goes hand-in-hand with interoperability. Although standards are adopted to create an environment for a service provider to select best-of-breed technologies that will work seamlessly together, many vendors will extend the standards in a unique way. Select a vendor that has not only implemented the standards in an open fashion, but can provide the flexibility to quickly support interoperability with other vendor extensions or rapidly adopt new standards.

With multiple standards, complexity increases. In some cases, it may be desirable to provide integrated media server, media gateway, or signaling gateway technology in the VoIP development platform to rapidly deploy services, and minimize cost, complexity, and potential interoperability issues. If you decide to go this route, select a platform that provides technology options such as SS7, CAS, PRI, and Media to future-proof your investment.

Reliability And Stability

Subscribers that flock to VoIP services are the same people who have come of age in the Internet era and demand immediacy. As VoIP services are widely deployed at service provider locations worldwide, service interruptions are not acceptable, and quality of service is a main concern. Therefore, it is extremely important that a VoIP development platform be put through its paces to ensure it is a reliable and stable foundation for a carrier's VoIP business. A VoIP development platform must be "carrier-grade." Accept nothing less. Demand to speak with companies that have already implemented the platform. After all, the future success of your business could be riding on the VoIP development platform that is deployed.

What exactly is "carrier-grade?" It means more than having a hot-swappable fan tray or redundant power. It means having redundant support for all active components, and separating transport, signaling and OA&M to reduce susceptibility to intruders and to ensure better system performance.

Scalability

It is assumed that you will be deploying a VoIP development platform to expand your operation's subscriber base and revenue. Therefore, you must ensure the VoIP development platform can scale to accommodate your organization's growth easily and cost-effectively.

A key factor that must be considered is being able to deploy a properly sized system at an attractive initial price that will enable you to cost-effectively scale the system. As you will be expanding the platform, make sure you purchase the base system at a competitive price entry point — in terms of dollars and ports, taking into account your total cost of ownership. The ease of scalability must also be investigated — it shouldn't be a major, disruptive effort to add additional services and capacity to the VoIP development platform. Finally, make sure the platform has technology flexibility, and can support other emerging network architectures, spanning fixed and mobile networks.

Investment Protection

As mentioned earlier in this article, telecommunications service providers must be prepared for change at all times. Plan ahead. It is essential that the VoIP development platform you select can easily accommodate the continuous telecom network evolution that will inevitably span multiple architectures. An ideal VoIP development platform will be easy to operate and upgrade, requiring less sparing and eliminating "forklift" upgrades.

Select A Proven, Experienced Vendor

There is no substitute for experience. Select a sustainable VoIP development platform vendor that has deep experience and an impressive track record of success equipping service providers with development platforms that have proven to increase revenue and subscribers. The vendor should have exemplary support and services to speed your time to market and improve overall satisfaction. The vendor should be knowledgeable of current and next-generation architectures and technology to enable evolution of services. Finally, the vendor should have a global presence, thereby enabling the application developer to implement services once and deploy worldwide — regardless of the network.

VoIP is today's hot networking technology. The VoIP market is akin to the gold rush — everybody wants a share of the treasure, and both seasoned professionals and common citizens will prospect. Selecting the right partner, with the right technology and vision, may be the most important decision on the track to success. **IT**

Harold Klett is vice president, product management at Excel Switching Corporation. For more information, please visit the company's Web site at http://www.excel.com.

Go To Table of Contents I Go To Ad Index

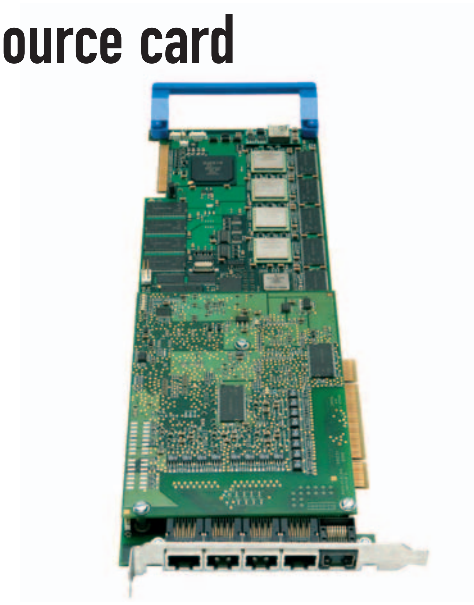
Advertising Supplement

This is a special advertising supplement to **INTERNET TELEPHONY**® magazine. The information contained in each corporate profile was prepared and submitted as it appears by the featured company, not by **INTERNET TELEPHONY**® magazine, and therefore, each company is solely responsible for the validity and accuracy of its profile. To qualify for a corporate profile, a company must be a contract advertiser; this feature is an element of the benefit package for those companies.

Corporate Profiles

Aculab offers unrivalled benchmarks in density and price with the Prosody X media processing resource card

Aculab offers solution providers a wide range of computer telephony hardware and software for integration into high performance communications solutions - from contact centers and IVR to prepaid services. Products for use within telco or enterprise solutions include media processing resources; speech processing, fax, conferencing and echo cancellation - in both TDM and IP environments, and digital network access including VoIP and SS7.



Support is available to help developers through each stage of their product's life cycle including pre-sales consultancy, technical support, training and marketing. Aculab's portfolio ensures solution providers have the mix of capabilities required to meet the ever changing needs of the communications market.

What's new from Aculab?

The latest edition to the media processing portfolio is Prosody X, which is a genuine breakthrough in card design that embraces the reality of IP in today's communication networks while also supporting TDM connectivity as an option.

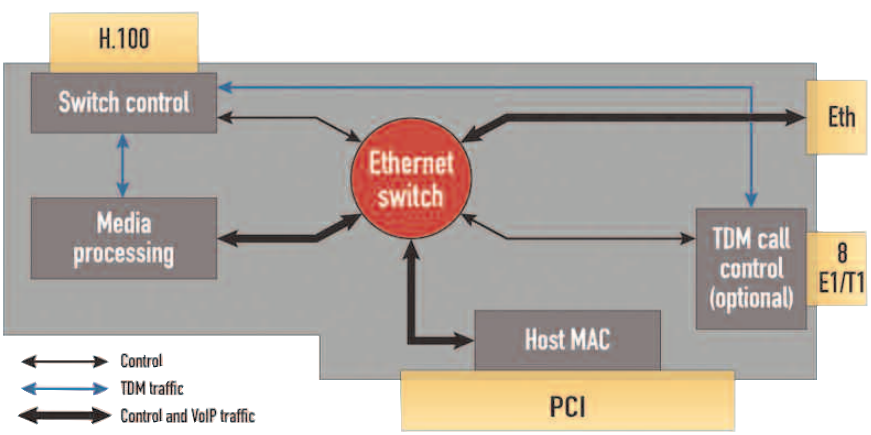
Designed around an IP core, its architecture makes the product distributable amongst different chassis platforms offering resilience and scalability as well as helping future proof solutions as they move to IP.

Prosody X PCI at a glance:

- 600 channels per card
- Speech, fax and data
- On-board IP architecture
- IP connectivity
- TDM connectivity
- Unrivalled benchmarks in density and price

Okay, cards on the table! Let's take a look at Prosody X hardware:

- Ethernet switch at its center, the beating heart of IP
- External Ethernet connection not only conducts VoIP traffic, but also allows the use of cards in remote chassis with support for SNMP
- Rich media processing DSPs give large channel counts of VoIP codec work and tasks like record, playback, tone generation/detection, conferencing, ASR and TTS
- Optional connectivity for up to eight E1/T1 trunks



How will it help you - an application developer - win business?

The possibility is opened up not only to create large-scale VoIP applications like IVRs and ACDs, but also compete with large-scale proprietary products by building equivalent solutions with Prosody X.
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With over 200 employees worldwide, Epygi continues to expand its global distribution through resellers and distributors. Sales and Development offices are located in the United States, Canada, Spain, France, Italy, Germany, United Kingdom, Romania, Turkey, Armenia and Japan. Epygi products operate as stand-alone units, and are interoperable with a large numbe of existing analog and IP products, while providing reliable service to carriers offering IP voice services. The Epygi Quadros are positioned to serve the SME and SOHO markets, teleworkers, and branch offices of large corporations and organizations.

Why Purchase a Quadro IP PBX?

It is difficult to get solid numbers when shopping for a small office IP PBX. Perhaps this will help.

According to three sources, the 2006 market size for PBX shipments is 26,000,000 ports (RBI, Dell-Oro, and Infonetics Research). Next year, IP PBX shipments will also pass traditional PBX shipments. And, trends indicate IP PBX shipments will continue to increase by 29% a year until 2008. This is great news if you are a big company.

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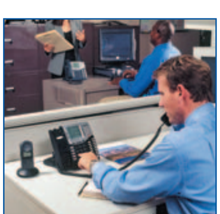

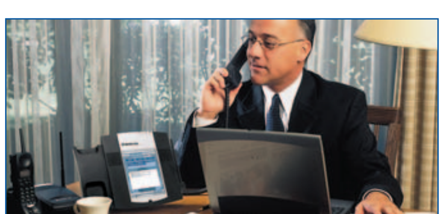

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Iwatsu Voice Networks' Enterprise Communications Server afford small to medium-sized businesses (SMBs) an application rich feature set that is unique to systems of its size. Unified messaging, speech recognition, call reporting, and text-to-speech applications that are typically used by Fortune 1000 companies have been brought within reach of SMBs looking to increase their productivity while reducing operation expenses.

Enterprise-CS

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Also included in the design of Iwatsu Voice Networks' newest communications server, the Enterprise-CS, is the integration of QuadFusion™ Technology. This technology was developed by an international design team, and integrates the four dominant protocols: TDM, VoIP, SIP, and H.323. The key feature of QuadFusion™ Technology is the ability to run any of these protocols either exclusively or in combination.

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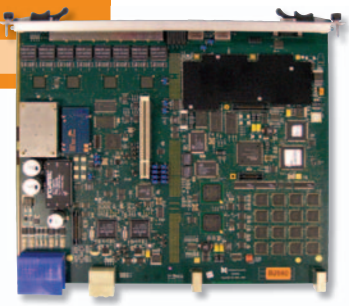


The Advanced Telecommunications Computing Architecture (AdvancedTCA®) is an open industry specification for building high-performance telecommunications and data communications systems. Developed by the PICMG consortium, AdvancedTCA is now poised to make considerable inroads into a market that has traditionally been dominated by vertically integrated proprietary systems.

The initial PICMG 3.0 specifications, which cover AdvancedTCA, were approved in December 2002, an event that launched the creation of an industry of AdvancedTCA offerings. Three and a half years later, a number of vendors have publicly announced products: chassis (12 vendors), CPUs (16 vendors), module carriers (8 vendors), switches and hubs (8 vendors), storage disks (3 vendors), media processing and signaling boards (4 vendors), and AMC modules (8 vendors). Several more vendors are active but have not yet announced products. This community of products and suppliers has reached a critical mass of capability and all of the major network equipment providers now have programs to evaluate and recommend AdvancedTCA components for use by their application groups.

Why all this interest in AdvancedTCA? AdvancedTCA builds on the success of CompactPCI®, making it easier to develop highly available network application platforms, including media servers. However, the most important change is that AdvancedTCA boards are all serial bus connected, in most cases with Gigabit Ethernet. This use of Ethernet eliminates the parallel PCI bus as a potential single point of failure and requires all boards in the system to be intelligent and autonomous. This is a key change for implementing board-level redundancy, a necessary technology for high availability in networking equipment.

AdvancedTCA Media Processing Blade

NMS recently announced a new AdvancedTCA media processing blade, the MG 7000A. NMS has been active in the AdvancedTCA community for the last three years, promoting AdvancedTCA and, more recently, AMC (Advanced Mezzanine Card) modules. Always an active participant in the PICMG community, NMS has been working with other companies on interoperability and has been participating in the AdvancedTCA Interoperability Workshop (AIW) events every three months. These events allow PICMG member companies to test their new products without publicity, creating an ecosystem of compatible chassis and board-level products.

The MG 7000A has been developed to meet the needs of new network-based media servers and enhanced services platforms. These new service platforms support both legacy TDM networks and new VoIP networks. And they must support both types of network signaling — ISDN/CAS and SS7 for TDM networks and SIP for VoIP networks. The MG 7000A provides all types of network connectivity and supports a full range of network applications.

The MG 7000A is being built on next-generation technology, including new DSP chips (TI 5441s), a high-performance PowerPC controller, and Gigabit Ethernet interfaces. This results in support for up to 480 VoIP ports via 8 or 16 T1/E1 interfaces. Maximum DSP resources provide 17,000 MIPS of processing capacity, supporting up to 1,000 IP media server ports.

Planning for the Move to AdvancedTCA

Equipment providers currently designing the next evolution of their systems should consider AdvancedTCA as an alternative to server-based systems, proprietary boards, or CompactPCI. The NMS MG 7000A is in the final stages of development and will be available for beta testing by the end of 2005.

For more information on the MG 7000A, contact info@nmss.com or visit www.nmscommunications.com/atca.

NMS Fall Developer Conferences

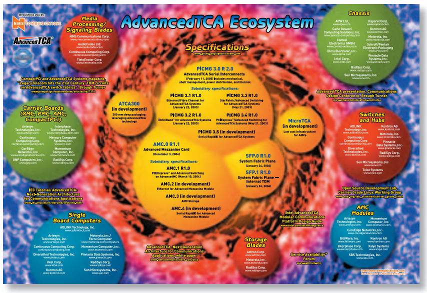
NMS will be hosting three Developer Conferences in Europe, the U.S., and Asia this fall. These Conferences are open to all developers who are either currently building solutions on NMS technologies or considering using NMS products for the first time. Conference attendees will enjoy access to important information and critical contacts, including:

- Insight into media server, video, IMS, and hosted services market opportunities and solutions
- Comprehensive technology and product information on industry-leading NMS product lines
- The opportunity to meet with key product and technical experts from NMS and our partners, who will share their design and implementation experiences
- The ability to network with other developers and explore product synergies and business development opportunities

Visit www.nmscommunications.com/devcon05 to register or get more information today!

September 14–15 Munich ■ November 7–8 Boston ■ December 7–8 Bangkok

Free AdvancedTCA Poster Offer!



NMS has created this fantastic 36x24 poster that includes the very latest AdvancedTCA specifications, useful reference information, and companies currently manufacturing products that fit into the AdvancedTCA ecosystem.

Request your copy today at www.nmscommunications.com/atca

NMS COMMUNICATIONS

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PHOENIX

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By Phoenix Audio Technologies

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The DUET delivers boardroom conference quality everywhere you go.

Boardroom Conference Quality Outside the Boardroom

FEATURES:
- Full Duplex Audio
- Light weight
- USB powered
- Mute. Volume Up, Volume Down buttons,
- A headset connection, external speakers connection

TECHNOLOGY:
Phoenix Audio Technologies' proprietary Acoustic Echo Cancellation, Noise Suppression, Voice Level Compensation, Line Echo Cancellation and acoustic design. Delivers boardroom Conference quality in and out of the office. Conforming to ITU G.167 standard for echo cancelers, the Phoenix DUET is the only true alternative to wearing headset for Voice Over IP.

CONNECTIVITY:
The DUET can be utilized in a number of configurations: Computer USB Speakerphone, Telephone / Cellular Speakerphone, or both Computer + Telephone.
The DUET works with most telephones equipped with a standard headset connection (2.5mm or RJ11), and computers equipped with a USB port. All necessary connection cables are included.

Cisco™ Compatible

www.phnxaudio.com

A New Approach to Voice

Pingtel is the leading supplier of enterprise-grade open source SIP based communications products. The only provider delivering 100% SIP, 100% open source, and 100% Enterprise-grade solutions, Pingtel truly understands how to deliver the benefits of open source-based solutions to the market - combining the innovation and economics of open source, with the feature sets, management platforms, reliability, predictability and support required by customers, channel partners and OEMs. Pingtel was an early pioneer in delivering SIP-based solutions to market and has been actively involved in the development of the SIP standard within the IETF.

SIP + Open Source = Market Transformation
Open Source has the potential to bring PC-like economics and multi-vendor solutions to the telephony market, providing the same magnitude of price and innovation benefits to the enterprise customer. Fundamentally, this is forcing the telephony market to re-examine the current model of vertically integrated single-vendor solutions to the more open model typical of the information technology (IT) sector. By offering enterprise-class, all SIP-based, open source IP PBX software under a Linux-style subscription license, Pingtel is combining the best attributes of open source development - low cost, adaptability and flexibility - with the reliable solutions and support enterprises require for voice applications.

Why Pingtel?
Pingtel's open source IP PBX software is the linchpin technology that will catalyze the movement of enterprise communications into the data center and away from purpose-built hardware. Like enterprise-grade Linux, this approach will drive commoditization of traditional telephony hardware and software and eliminate technology lock-ins that has plagued the industry for decades. This new business model will lead to pervasive adoption and shift significant market share away from vertically integrated, proprietary-solution competitors.

It Takes An Ecosystem
Pingtel's ecosystem partners include a wide variety of SIP desktop phones, analog telephone adapters, gateways and other SIP-based enterprise communications products from innovative equipment and application vendors. Solution providers and end customers can mix and match with confidence that the ultimate configuration of SIPxchange and ecosystem partner components will work together seamlessly to deliver the value they expect. By certifying interoperability with Pingtel's award-winning SIPxchange, companies such as AudioCodes, Grandstream, Ingate, Polycom, VegaStream and others deliver key components to enterprise customers and channel partners interested in building enterprise communications systems based on their exact needs.

A New Approach to Voice
By certifying a high-quality, low-risk solution based on open source, Pingtel is engendering a dramatic shift in the cost structure of VoIP technologies. This shift, combined with the inherent flexibility and adaptability of open source gives IT departments the power to decide when and how it leverages VoIP technologies in its network.

To learn more about how SIPxchange PBX, the SIP PBX for Linux, can add value to your network, visit www.Pingtel.com.

pingtel

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PowerDsine

Company Overview

PowerDsine (NASDAQ: PDSN) is a leader of the fast-growing Power over Ethernet (PoE) technology that allows power to be transmitted over the same network cable as data. PowerDsine has been a pioneer in the development of the PoE market place from its inception.

PowerDsine first developed PoE technology in 1998 and was a founding and active member of the IEEE 802.3af Task Force, which ratified 15.4 Watt PoE in June 2003. Subsequently, the company has been a leader and major market shaper in this space.

The company designs, develops and supplies integrated circuits, modules and systems that enable the implementation of Power over Ethernet in local area networks, providing the capability to deliver and manage electrical power over data network cables. PowerDsine offers system (midspan) solutions and integrated products to its customers who want to enable PoE functionality in their Ethernet switches or networks in order to deliver PoE for applications such as VoIP phones, Wireless LAN access points and IP security cameras.

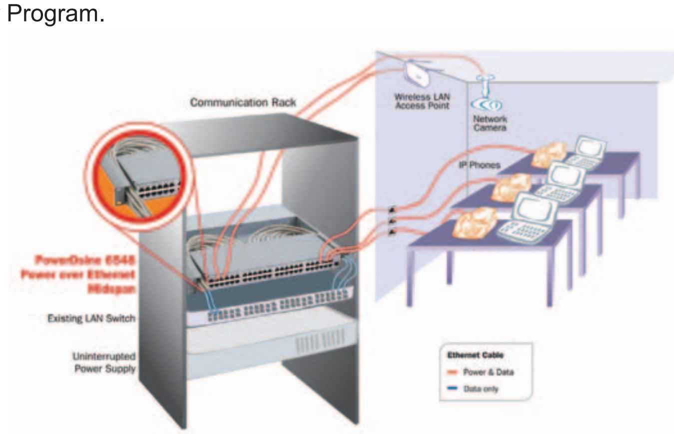
PowerDsine is a member of the Cisco Technology Developer Program.

Power over Ethernet Overview

Power over Ethernet (PoE) integrates the power source onto the same cable infrastructure as the IP data, removing the need to have AC power available at all locations.

The PoE Midspan is an independent power injector that provides Power over Ethernet capabilities to existing Ethernet switches. The PowerDsine Midspan is installed in the communications closet next to the switch.

The output of the switch is connected by a jumper cable to input of the Midspan. The PoE Midspan output is connected to the horizontal run up to 100 meters to the powered device. As a plug-and-play device, the PoE Midspan installation requires no setup, configuration or downtime. Midspans provide power to IEEE802.3af standard devices and pre-standard Cisco devices.



PowerDsine's Power over Ethernet Midspan Solution

Part of VoIP

PoE addresses the reliability issue of VoIP phones by ensuring continuous operation of IP phones during power failures and disruptions. Using PoE technology, an Ethernet cable delivers uninterrupted operating power to an IP phone from the same backup power source that supplies the network servers and prevents service disruption in the event of a power failure. This enables a VoIP phone to be just as reliable as a standard telephone. Powering IP phones through Ethernet cables reduces the cost of new installations or expansions of voice networks by eliminating the need to add separate electrical outlets and wiring for each end-device.

PoE Midspan Main Benefits:

- IEEE 802.3af compliant providing safe and reliable power to IP phones
- Investment protection of existing Ethernet switches and cabling infrastructure
- Saves installation costs by eliminating the need for electricians to install AC outlets near each IP Phone
- Simplifies installation using a single standard data cable for voice, data and power
- Web-based and SNMP MIB support for remote management, on-line supervision and resetting of IP phones and IP terminals
- Increased reliability - get a dial tone even during power outage. By connecting a UPS to a Power over Ethernet Midspan in the communications room, the entire network, including the remotely located resources, is capable of continuous operation during a power outage

For more information on PowerDsine products and solutions please contact sales@powerdsine.com

Enjoy the freedom of secure IP communication with snom VoIP solutions!

snom 360

snom 320

snom Headsets

snom VoIP Box

snom technology AG (Berlin) makes VoIP business telephones based on SIP and related IETF open VoIP standards -- feature-rich, secure, and highly compatible with a broad range of IP PBXs and SIP-based carrier VoIP services. German-engineered for superb sound quality, ergonomics and elegance, snom's broad product line ranges from entry-level information-worker telephones to large-display, extensible executive and attendant desksets.

All snom VoIP phones are highly configurable and easy to manage via keypad-cursor, dedicated function keys and menus, or web browser, and offer user-pleasing features like fully-programmable keys and downloadable ringtones. Most are headset-compatible.

Ideal for general office and knowledge-worker environments, the snom 320 features built-in, full-duplex speakerphone and conference bridging for three-way calls.

Secure and Reliable

snom phones like the entry-level snom 320 and executive snom 360 additionally offer a complete implementation of the IETF's latest recommendations for standards-based authentication and content security (SIPS/SRTP), making them appropriate for demanding applications in general business, research, finance, healthcare, government and the military.

Committed to Open Standards

snom is committed to closely following IETF recommendations for SIP and ancillary open IP telephony standards -- working against lock-in, and controlling costs, while assuring customers of high levels of compatibility, feature accessibility, ease of configuration and general system manageability with the broadest possible range of VoIP premise equipment, services, and solutions.

Focused on the Phone

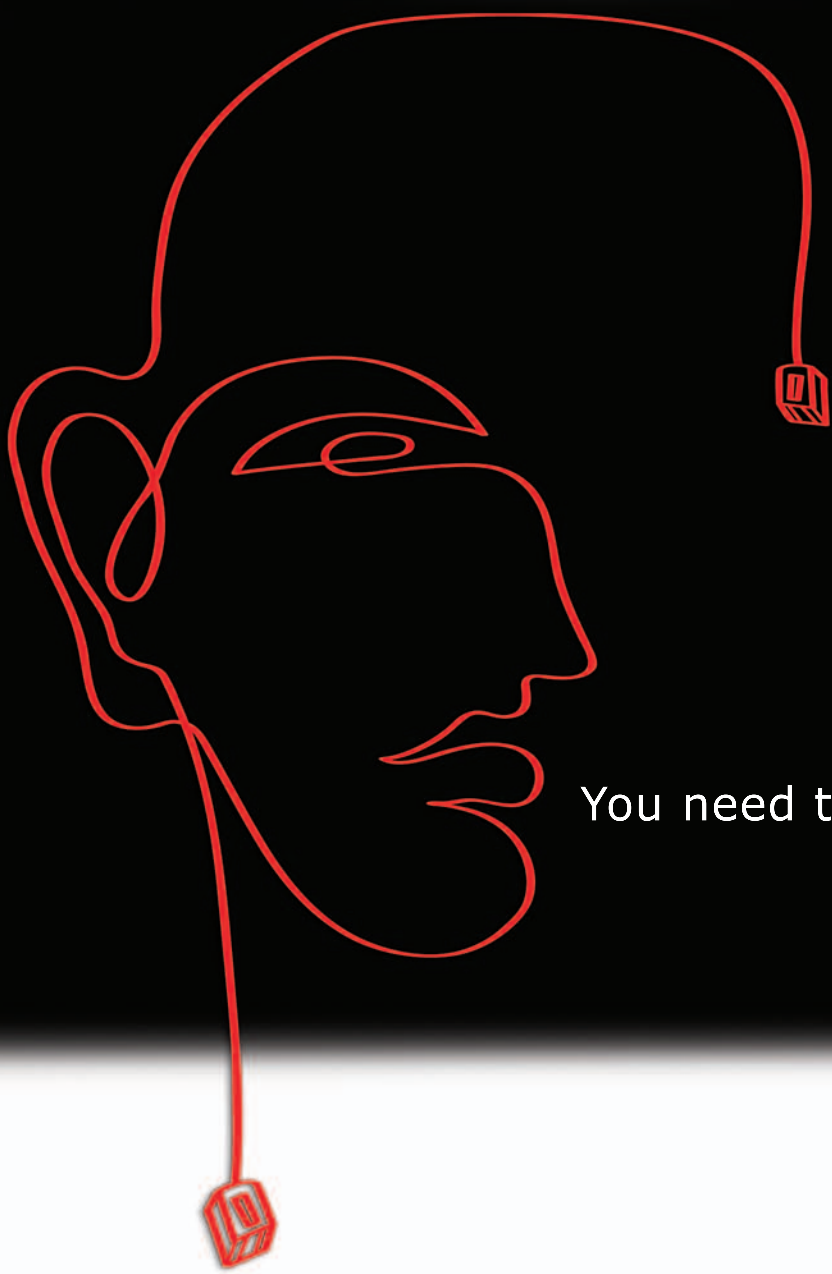
Founded in 1996, snom has, since 1999, specialized in the development of SIP-standard-based VoIP telephones for global business. Through its network of distributors in over 20 countries, snom supplies telephones and collaborates to provide complete solutions for VoIP carriers and service providers, dealers, resellers, SMEs and large corporate end users.

snom VoIP phones

Contact snom for nearest distributor: sales@snom.com

Gradestr. 46 • D-12347 Berlin • Germany
Tel.: ++49 30 39833 0 • Fax: ++49 30 39833 111

www.snom.com



SysMaster
www.sysmaster.com

INTEGRATED VOIP SOLUTIONS

- ACCELERATED TIME-TO-MARKET FOR MULTIPLE VOIP OFFERINGS
- COST-EFFECTIVE LAUNCH OF VALUE ADDED VOIP SERVICES
- SIMPLIFIED SYSTEM CONFIGURATION AND MANAGEMENT
- HIGH LEVEL OF EQUIPMENT CUSTOMIZATION

CORPORATE OVERVIEW

SysMaster develops and markets integrated Voice-over-IP solutions for carriers, service providers and enterprises. Its field-proven offerings allow customers to capitalize on the revenue-generating opportunities provided by the increased demand for next generation telecommunication services. Distinguished for their functionality, scalability and affordability, SysMaster's solutions enable customers to cost-effectively deploy and manage multiple VoIP services, including Calling Cards, IP PBX/IP Centrex, Voicemail, Conference, Follow-me, Virtual Office, and Callback.

PRODUCTS

SysMaster's success is the result of offering integrated VoIP solutions which solve business problems in a cost-efficient manner. SysMaster's products encompass all key infrastructure elements of modern VoIP networks, including billing servers, gatekeepers, gateways, and softswitches. Because all SysMaster products are engineered to work seamlessly together, customers benefit from accelerated deployment, lower total cost of ownership (TCO) and higher return on investment (ROI).

VoiceMaster is fully integrated, highly functional and extremely scalable billing platform, specifically designed to simplify VoIP traffic management and billing. As one of the most advanced billing platforms in the industry, VoiceMaster enables providers to design service offerings with unlimited level of pricing creativity in a cost-effective way. Designed for easy maintenance and upgradeability, VoiceMaster substantially reduces operating costs, lowers TCO and improves ROI.

SysMaster VoIP Gateway is fully integrated, extremely functional and highly scalable switching device, specifically designed for service providers who want to deliver high quality VoIP services to their end users. Supporting multiple communication protocols and voice codecs, SysMaster VoIP Gateway transparently interconnects circuit-switched and IP-based networks while delivering all the switching and signaling functionality necessary to launch revenue-generating VoIP services. With industry's best value-to-cost ratio and modular design, SysMaster VoIP Gateway offers to service providers the opportunity to quickly roll-out multiple value-added services in a cost-efficient manner.

Norfa is an advanced broadband communication services platform which delivers a comprehensive set of next generation services, including IP PBX/IP Centrex, Voicemail, Follow-me, Callback, Conference, Virtual Fax, Web Conference, Desktop Sharing, SMS Messaging, Virtual Office and Calling Cards. By consolidating all major next generation communication services on a single platform, Norfa substantially reduces integration costs at the service provider level and accelerates time-to-market for launching value-added VoIP services. Through bundling and pricing Norfa services creatively, service providers can successfully differentiate themselves and build viable and profitable businesses, competing not purely on price but also on service availability, quality, and reliability.

MARKETS

SysMaster targets both emerging and traditional telecoms who are building next generation networks to provide real-time voice and video over IP services to their customers. Today the company is focused on expanding its presence in the Tier 2 domestic and Tier 1 international carrier segments while aggressively pursuing opportunities in the telecom sector worldwide.

SysMaster Corporation
5801 Christie Ave, Suite 400
Emeryville, CA 94608, USA

Toll free: 1-877-900-3993
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Learn more by calling SysMaster at 1-877-900-3993 or visit us at www.sysmaster.com.

AWARDS:


CallCenter Anywhere®
Open the world to your possibilities.
Why centralize IP Contact Center technologies to support operations across geography? To eliminate the costs, risks and delays ordinarily associated with integrating, implementing and maintaining multimedia IP contact centers at every location. The problem: Most vendors can't empower multi-site organizations to centralize their contact center technology resources and also preserve autonomy for local managers over their own technology-driven business processes. In fact, even those vendors who offer “multi-tenant” centralized solutions often neglect the issue of local autonomy – which inevitably puts more distance between customers and the local managers tasked with serving them.
Whether deployed in-house or hosted by a world-class service provider such as MCI, TELUS or Siebel, our CallCenterAnywhere™ IP Contact Center solution empowers technology centralization while enabling local managers to optimize their own business processes in real-time. No sacrifices.
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VOIP ADMINISTRATORS

Trust Tripp Lite Power Protection

With the rapidly growing popularity of Voice over IP (VoIP) telephony technology, system availability is becoming a critical issue for businesses and organizations of all kinds. VoIP telephony centralization reduces system redundancy and increases the risk of costly downtime due to power problems. At the same time, thanks to rising demand and aging infrastructure, the overall power environment is becoming more and more unstable. Tripp Lite serves this need with a comprehensive range of power protection solutions for VoIP systems.

Since 1922, Tripp Lite has been recognized for innovative products and superior service. The company is a leader in the power protection industry, with 1,000+ products, including a range of uninterruptible power supply (UPS) systems optimized for the protection of VoIP telephony systems.

THE AWARD-WINNING SU6000RT3U

Tripp Lite SmartOnline™ True On-Line and SmartPro® Advanced Line-Interactive UPS Systems assure 100% VoIP availability by providing complete protection from power problems, including blackouts, brownouts, surges, and line noise. Many of these UPS Systems provide expandable runtime options for VoIP applications that require long-term battery backup.

INNOVATION TMC™ LABS 2005 AWARD

SMART2200RMXL2U

Tripp Lite UPS Systems include the company's PowerAlert Software, providing a complete power protection network management and control solution for VoIP applications. Tripp Lite UPS Systems and PowerAlert Software have tested compatible with Cisco Media Convergence Servers running Cisco CallManager, versions 3.3(4)-MCS and 4.0(2)-MCS.*

With years of industry experience and a strong commitment to the emerging VoIP market, Tripp Lite offers the best, most cost-effective power protection solutions for VoIP telephony systems.

Available FREE! Download
Tripp Lite's Latest White Paper:
"Practical Power Guidelines for Cisco VoIP and Internet Telephony Applications"
www.tripplite.com/tele2

TRIPP·LITE POWER PROTECTION

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Meet XO®

XO Communications is a full-service provider of communications services for small & growing businesses, large enterprises and carriers. XO® possesses a wealth of local fiber, fixed broadband wireless, data networking, Internet and long-haul network assets that few — if any — U.S. service providers can match. In fact, XO is positioned as one of the only national, local end-to-end broadband communications companies in U.S. That means XO is about breadth, depth and execution on a national scale in the best local markets.

Proven Leadership and Innovation
The XO leadership team is comprised of experienced communications industry executives who recognized a need for a company that could stand apart from the others. Under their leadership, XO has created and improved the solutions it delivers and has expanded its geographic coverage across the United States. Their mission — and that of the company's more than 4,700 employees — is to provide communications solutions that are simple yet backed by a sophisticated network of systems and products with outstanding customer service.

Network Assets
XO has a wealth of network assets that ensure they can handle your current needs and that they're well positioned for the IP convergence of voice and data services. XO has an OC-192 IP backbone with OC-12 uplinks in our markets and data centers; that means they have one of the highest capacity and scalable IP backbones in the industry, along with the highest levels of performance and reliability. A suite of world-class tools that facilitate the exchange of customer information and continuous network monitoring set the XO network apart from its rivals.

Customer Centric Focus
XO puts customers first. XO customer care is available 24 hours a day, seven days a week, which means an XO specialist is available any time, all the time.

Extensive Product Portfolio
The XO product suite offers businesses of every size flexible and powerful communications solutions. From local and long distance voice, to High-speed Dedicated Internet Access and Firewalls, XO delivers a breadth of product choices.

- Voice Services: A broad portfolio of voice services from local and long distance to conferencing.
- Data Services: Offerings from Dedicated Internet Access to Ethernet and VPN.
- Integrated Services: XOptions® bundled solutions were created especially for businesses looking for a one-stop communications approach. And the newly introduced XOptions® Flex brings the power of Voice over IP to the bundle with virtually unlimited local and long-distance calling as well as web and data services, all on a single bill.
- Managed Services: Complete solutions for outsourcing your most time-intensive telecommunications applications like web hosting, security and IVRs.

XO® Network at a Glance
- 3000+ on-network buildings
- Highly secure data center and a 24x7 network operations center
- 300-plus DSL access points
- 100+ Tier One peering Points of Presence (POPs) offering direct access to 100% of Internet traffic
- OC-192 IP-based backbone
- Total fiber: approximately 1,158,000 miles
- 34 Nortel DMS-500 switches plus additional Class 5 switches for local and long distance voice
- Sonus Networks softswitches for handling next-generation traffic
- Fixed wireless licenses covering 95% of the top U.S. business markets

For more information, see www.xo.com or call 1.866.963.9696.

VoIP MARKETPLACE

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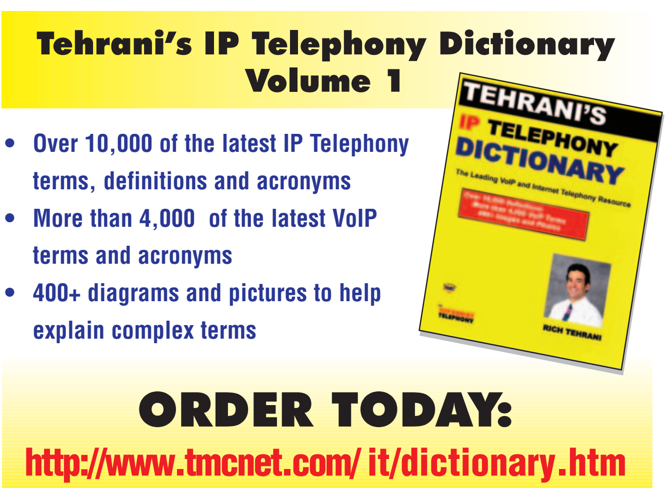

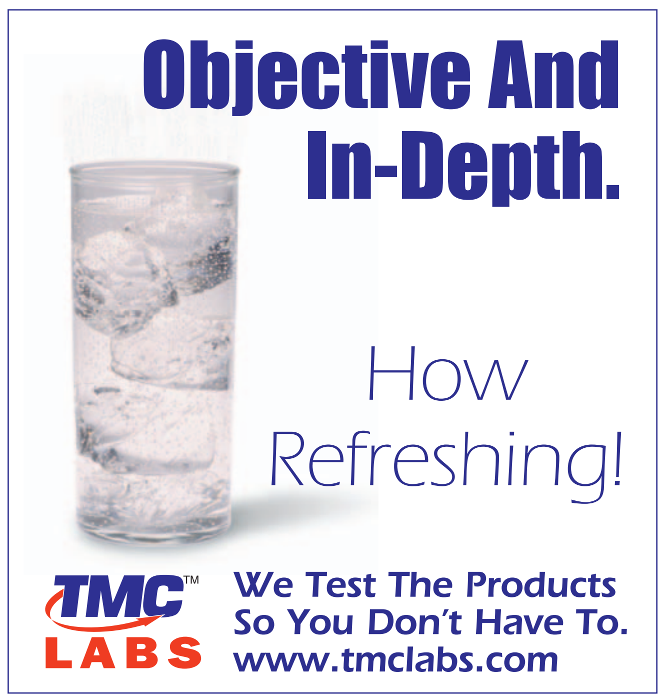

To participate in the VoIP Marketplace, please contact Anthony Graffeo at 203-852-6800 x174 or via e-mail to agraffeo@tmcnet.com or John Ioli at 203-852-6800 x120 or via e-mail to jioli@tmcnet.com.

CEO Spotlight

William Rich
President and CEO
Pingtel

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 William Rich, President and CEO of Pingtel.

GG: What is Pingtel's mission?

WR: At the most basic level, Pingtel's (news - alert) mission to fundamentally improve the way enterprise communications products are developed, packaged, and consumed. We combine the two best ideas in communications — standard SIP and open source — to create a fundamentally new approach to enterprise communications products and solutions. Pingtel's approach will deliver — for the first time — both better, more innovative communications products, and PC-like economics to an industry that has always been hamstrung by lack of choice, high costs, and low rates of innovation.

We do this by giving the customer a 100 percent open source, 100 percent SIP platform (the solution is SIP from the ground up) that runs on off-the-shelf Linux delivering a number of clear benefits:

1) customers can choose the IP phones, other hardware and applications that best meet their price and performance needs — just like IT;

2) customers are not locked into hardware, so hardware (IP phones, etc.) markets become competitive, driving prices down and driving innovation up;

3) customer risk is greatly reduced because the solutions are IETF standard SIP, eliminating obsolesce and allowing customers to leverage innovation;

4) capital and operating costs are reduced dramatically because Pingtel charges for support on a per-server basis, it does not sell per seat software licenses. Or customers can use the free open source version. Both run on completely off-the-shelf Linux, sliding right into existing IT environments.

To further enable customers, Pingtel has assembled an unprecedented group of ecosystem partners committed to this vision including Polycom, AudioCodes, VegaStream, Mediatrix, and dozens of others. Pingtel thoroughly tests each of these partners' products for interoperability and feature transparency with both Pingtel's solution and the open source solutions, and Pingtel guarantees interoperability partners' products with Pingtel solutions.

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

WR: Over the next 18 months, I expect that Pingtel will be recognized as the "Red Hat" for communications, driving a whole new wave of VoIP uptake and innovation. I think the combination of our standards-based design and economics of open source present a compelling value proposition that is hard to ignore.

But of course, we're not done there. Pingtel was an early pioneer with the SIP standard, delivering best-in-class VoIP (define - news - alert) end points and IP PBX (define - news - alert) products. Combining our belief that standards are what truly bring value to the marketplace with the cost benefits of open source development, we see Pingtel as a clear leader not only in PBX and desktop client markets, but in related applications that enable enterprise customers to communicate. With the advent of presence-based technologies, wireless, and innovative ways to connect in an increasingly IP-based world, we think our combination of SIP and open source is well positioned to create a meaningful difference in how people buy, deploy, and use communications systems.

> We combine the two best ideas in communications — standard SIP and open source.

GG: Describe SIPfoundry and Pingtel's role in that organization.

WR: Pingtel was a

founding sponsor of SIPfoundry and we continue to play a leading role in the organization both as contributing developers but also in terms of organizational support. SIPfoundry is an international open source community dedicated to advancing the SIP standard and distributing under open source licenses SIP-based solutions for consumption by end-users and the development community. In addition to the sipXpbx open source SIP PBX on Linux, SIPfoundry projects include rePro, a carrier-grade SIP proxy and infrastructure platform, several SIP clients and high-performance stacks, including the sipXtapi user agent SDK and the reSIProcate stack, and the SIP (define - news - alert) Forum Test Framework, open source test tools developed under the auspices of the SIP Forum and hosted by SIPfoundry.

In addition to being an active participant in SIPfoundry, Pingtel is also an active consumer of the technology developed on SIPfoundry. Pingtel's

The move to VoIP is inevitable.

award-winning SIPxchange is based on a SIPfoundry project called sipXpbx and Pingtel also provides commercial solutions based on the sipXtapi SDK user agent called SIPXua.

For Pingtel, and for me personally, the work that SIPfoundry has done since founding a little over a year ago is nothing short of astounding. We now have over 2,300 participants in SIPfoundry, the organizations hosts seven individual projects that are being used commercially in dozens of products around the world, and we have eight corporate sponsors. SIPfoundry is rapidly becoming the center of industry innovation and a catalyst for a number of important developments in such areas as security and presence. The community is about evenly split between North America and international participation.

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?

WR: I think the move to VoIP is inevitable. The key challenge is to generate value for the market, not simply replace TDM with VoIP. We think combining standard SIP and open source is the key to delivering value by opening the market to innovation (open source) and allowing users to assemble the solutions they need is a normalized and plug compatible way (SIP).

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

WR: We are absolutely committed to delivering innovative, reliable IP PBX and SIP client/user agent products and will continue to focus on these areas to support the company's growth. Natural evolutions of our solutions are in the areas of presence, security, and mobile convergence.

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

WR: Open.

IT

CEO Spotlight

Neal Shact
CEO and President
CommuniTech

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 Neal Shact, CEO and President of CommuniTech.

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

NS: My vision is for CommuniTech (news - alert) is to leverage our relationships as a trusted supplier of our traditional products lines such as messaging and headsets, to our next generation products such as unified messaging and speech enabled products and VoIP endpoints.

The best way to be well positioned in the next-generation telecom market is to be in the current-generation telecom market while developing the expertise with next generation technologies. As new products and services emerge, customers will be facing a baffling array of offerings and alternative ways to access these offerings along with myriad vendors to provide them. In this environment, customers will find comfort and security by dealing with established vendors that have proven themselves over time by standing behind products and services. Those vendor-customer relationships are golden.

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

NS: We have been in business since 1983. Over the years, we have established a large base of satisfied customers. We have added engineers and invested in training and updating the technical skills of our employees. This is a big differentiator because products and services are evolving at a rapid rate, but the number of highly skilled and experienced personnel has grown more slowly. We have been in a position to capitalize on these trends by being an early entrant into the VoIP markets, having been an active player for more than five years, and the founder of the VoIP Council (now part of the IPCC).

CommuniTech became an early pioneer in assisting ITSPs (Internet Telephony Service Providers) in offering their customers endpoint products to enhance their VoIP (define - news - alert) calling experience. This allowed the ITSPs to concentrate on their core strengths while we provided distribution and fulfillment services. We operate nearly all of the on-line stores for the ITSPs that offer VoIP telephony services. We learned early on that there is a direct correlation between higher quality devices and usage. The greater the usage — the greater the revenue for the service provider.

Key elements of our VoIP product strategy include:

- Provisioning of IP endpoints
- Operating third-party fulfillment services of IP endpoints
- Manufacturing the industry leading USB phone/speakerphone, the Claritel phone from Clarisys. Our Clarisys USB phone made us an early entrant in the softphone market. The unique nature of the product led key industry relationships with Cisco and Avaya writing software to integrate their softphones to our Clarisys software. We also have had success with independent softphone suppliers like DiamondWare, EyeP Media, IP blue, and Xten, all integrating their products with ours.
- The North American distributorship of Swissvoice, a high-quality, low-cost VoIP (MGCP and SIP) phone.
- Reselling Polycom, Sipura, and other IP phones
- Distributing of Cisco, i3 Micro, Motorola, and Sipura analog terminal adaptors.

The extensive business that we do overseas, where the cost justification for VoIP is the strongest, has given us the foundation for our VoIP business. While domestic VoIP has many applications, many of these opportunities remain elective. Our strong foreign demand and global customer base gives us unbeatable economies of scale.

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?

NS: As new, inexperienced players enter the market, they are prone to over-hyping and over-promising what new VoIP products and services can do. The danger is setting unrealistic customer expectations that will cause a backlash for the entire industry.

Another significant hurdle will be the emergence of ferocious competition among the three major segments, endpoints, transport, and platforms. Each segment is focused on adding value to their offerings while attempting to commoditize the other segments. In addition, there will be intense pricing pressure within each category (i.e., among IP phone vendors) as well. Unlike traditional products of the past, VoIP products and services require continued manufacturer investments to interoperate and remain compatible with the other required elements in their environment.

> The danger is setting unrealistic customer expectations.

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

NS: We are really focusing strongly on our voice messaging business. We support almost 4,000 voice-messaging platforms. In addition to selling systems, our national field service organization offers support, installation, project management, training, maintenance, and a help desk twenty-four hours a day, seven days a week.

As a VAR specializing in Mitel and SS8 messaging platforms, our greatest opportunities lie in helping our major end users migrate to next generation products that will support a variety of applications that are natural complements to Messaging. This goes beyond Unified Messaging. Applications include IVR (define - news -alert), Instant Messaging/Presence and Speech Recognition. Our core customers want best-of-class capabilities and partners that can support them. We are uniquely positioned to offer it all.

Our specialization leads us to unusually tight working relationships with Mitel and SS8. Both are long term relationships that grew out of working with their common predecessor, Centigram that separated into Mitel's line now focusing on enterprise customers and SS8's that targets Service Providers. Mitel's new platform runs on a Linux operating system and SS8's equipment runs on Solaris.

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

NS: It is very bright, but it is going to take a while to get there. We have the best switched circuit infrastructure in the world and it takes exciting, compelling applications to drive the move to IP. IT

ADVERTISING INDEX