TMC
# INTERNET TELEPHONY

www.itmag.com Volume 8/Number 11 November 2005

The VoIP Authority Since 1998™

**Web 2.0 Meets VoIP 2.0**
Page 8

# Securing VOICE In An IP World

## Also In This Issue:

- Are VoIP Providers Ready to Take Center Stage?
- Cable Operators Face Opportunity, Challenges
- Case Study: VoIP Hits the Real Estate Market

**INTERNET TELEPHONY®**

*Group Publisher and Editor-In-Chief,* **Rich Tehrani**
(rtehrani@tmcnet.com)

**EDITORIAL**
*Editorial Director,* **Greg Galitzine**
(ggalitzine@tmcnet.com)
*Associate Editor,* **Erik Linask**
(elinask@tmcnet.com)
*Contributing Editor,* **Johanne Torres**

**TMC LABS**
*Executive Technology Editor/CTO/VP,* **Tom Keating**
(tkeating@tmcnet.com)

**ART**
*Senior Art Director,* **Lisa D. Morris**
*Art Director,* **Alan Urkawich**

**EXECUTIVE OFFICERS**
**Nadji Tehrani,** *Chairman and CEO*
**Rich Tehrani,** *President*
**Kevin J. Noonan,** *Executive Director, Business Development*

Editorial Offices: 203-852-6800
Customer Service: For all customer service matters, call 203-852-6800.

**ADVERTISING SALES**
**Sales Office Phone: 203-852-6800**
*Sr. Advertising Director - Eastern U.S.; Canada; Israel*
**Anthony Graffeo,** *ext. 174,* (agraffeo@tmcnet.com)
*Advertising Director - Western U.S.; International*
**John Ioli,** *ext. 120,* (jioli@tmcnet.com)

**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.

**SUBSCRIPTIONS**
*Circulation Director,* **Shirley Russo,** *ext. 157*
(srusso@tmcnet.com)
Annual digital subscriptions to **INTERNET TELEPHONY®**: free to qualifying U.S., Canada and foreign subscribers.
Annual print subscriptions to **INTERNET TELEPHONY®**: free, U.S. qualifying readers; $29.00 U.S. nonqualifying, $39.00 Canada, $60.00, foreign qualifying and nonqualifying. All orders are payable in advance in U.S. dollars drawn against a U.S. bank. Connecticut residents add applicable sales tax. For more information, contact our Web site at www.itmag.com or call 203-852-6800.

**EXHIBIT SALES**
**Sales Office Phone: 203-852-6800**
*VP of Publication, Conferences and Online Media*
**Dave Rodriguez,** *ext. 146,* (drodriguez@tmcnet.com)

**READER INPUT**
**INTERNET TELEPHONY®** encourages readers to contact us with their questions, comments, and suggestions. Send e-mail (addresses above), or send ordinary mail. We reserve the right to edit letters for clarity and brevity. All submissions will be considered eligible for publication unless otherwise specified by the author.

**IDENTIFICATION STATEMENT**
**INTERNET TELEPHONY®** magazine (ISSN: 1098-0008) is published monthly by Technology Marketing Corporation, One Technology Plaza, Norwalk, CT 06854 U.S.A. This issue, Volume 8, Number 11 is dated November 2005. Annual print subscriptions: free, U.S. qualifying readers; $29.00 U.S. nonqualifying, $39.00 Canada, $60.00, foreign qualifying and nonqualifying. Periodical postage paid at Norwalk, CT and at additional mailing offices. Postmaster: Send address changes to: **INTERNET TELEPHONY®**, Technology Marketing Corporation, PO Box 21642, St. Paul MN 55121 U.S.A.





**REPRINTS AND LIST RENTALS**
For authorized reprints of articles appearing in **INTERNET TELEPHONY®**, please contact Reprint Management Services at 1-800-290-5460 • tmc@reprintbuyer.com • www.reprintbuyer.com.

For list rentals, please contact Lisa Horder at lisah@l-i-s-t.com or call 914-765-0700, ext. 107.

**A Technology Marketing Publication, One Technology Plaza, Norwalk, CT 06854 U.S.A. Phone: 203-852-6800 Fax: 203-853-2845, 203-838-4070**

Subscribe FREE online at http://www.itmag.com


## The VoIP Authority

*By Greg Galitzine*

# 10 Years On: Walking Among Giants

Avaya (quote - news - alert) recently made headlines when it announced it was marking the 10-year anniversary of its first enterprise Internet Protocol (IP) telephony product. Ten years may not seem like much at first glance, but when one considers how far our industry has come in that short span of a decade, it makes for some interesting perspective.

Despite some slow, post-Internet bubble, “the-industry’s-out-of-business” years, VoIP (define - news - alert) has steadily crept forward, surpassing the waves of early hype, taking its place as the technology to watch going forward. IP Telephony is real, and the fact that the technology is increasingly being deployed in office after office across the United States and indeed the world, is not to be taken lightly. As I said in the very first issue of **INTERNET TELEPHONY®** Magazine, VoIP is more of an evolution, not a revolution. Today, VoIP continues to evolve and there is no doubt in my mind that VoIP is fast becoming THE way we will all communicate in the not too distant future.

Recently, I wrote online how Microsoft and Yahoo! announced that they will make their instant-messaging programs work together, a partnership ostensibly designed to give the combined companies more heft to compete against AOL. The partnership, which would allow users of the two services to exchange messages seamlessly, gives the companies almost as many users (combined) as AOL has on their own. AOL’s AIM, has over 50 million unique users, compared to about 27 million for MSN Messenger and 22 million for Yahoo’s Messenger.

Suddenly you have 50 million people able to see each others’ presence, able to IM each other, and able to speak to each other utilizing VoIP. That’s just the Microsoft/Yahoo! camp. Now add 50 million AOL users, and let’s not forget about the 50 million+ Skype users plus the estimated 135 million registered eBay users.

That’s a lot of VoIP.

eBay’s Meg Whitman recently opined that, in a few short years, users should expect to make free voice calls as part of a larger package of services supported by advertising or transaction fees.

“In the end, the price that anyone can provide for voice transmission on the ‘Net will trend toward zero,” she said.

“Our belief is that the winners in this space will be those that have the largest ecosystem,” Whitman said, meaning “…the largest number of registered users, the largest number of voice minutes, the largest number of developers who develop the platform, the best product that users are willing and want to pay for.”

And there’s only one place to learn about the future of ecosystems the likes of which Meg Whitman envisions. Internet Telephony Conference & EXPO EAST, taking place this January 24-27, 2006 in Ft. Lauderdale is proud to host the first-ever Voice Communities Summit. We have developed a conference program light years beyond any other educational offering out there, one that includes a full day dedicated to learning about voice-enabled communities. I urge you to check out http://www.itexpo.com and see for yourself. As nice as it will be to find yourself in Ft. Lauderdale in January, imagine how much nicer it will be to learn all about the hottest trends in the hottest industry. It’s the very first VoIP event of 2006. I can’t wait. See you there!

Greg

*-Greg Galitzine*, ggalitzine@tmcnet.com

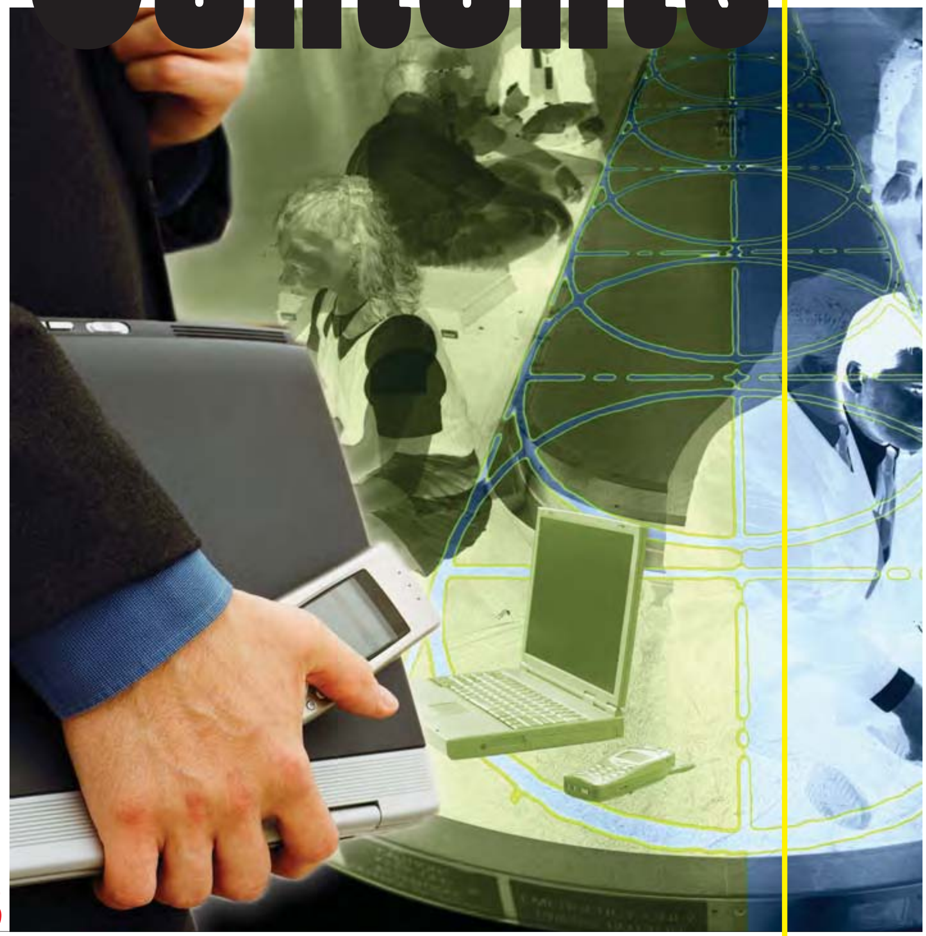



**Go To Table of Contents | Go To Ad Index**

INTERNET TELEPHONY®

Volume 8/ Number 11 November 2005

# Contents



98





Go To Ad Index

# Get centered.

((( No matter what your life is centered around. )))

**Proximiti® Center™ lets you deliver the most powerful set of communications services in the industry today – personalized for every type of customer**

POWERED BY SipStorm®

PROXIMITI®

Forget niche products and narrow market segments... or trying to find someone really interested in saving $10 on their phone service. IP communications is a lot more – and Proximiti brings it to market for any type of customer. Proximiti's IP communications solutions all include its revolutionary Center, making any user feel like the center of the universe. Proximiti solutions range from "Bring Your Own Broadband" Services to IPBXs to Business-to-Softphone Only Services for individual users.

Proximiti solutions include Voice over IP (VoIP), domestic and international numbers, voice recognition, web dialing, personalized address book, and hundreds of other useful capabilities – all customized by businesses and users to fit the way each individual works and plays. So instead of selling cheap calls, you can offer higher productivity and enhanced lifestyles. Powered by SipStorm,® the next generation services engine – Proximiti is communications without boundaries... and your key to getting centered.

**Need proof? Call us at 1-866-363-1422, or contact us at bizdev@proximiti.com**

www.proximiti.com

# Contents

## VoIP BY THE NUMBERS

**54,000**
The number of broadband-enabled hotel properties expected by 2009.

**1 Billion**
Google's initial commitment to the creation of a philanthropic arm devoted to causes that mesh with the online search engine leader's crusade.

**1%**
The percentage of Google's stock and future profit earmarked for the charitable foundation.

**1,500**
The number of XO Communications' XOptions Flex business customers. The lucky company is Villa Financial Services.

## QUOTE OF THE MONTH:

"The truth is, the minute is dead and distance is gone. It's all about access. Vonage, Skype, and now GoogleTalk have taught the world a harsh lesson about radical change in a short period of time. VoIP as a technology within a standalone business model has been introduced to the mainstream, matured, and burned out in three years! VoBB, Voice on the Internet (that's free and for a fee) and now an open source, global free service. Where's the value in voice when the money from minutes is gone? For the end user it's become a give-away service, like e-mail. It isn't going to all happen overnight, but all of the pieces are in place and big names are starting to show their cards."

*– Hunter Newby (page 52)*

## WHAT'S ON TMCNET.COM RIGHT NOW

To stay current and to keep up-to-date with all that's happening in the fast-paced world of IP telephony, just point your browser to http://www.tmcnet.com for all the latest news and analysis. With over 5.9 million unique page views per month, translating into over 617,000 visitors, TMCnet.com is where you need to be if you want to know what's happening in VoIP.

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

**Level 3-Cogent Dispute Puts VoIP Peering In Spotlight**
The ongoing peering dispute between two well-known Internet backbone providers highlights the risks of relying on the public Internet for the mission-critical tasks such as carrying voice traffic.
http://tmcnet.com/186.1

**Sprint Nextel Charges Vonage, Others Infringed On VoIP Patents**
Sprint Nextel Corp. is charging three well-known voice-over-IP (VoIP) service providers including industry darling Vonage with "willfully" infringing on seven patents related to voice packet technology.
http://tmcnet.com/187.1

**Yahoo, Internet Archive To Create Free Collection Of Digital Media**
In what could amount to one of the largest collections of online digital text and multimedia content available, Yahoo Inc., the non-profit Internet Archive and several other groups have launched the Open Content Alliance (OCA).
http://tmcnet.com/188.1

**CLECs Washing Those RBOCs Out Of Their Hair**
To be sure, the RBOCs have never made life easy on CLECs, what with their expensive collocation fees and limits on customer access. With the proposed mega mergers between SBC-AT&T and Verizon-MCI nearing final regulatory approval, CLECs lives don't figure to be any easier.
http://tmcnet.com/189.1

**VoIP Inc. Connects With Stealth's Voice Peering Fabric**
While ILECs aren't exactly quaking in their boots just yet, there is no doubt that VoIP peering is gaining significant traction in the market. Indeed, many an eye was opened when XO Communications announced that it connected its facilities to Stealth's Voice Peering Fabric.
http://tmcnet.com/190.1

**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.

**TMC's Triple Play Channel**
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.



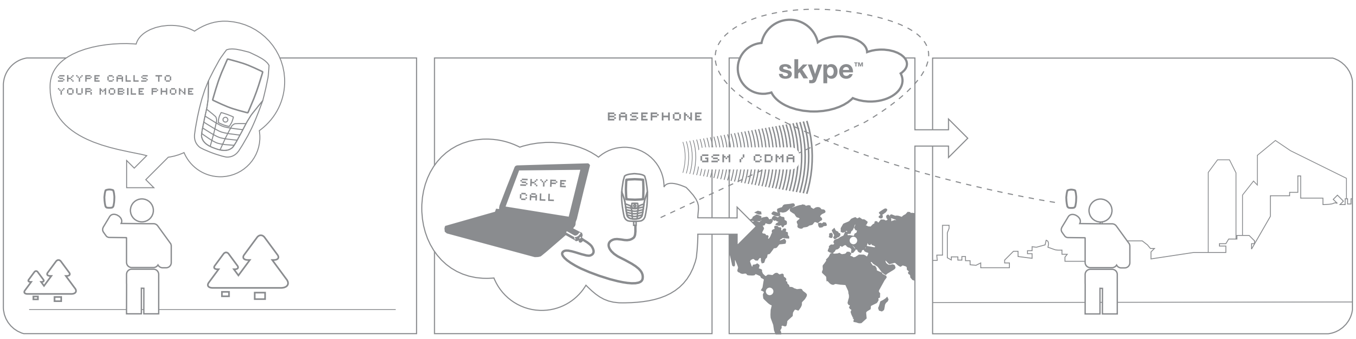
Go To Ad Index

ipdrum
Mobile IP communication

MOBILE SKYPE CABLE

# FREE* MOBILE CALLS - WORLD WIDE!

## CONNECTING SKYPE TO YOUR MOBILE




User wants to call his friend on Skype via his mobile phone

The users call is automatically connected to Skype

The call is transferred across the world

His friend receives the Skypecall on his mobile phone while being away from his computer

### How it works

The IPdrum Mobile Skype Cable connects a mobile phone to the USB and sound ports of a computer. When the software is installed, the handset can be used as a bridge for the Skype application, allowing you to forward calls to the mobile phone you currently use. Similarly, you can use the gateway to place a call using Skype.

### The main features of the IPdrum Mobile Skype Cable:

- Forwarding of incoming Skype calls to fixed- or mobile phone
- Outgoing calls to Skype users
- Outgoing calls to non Skype users using SkypeOut
- Synchronizing with Skype contacts
- Make calls abroad without roaming charges

For more information visit **www.ipdrum.com**

* With a mobile subscription that allows free calls between phones (e.g. family&friends)

By Rich Tehrani

# Web 2.0 Meets VoIP 2.0

The recent Web 2.0 Conference in California has to be one of the most positive growth signs in Silicon Valley in post-bubble times. If you missed it, you are like me. But perhaps also, like me, you saw a number of positive articles written as a result of the gathering as well as a few tempered with a bit of concern about 'irrational exuberance.'

But what exactly is Web 2.0? The broadest definition I have read is it's the next-generation of Web-based applications allowing them to be more desktop-like via AJAX. AJAX stands for Asynchronous JavaScript and XML (define - news - alert) and is basically a way to program Web-based applications that are speedier as they do not need to interact with the server as often.

Another thought on Web 2.0 is that it stands for tomorrow's applications — Applications, which are going to be truly multimedia in nature.

An article in *USA Today* likens Web 2.0 to WD-40 as it is a multipurpose lubricant — it lets the different pieces of computing, software, and the Web slide into one another and even blurs the distinction between users and Web sites.

After reading a number of articles, Web 2.0 seems more of a concept than anything concrete. Perhaps that is what scares me a bit about it. How can we judge if an industry is doing well if it is just a concept. I am afraid that concepts not turning into concrete business plans do, indeed, become the petri dishes where bubbles are concocted.

And yet, perhaps we were too hasty bursting bubbles in the Web 1.0 world. Many of the concepts from Web 1.0 (such as eyeballs) were extremely valid. Hindsight is, of course, 20/20. Look at all the eyeballs Google has and witness how they are monetized. Perhaps Web 2.0 is not something to worry about. Indeed, the investors and technology executives involved in building Web 2.0 all got burned pretty badly by Web 1.0. I would say many of them — not all, of course — have learned their lessons.

**VoIP 2.0 is truly something that will change the way we all communicate and interact with one another.**

I figured, while I am repeating what others are saying about Web 2.0, I might as well throw my thoughts into the mix. For me, Web 2.0 is all of the above plus the advent of interconnectivity between sites and disparate ecosystems that will form larger communities of interest and even more diverse ecosystems.

On the interconnectivity front, companies are becoming more and more free with their data and are allowing legions of developers access to what used to be considered proprietary. For example, a new sort of Web site is popping up that combines data from one or more sites. The term for these new sites is "mash-ups" and they do some very interesting and possibly useful things.

Google (quote - news - alert) has helped the cause by making its Maps API available and using it. Others are designing maps that can show you train stations in your city or the locations of crime scenes in Chicago on a map (chicagocrime.org). Where the concept gets more useful is when multiple sites are queried and tied into one another. A dating site called hotornot.com opened up its APIs and there is now a new site called the Social Software Weblog (socialsoftware.weblogsinc.com/entry/ 1234000533051571/) that allows you to click on icons on a Google map to see single people in your local area. In this way you can screen out the geographically undesirable candidates quickly. Once you decide that your commute distance is acceptable you can see what the person looks like and then arrange to meet.

With the above in mind, how can we define broadly what VoIP 2.0 is? In some ways, it is similar to Web 2.0 and the two concepts are complementary, but I like to think VoIP 2.0 is much more concrete in some ways. I would have to say that VoIP 2.0 is at the cutting edge of innovation and implementation. And, the applications are what make VoIP 2.0 exciting — the integration of VoIP into our lives, into the ecosystem of applications with which we already interact.

More specifically it deals with lots of the latest technologies embracing VoIP. I first wrote about this topic almost a year ago, in the December 2004 issue of Internet Telephony (http://tmcnet.com/191.1). What I would like to do is update that list. Some of the items on my list this year are new and some are the same. I will address all the items from last year's list first.



Go To Table of Contents | Go To Ad Index

### Service Provider Opportunities

Triple Play becomes quadruple play, and so forth: This is still a valid concept. The MVNO, or mobile virtual network operator, opportunity is a real one. However, I recently spoke with the spokesperson for Sprint PCS and it seems unless you have some compelling new feature or can attract a market Sprint can't easily target, they may not want you as an MVNO. Obviously, this makes it tough to deliver quadruple play services. Sprint's strategy may change, however, if other providers become more aggressive in the space. I still believe quadruple play is coming — I am just less sure who will provide the wireless component.

IPTV (define - news - alert)is a natural service to deploy for today's service providers. Figuring out how to get the content is the problem with this market. At some point, I hope a central clearinghouse will allow the buyers and sellers to come together in the content distribution world.

### The Device Sells The Service

The MP3 market was plain boring before Apple came onto the scene. The RAZR phone really livened up the wireless market. We still need really cool VoIP phones to help differentiate service. I have yet to see one. Steve Jobs, I have been asking for a VoIPOD for a year now… Will you help us out?

### VoIP E911

When I last wrote about it, 911 was not in the headlines relating to VoIP. This is what I wrote:

*I have said it before and I will say it again: if we don't get our act together soon as an industry, we will have some serious headaches to contend with. The positive press friendly to VoIP that we witnessed for a year will vanish the moment someone is injured, or worse, because there is a problem with VoIP and E911 connectivity.*

*The current state of 911 over today's VoIP providers is not good. The incumbents aren't as much of an issue as the newer carriers who transfer 911 calls to lower priority administrative lines in PSAPs. E911 over VoIP can be much better than PSTN 911. We need to come together as an industry and discuss the challenges and standards issues and make sure that e911 over VoIP becomes a reason to adopt and not a reason to pass on VoIP.*

*I consider this a stumbling block that needs addressing on our way to achieving VoIP 2.0. Companies like Vonage, who use technology from an innovative company called Intrado, are taking bold steps to ensure the safety of their customers. They should be commended for their efforts and others need to follow.*

Since I wrote this, the FCC has gotten involved and mandated compliance. I am pretty sad that the industry didn't act sooner to make E911 a reality. Hopefully, this is finally resolved.

### Taxation/Regulation

The Universal Service Fund is still in sad shape and government regulators may still try to come after our industry and saddle us with huge taxes and fees to fill the USF. In my opinion, this concept won't work. Rather, the government will push people to free VoIP services if they drive the cost up artificially.

### VoIP Peering

When I wrote about this last year, most people had no idea what it was. A week ago, I declared 2006 the year of VoIP peering. CLECs are all clamoring to connect with one another to cut down on fees and improve call quality. Since my recent proclamation went out, I have received calls from countless people in the industry who agree with my analysis. Let's see how 2006 plays out. I am pretty sure I have called this one correctly.

### Open Source

This market just grows and grows. In the last few months I have mentioned how much traction this concept has gained and, more importantly, how PBX and ACD manufacturers are working to take Asterisk systems and augment them and sell them as entry-level systems of their own.

### The Reseller Opportunity

This is still here. Resellers stand to make a nice living selling to service providers and enterprise customers alike. Internationally, VoIP is an easy sell, which is why we see so many of these resellers coming to events like Internet Telephony Conference & EXPO show after show to get a handle on how they can make profit in this fast-growing market.

**I declared 2006 the year of VoIP peering. I am pretty sure I have called this one correctly.**

### Peer-To-Peer

This market has heated up tremendously. One of the biggest deals of the year on Wall Street was eBay's purchase of Skype, the world's largest P2P service provider. Moreover, the purchase of Nimcat Networks by Avaya has fueled interest in enterprise P2P systems. It is only a matter of time before Dell starts selling P2P-based phones. Avaya may supply the phones to Dell, as well as





Go To Table of Contents | Go To Ad Index

potentially a gateway that will tie large numbers of the P2P phones together for more centralized control and increased stability and security.

### WiFi Telephony & WiMAX

This segment is as hot as ever. WiMAX (define - news - alert) is still in its infancy, but WiFi phones are coming out of the woodwork. I would add dual mode phones to the list.

### Ambient Telephony

The ability to stay connected to others indefinitely via VoIP is still a fascinating one. The concept is still just that— a concept — and I haven't seen it take off yet. At a recent conference, the idea came up that workgroups in corporations could use this idea to be connected informally at lunches — even if they are geographically separated from one another. Let's see how this plays out.

### UNE-P To VoIP

CLECs are basically all moving to VoIP or closing shop. This is a booming market for equipment providers.

Now for the new additions…

### IMS

IMS, or IP Multimedia Subsystem, is a set of technologies that will give the ability to have wireless and wired networks work with one another using SIP so subscribers can seamlessly transition from one network to another. In addition, tomorrow's IMS networks could allow developers access to subscribers, meaning a network operator can generate revenue from companies that produce compelling applications and want to run them on their network. IMS is far-reaching and ambitious. Everyone in the industry is behind it. I am a bit skeptical because any time there is 100 percent acceptance of a next generation technology I start to think about OS/2. If IMS delivers as promised or even close, it will be a central part of VoIP 2.0.

### Voice Communities

Another leading edge concept. I thought about this seriously once Skype was acquired by eBay. Voice communities will pop up whenever buyers meet sellers or wherever communities naturally congregate online. ESPN is in a great position to host talks on lots of topics, like who will win the World Series or the World Cup. People would likely pay to listen to these discussions or advertisers may pay to sponsor them. These conversations will be recorded and podcasted. Others will be able to comment via text or voice. Search engines will convert these discussions to text so you can find specific conversations.

## VoIP 2.0 Live

It almost doesn't seem possible that yet another Internet Telephony Conference & EXPO is upon us, but it is, and we can't be more excited. This event burst out of the hotel venue it was in last year and is now calling the Ft. Lauderdale Convention Center its home. Last year we had over 6,200 attendees at this show and this year we expect many more. Sign up now to what will be the first major VoIP show of 2006. We are so thankful to you for making our IT EXPO so successful in the past. As a sign of our appreciation, if you sign up for the January 2006 conference by December 2, 2005, you'll not only save up to $1,000, but you'll get a free iPod nano in the process! Check out **http://www.itexpo.com** for details. IT

### Just In Time Communications

You have heard me speak on this at countless seminars and expos across North America. I really believe we are on the verge of communications going through a dramatic transformation. The exact same one the manufacturing industry went through when productivity was squeezed to the max and new levels of efficiency were reached. We will call this Just in Time Communications or JiTC.

### VoIP 2.0 Here To Stay

VoIP 2.0 is here to stay and it is so much more than what VoIP 1.0 was. It is truly something that will change the way we all communicate and interact with one another. It describes the cutting edge of communications innovation and implementation. I hope I have given you enough of a visionary description of what VoIP 2.0 is as well as some concrete ideas on which you can build your business. You have heard me say before that it is good to be in VoIP, and that statement remains true. There continues to be more opportunity for all of us. What companies will win the VoIP 2.0 race? Just like a real road race, it will be the companies that move fastest and are the most innovative. To all the companies jockeying to be at the lead of the VoIP 2.0 race, good luck crossing the finish line first! IT


We're Putting VoIP into the Hands of Small Business
TalkSwitch telephone systems deliver big business features at small business prices. Customer installable and compatible with analog phones, TalkSwitch hybrid VoIP/PSTN systems are designed specifically to fit small businesses with single or multiple locations. To find out what TalkSwitch can do for your business, call us toll-free at 1.888.332.9322 or visit us at www.talkswitch.com.
talkswitch
Power
Line 1
Line 2
Data
Line 3
Line 4
WWW.TALKSWITCH.COM
TOLL FREE 1.888.332.9322
talkswitch®
ANSWERS WITH INTELLIGENCE


Industry
NEWS

# Polycom Delivers SIP-Based Attendant Console

Polycom, Inc., (news - alert) recently announced a full-featured, SIP-based attendant phone console for executive assistants, receptionists, secretaries, and administrative assistants who manage and monitor multiple simultaneous calls. The SoundPoint IP attendant phone console comprises the new SoundPoint IP 601 SIP desktop phone and up to three SoundPoint IP Expansion Modules.

Designed to improve productivity and effectiveness of telephone attendants, the attendant console allows users to accept, screen, dispatch, and monitor up to 24 concurrent calls. These advanced call-handling capabilities help augment telephone attendant productivity and performance and improve customer satisfaction through faster, more accurate call routing and fewer dropped calls.

"The SoundPoint IP attendant phone console gives our customers the familiar push-button interface as traditional PBX and key system attendant console solutions, and through SIP delivers expanded capabilities that were not available before," says Robert Haley, senior engineer with Alltel. "The Polycom attendant phone console offers multiple ways for managing calls, which enables users to adapt the phone to their preferred style. This complements Polycom's existing line of VoIP phones and fulfills the previously unmet customer need for a desktop phone solution that effectively replaces legacy attendant consoles."

The SoundPoint IP 601 SIP desktop phone supports six lines and provides a powerful tool for users requiring an advanced feature set. Based on the award-winning SoundPoint IP 600, the SoundPoint IP 601 delivers high-quality voice with Polycom Acoustic Clarity Technology, the expandability to support up to three SoundPoint

IP Expansion Modules, and advanced functionality including multiple call and flexible line appearances, HTTPS secure provisioning, instant messaging, presence, custom ring tones, and three-way local conferencing. The phone features built-in, dual-mode, auto-sensing Power over Ethernet circuitry.

The SoundPoint IP Expansion Module augments the user interface of the SoundPoint IP 601 with a high-resolution graphical LCD and 14 illuminated multi-functional keys that can be automatically configured as a line registration, call appearance, or speed-dial/DSS. The SoundPoint IP Expansion Module is a true "plug-and-play" device that requires no configuration or dedicated powering as signaling and power are provided by the host SoundPoint IP 601 phone.

When equipped with up to three SoundPoint IP Expansion Modules, the SoundPoint IP 601 delivers the advanced call handling capabilities and enhanced user interface of a high-performance attendant phone console — featuring up to four high-resolution LCDs and up to 48 illuminated multifunctional keys — that allows users to effectively accept, screen, dispatch, and monitor up to 24 concurrent calls.

http://www.polycom.com

## AltiGen Dials Up New VoIP Phones

**By David Sims**

AltiGen Communications, Inc., (news - alert) has announced the release of its IP 710 business VoIP telephone, which AltiGen CEO Gilbert Hu describes as filling "the gap between advanced business features on the telephone and IP telephony."

The 710 gives users single-button access to voicemail, activity/presence selection, voicemail greeting selections, call recording, call conferencing, call transferring, and even placing calls to employees in other countries.

The four-line, backlit liquid crystal display of the IP 710 is capable of displaying time, Caller ID name and number, real-time call center workgroup statistics, do-not-disturb, and call forwarding status.

It can be personalized with 15 backlit user-defined keys for "any combination of configurable features like, but not limited to, speed-dialing, extension busy/ringing appearances, call appearances, line appearances, and workgroup activity status," company officials say.

The IP 710 is compatible with third-party hands free headsets using AltiGen's dedicated headset answering button and amplified industry standard RJ9 and 2.5mm headset ports and also provides 14 combined traditional and melodic ring tones, which can be assigned to line and extension appearances on the programmable keys.

This past May AltiGen announced a partnership with Commtech, a distributor of Internet, networking and security products in Ireland, to distribute AltiGen's IP phone systems and call center products to the Irish market as part of what company officials called "AltiGen's continuing strategic plans for growth in Europe."

Commtech will be a specialist distributor of AltiGen technology and a component in AltiGen's plans to sell VoIP and IP convergence products to the small and mid-sized Irish enterprise market. Commtech will also provide independent advisory and installation services to support the AltiGen reseller channels.

**http://www.altigen.com**

## FacetPhone Version 3 Announced

FacetCorp (news - alert) has announced feature additions to FacetPhone. With the release of FacetPhone Version 3, significant new features have been added for both call center and multi-location businesses. Version 3 also expands the computer telephony integration capability of FacetPhone, adds T1 support, and offers numerous other features.

Companies with call centers have a host of new options available to help manage their businesses. In addition to call recording, FacetPhone V3 supports call monitoring and call "barge-in." Supervisors and managers may monitor calls, and may optionally record calls or barge in to calls "on-the-fly" as needed. The FacetPhone graphical user interface (GUI) has a new, separate window for the automatic call distribution queues, and new status information to show if a call is in the queue or is being monitored.

Companies with multiple locations now have a tremendous new tool to help their offices communicate easily and cost-effectively. FacetPhone customers may continue to run their local phone system from a local FacetPhone server/controller. But with FacetPhone V3, servers in different locations are aware of, and cooperate with, other FacetPhone servers over the wide area network. Through FacetPhone's cooperating server functionality, employees may call co-workers in a remote office just as if it was a local extension.

FacetPhone Version 3 further expands upon the capability provided with the UTAPI (Universal Telephone API) protocol. The original UTAPI implementation allowed customers' Linux and UNIX data base applications to quickly and easily dial the phone and retrieve the CallerID from FacetPhone so as to generate a customer database "screen pop." With Version 3, the UTAPI protocol has been extended to handle customer applications running on a Microsoft Windows platform.

FacetPhone Version 3 has many other new features, including:

- Call Detail Reporting (CDR) Web interface for flexible and quick ad-hoc reports
- Soft phone support with dynamically assigned IP addresses
- Dialing restrictions — both system-wide and station level
- SIP T1 PSTN Gateway Support
  - Direct Inward Dial (DID)
  - Carry forward original CallerID on forwarded calls
- Optionally display the called group name on the telephone CallerID display (and the GUI) rather than the CallerID data
- Automatic GUI user login and Telephone User Interface (TUI) login

**http://www.facetcorp.com**

## Switchvox Launches SOHO IP PBX System

Switchvox (news - alert) has unveiled Switchvox SOHO, described as a full-featured, easy-to-use IP PBX solution starting at an affordable price of $995.

Switchvox's flagship product, Switchvox, has been upgraded to offer a more feature-rich experience for larger companies. Customers with an existing support contract will be automatically upgraded to the full-featured version of the product and continue to receive updates. Customers without a support contract will be given a 30-day option to purchase a support contract in order to be upgraded to the advanced version.

"We have been listening closely to our customer feedback and, by adding more advanced features to our flagship product, we are able to meet the unique requirements of growing companies," says Joshua Stephens, CEO of Switchvox. "With the introduction of Switchvox SOHO, we are continuing to give small businesses the opportunity to deploy an affordable, easy-to-use IP PBX system. As the market continues to grow, we will work with customers to meet their demands and provide affordable, feature-rich communications solutions."

According to research firm In-Stat, between now and 2009, server-based IP PBX shipments will grow from 9.5 million lines to 28.1 million, at which point such systems will represent over 91% of total PBX shipments.

Switchvox is built on open source standards. It is designed to work with all SIP compatible hardware and software phones as well as standard analog handsets. Calls can be sent over the Internet to VoIP providers worldwide and directly to remote corporate offices, by peering Switchvox systems using the SIP or IAX protocols.

Advanced features in the upgraded version include:

- *Conference bridge:* Set up advanced conference rooms with user-defined private access codes.
- *Busy call rules:* Set up different messages to play when users are on a call versus normal voicemail recordings.
- *Unified messaging:* Users now have advanced control over their voicemail messages from their e-mail.
- *Extension groups:* Administrators can now group extensions into business groups to help organize communications.
- *Voicemail forwarding:* Switchvox can automatically forward messages to other users or groups on the system when a message is received.
- *Call API:* Allows external programs to send commands to Switchvox and these commands trigger the IP PBX to perform certain tasks.
- *Call center statistics:* Improved queue statistics with real-time graphs give valuable insight into the performance of agents.
- *Paging and intercom:* Contact groups of users easily using one-way paging or two-way intercom features.

Switchvox SOHO is available immediately starting at $995 and the advanced Switchvox IP PBX product is available for $2,495.

**http://www.switchvox.com**

## Announcing BlueNote Networks

BlueNote Networks (news - alert) announced the official launch of the company and its SessionSuite product platform. Based on open standards, SessionSuite is an interactive communication platform designed as a component-based, distributed software solution, with an eye towards providing a low cost alternative to today's VoIP solutions.

SessionSuite brings together real-time, interactive communications and service-oriented architectures to advance business process productivity. According to a company announcement, BlueNote was created around the premise of fulfilling unmet customer needs for globally-reachable services via efficiently deployable and cost-effective IP telephony products.

Tom Burkardt, Chairman and CEO of BlueNote has some experience in telecom; he has been CEO of Wavesmith, he founded Castle Networks, he served as COO of Unisphere Networks, he was the business unit director at Cascade Communications, founder and director of Cabletron Systems' IBM Interconnectivity Products Business Unit, and held various engineering and management positions at Wang Laboratories. He also served as a Venture Partner at Fidelity Ventures, where the concept of BlueNote came together.

"While at Fidelity Ventures I became aware of a project to fulfill Fidelity Investments' desire for a cost-efficient Internet implementation, allowing global reach and increased flexibility. It turns out that these goals are common to many other enterprises," said Burkardt. "Simply put, enterprises wonder why they can't leverage the Internet along with their existing IP infrastructure to realize the same benefits that consumers enjoy from services such as those offered by Vonage and Skype. This is the issue that BlueNote Networks is addressing."

According to a company press release, "The traditional IP/PBX is based on a mainframe-like architecture that is fast becoming legacy infrastructure. Businesses with communication-intensive workflows are attempting to integrate communications with data center applications to accelerate business processes. Companies need an easy-to-integrate platform and simple, yet powerful, applications that change the way businesses communicate and operate. The traditional IP/PBX does not easily meet the demands of decentralized or distributed communication needs."

**http://www.bluenotenetworks.com**

## Oki Electric Offers New Mid-sized IP-PBX, IPstage EX300

Oki Electric Industry Co., Ltd. (news - alert)announced the launch of its latest IP-PBX model "IPstage EX300," for mid- and small-sized offices. The release adds a new multifunctional IP phone to Oki's lineup, which includes Oki's original "eSound" (excellent sound) technology.

"As IP networks penetrate into the office environment, demands to link them with various applications and be SIP-compliant have been increasing. This IPstage EX300 is the answer to such needs, responding to the changes we, as the market leader, recognize," says Masashi Tsuboi, President of IP Systems Company at Oki Electric. "Not only can users experience the enhanced service functions of the IP-PBX, but they can also enjoy easy-to-use IP phone with excellent sound quality and advanced functions."

IPstage EX300 is a successor to IPstage EX100 launched in September 2002. The newly added IP multifunctional telephone, the MKT-IP-30DKW, has Oki's high sound quality "e-sound" technology built-in. The phone is compliant with IEEE802.3af and enables users to set up to 30 functional keys. The phone also includes address book and call record functions. In addition, SIP terminals, such as Com@WILL Softphone and N900iL the FOMA mobile phone and wireless LAN handset, can be connected to the system. Combing the new phones with Oki's wireless LAN access point, MWINS BR2100 Series, users can establish a mobile centrex system at a low cost.

Users can also include Oki's IP-compliant PHS (Personal Handyphone System) cell station in the system, which eliminates the need to install telephone systems at each location, and can establish a low-cost mobile environment by controlling the PHS handsets at satellite offices via IP networks.

http://www.oki.com

# LineSider Communications Launches SecureVoIP

LineSider Communications (news - alert) has announced the availability of SecureVoIP, a policy-based VoIP security service. LineSider's SecureVoIP solution is designed to provide the enterprise customer with a policy-based security management and network control capability to automatically provision, implement, and manage VoIP communications, to the device and user application level.

LineSider Communications has deployed SecureVoIP at a number of enterprises, including Omni Capital Group, Centennial Insurance, and DecisionOne.

The solution was created to provide any enterprise — particularly security-sensitive industries such as healthcare, financial services, and government agencies — a policy-based security management and network control capability critical in managing converged VoIP, data, and video enterprise networks.

"Security, as well as the reach and economics of the Internet, were critical considerations when selecting and implementing a VoIP platform," says Neal Bibeau, CEO of DecisionOne. "Ensuring our VoIP traffic was as secure as our data network was imperative in order to take advantage of network convergence."

At the core of LineSider's SecureVoIP solution is WirePower Converged Control technology, a powerful network operating and directory management system that offers a set of policy features and management capabilities. These capabilities create a secure VoIP networking environment, provision new sites and end users on-the-fly, and provide the management tools critical in controlling a converged enterprise networking environment. SecureVoIP offers Real-Time Security encryption of Session Initiation Protocol (SIP) call processing traffic and Real-Time Protocol (RTP) audio streams without compromising voice quality or network performance.

"As VoIP becomes a more popular means of communication, enterprises will need to look at ways of making it more secure," says Elka Popova, industry analyst with Frost & Sullivan. "Our predictions at Frost & Sullivan are by 2010, IP PBX and hosted IP telephony will account for about 30 percent of enterprise CPE and hosted lines, respectively. As those numbers become reality, there is no question that the need for security in the VoIP (define - news - alert) space will become a top-of-mind issue for any size enterprise."

"Security management is critical to driving widespread adoption of VoIP in the enterprise market; it is one of the top concerns among CIOs who are considering VoIP and seeking a way to reduce their dependency upon costly frame relay and MPLS networks," says Harley Stowell, CEO of LineSider. "Our mission at LineSider has been to provide businesses with peace of mind and quality of service through our SecureVoIP platform, while providing a single interface to manage it all. We anticipate that before long, every enterprise using a VoIP network will seek the necessary security safeguards."

LineSider's Zero-1 Enterprise product line, which includes the SecureVoIP offering, delivers Real-Time Security of SIP and RTP, through the use of Policy Enforcement Points (PEPs) located at customer site locations and the BroadSoft VoIP platform, BroadWorks through New Global Telecom (NGT). LineSider provides policy administration and management of the PEPs for the delivery of secured business-class VoIP and IP VPN networking. SecureVoIP is enabled for site-to-site calling across the enterprise network as well as securing the VoIP call from customer premises, through the Internet, to BroadWorks before the call is handed off to the PSTN.

**http://www.linesider.net**

# The Next - Generation of Telephone Adapters
# Intelligent Multimedia Devices

## The Power of Vision

## the Remarkable way to stay in touch !

IMD-300

### 3-Way Video Calling, World-Wide Free Internet Voice & Video Calls

- Crystal Clear Voice & Video Calls.
- 3-Way Video Conferencing.
- Enhanced Voice services: 3 Way Calling, Caller ID, Call Waiting, Call Forwarding etc.
- Quality of Service (QoS) for excellent voice quality.
- Free Voice & Video Calls up to 5 other users with no monthly charges.
- Portable device.
- Universal Plug and Play.
- Routing capabilities.
- NAT traversal.
- Fax pass through and T.38 fax relay.
- Supports two G.723.1, G.729A or T.38 fax calls simultaneously.


## TRULY PLUG & PLAY Easy Installation!

Visit InterEdge's website to find more about InterEdge's new VoIP Dial-Up Adapter

**InterEdge Technologies, LLC.**
sales@inter-edge.com
Tel : 1-866-593-2700

InterEdge™ Technologies

http://www.inter-edge.com

# Telcordia Launches Maestro IMS Portfolio

Building on its growing global momentum in IP Multimedia Subsystem (IMS), Telcordia (news - alert) recently announced the Maestro IMS Portfolio. A powerful set of wireless, wireline, and converged products, services, and applications, the Maestro IMS Portfolio enables carriers to offer any service, over any network, via any device.

The Maestro portfolio is a major IMS initiative for Telcordia as it helps operators fully exploit the emerging capabilities of the IMS architecture and the wide interest and deployment of service delivery environments. With this portfolio, carriers can capture new market opportunities, strengthen competitive position, and rapidly offer revenue-generating advanced services that subscribers demand.

"IMS is the path to true convergence in the industry and our Maestro portfolio will make it a reality," says Scott Erickson, President, IMS Service Delivery Solutions, Telcordia. "Global operators are already relying on Telcordia to guide them through the complexities of adopting IMS so that they can remain competitive, maximize their network and find new ways to generate revenue."

As a result of its Elementive approach, which offers products and services that are open, flexible, and configurable, Telcordia provides an IMS-compliant and ready-to-deploy portfolio that incorporates its access independence, strong knowledge of OSS systems integration, and exceptional reliability and flexibility in service platforms. The Maestro portfolio includes new products and upgrades of existing ones, complemented by Telcordia's professional services capabilities to enable operators to develop and launch converged services at their own pace. Telcordia Maestro portfolio products, including the Telcordia Converged Application Server and Telcordia Converged Real-Time Charging, are already deployed globally at Oi, Telus, Swisscom, Tata Teleservices, Virgin Mobile USA, and others.

"IMS is clearly where the market is heading and represents significant revenue opportunities for vendors and operators alike," says Sanjay Mewada, Vice-President, Telecom Consulting & Research at The Yankee Group. "As competition intensifies and consolidation continues, the ability to quickly deploy profitable services like push-to-talk and personal information services, will be critical for customer retention and revenue growth."

The Maestro portfolio is designed to enable wireless, wireline, cable, and converged operators to realize the broad advantages of complete convergence and to benefit from added revenue potential, greater network and operational efficiencies, reduced churn, and increased customer satisfaction and retention. There are also distinct benefits for each market segment. For instance, wireless carriers can realize increased speed-to-market for new services, protection of their existing customer base, and increased ability to compete with emerging carriers for enterprise customers. Wireline carriers can be better positioned to offer seamless wireline and wireless services to their customers. Mobile Virtual Network Operators (MVNOs) can experience increased speed-to-market and simplified service creation. Cable operators can evolve their content delivery and charging capabilities with a converged infrastructure.

http://www.telcordia.com/maestro



Options
Back

# Service Providers Doing A Poor Job Of Selling VoIP To SMBs

**By Ted Glanzer**

It is no secret that Voice over IP service providers have not tapped the enormous potential of the small- and medium-sized business market.

What appears to be a secret is the reason why.

Indeed, according to a study by technology consulting firm Savatar, (news - alert) service providers have done a poor job marketing and selling VoIP to the 1.8 million or so SMBs in the United States, focusing too much on features rather than the service's reliability, lower cost, and ease of administration.

"Why no one has figured it out is shocking," John Macario, Savatar CEO and study co-author, told TMCnet in a telephone interview.

The study, in which 300 SMB decision makers were surveyed, revealed that the SMB market is ready to migrate to IP telephony; that market, however, is poorly educated with regard to VoIP service.

While 75 percent of the SMBs surveyed are generally or somewhat familiar with VoIP, a whopping 25 percent are "unfamiliar" with VoIP and its advantages.

Another major problem is that SMBs are all over the map with regard to identifying who is a VoIP provider.

Indeed, the survey data reveal that 25 percent of the respondents said that nontraditional telecoms are VoIP providers; 17 percent said telcom equipment providers are; and 10 percent thought that ISPs are.

The general confusion is highlighted by the following: 13 percent of the respondents viewed cable companies as VoIP providers, even though they are not major players in the SMB market as of yet.

In what can only spell bad news for the Verizons and SBCs of the world, just 14 percent of the respondents said that traditional telecoms are VoIP providers, just one percentage point above the 13 percent who responded that they were not sure who offered VoIP.

"[SMBs] don't know who to turn to," Macario said. "It is a marketing and selling failure."

Service providers, according to the study, are not doing themselves any favors by having sales forces that are not educated on their VoIP offerings and by having Web sites that do not highlight their VoIP services.

Also, Macario said that if someone is interested in obtaining VoIP service, "It is incredibly difficult to find a person to talk to."

The end result is that providers who understand the market have a unique opportunity to cash in on the chaos.

"The traditional telcos do not understand what it takes to sell this stuff," Richard Grange, CEO of study co-sponsor New Global Telecom, said in a telephone interview. "It takes a new approach."

The telcos do not have effective channels to sell to market, Grange said. He added that, instead of direct sales forces or telemarketing campaigns, telecoms should be using effective channel partners, such as NGT, to deliver the product.

NGT has over 45 service provider customers and 39,000 end-user seats, 75 percent of which are business seats.

"Better training and understanding the portfolio is really what is needed," Grange said. "We've spent a lot of time working on the delivery process."

And while features such as click-to-dial and remote capability are not important to SMBs initially, they become an important issue downstream, Macario said.

"Those features become very sticky," Grange said. "People love that stuff."

**http://www.savatar.com**

## Qualcomm Turns To Philips For Dual Mode Chipsets

**By Robert Liu**

Qualcomm (quote - news - alert) announced recently that its Mobile Station Modem (MSM) chipsets will support Philips' wireless local area network (WLAN) module allowing users to connect to 802.11b and 802.11g hotspot networks. However, while dual-mode phones are popular in Asia and Europe, it remains a big question whether North American mobile operators will embrace handsets utilizing both WLAN and cellular technologies. Operators have already invested billions to build out their own high-speed networks based on EV-DO (Verizon, Sprint) or WCDMA (Cingular, T-Mobile) protocol.

Qualcomm said the dual-mode capabilities will initially be supported on the MSM6550 chipset. The 802.11b/g-compatible broadband capabilities enabled by Philips' WLAN module are scheduled to be commercially available by the end of 2005.

"This strategic relationship delivers a number of new possibilities, which we look forward to making a reality," says Mike Concannon, vice president of strategic products for Qualcomm CDMA Technologies.

In order to make the solution more operator-friendly, the company says the integrated solution will offer connectivity to WLAN networks as well as to existing cellular networks on both CDMA2000 and WCDMA (UMTS) networks.

**http://www.qualcomm.com**

## FCC Chairman Calls For A More Resilient Communications Network

**By Ted Glanzer**

The destruction left in Hurricane Katrina's wake demonstrates the need for a more resilient and redundant communications network in the United States, Federal Communications Chairman Kevin Martin told a U.S. Senate subcommittee recently.

Martin reported that, as a result of Katrina, more than three million customer telephone lines were knocked down in Louisiana, Mississippi, and Alabama; thirty-eight 911 call centers were out of service; one thousand wireless cells were inoperable, and a hundred broadcast stations were off the air.

"It was at times like these that we were reminded of the importance of being able to communicate," Martin said. "While no communications network could be expected to remain fully operational in the face of a direct hit from a Category 4 or 5 hurricane, that fact was little consolation to the people on the ground."

Martin offered several suggestions to the members of the Senate Committee on Commerce, Science, and Transportation to improve communications services in the event of another disaster, including the following:

- Create a comprehensive emergency alert system that enables national, state, and local officials to reach affected citizens more efficiently.
- Require all service providers to comply with the FCC's 911 rules, "regardless of whether the provider is wireless, wireline, cable, or VoIP," Martin said. "The 911 system is quite literally one of life or death. It is critical to our nation's ability to respond to a host of crises."
- Ensure that all Public Safety Answering Points are redundant.
- Enhance network resiliency by having communications providers develop and use best practices to maintain service in the event of a disaster.
- Provide first responders with an interoperable, mobile wireless communications system, using multiple, flexible technologies and truly mobile infrastructure.

"If we learned anything from Hurricane Katrina, it is that we cannot rely solely on terrestrial communications," Martin said. "Smart radios would enable first responders to find any available towers or infrastructure on multiple frequencies, and WiFi, spread spectrum, and other frequency-hopping techniques would enable them to use limited spectrum quickly and efficiently."

"We also should take full advantage of IP-based technologies to enhance the resiliency of traditional communications networks," Martin said. "IP technology provides the dynamic capability to change and reroute telecommunications traffic within the network. In the event of systems failure within the traditional network, greater use of these technologies will enable service providers to restore service more quickly and to provide the flexibility to initiate service at new locations chosen by customers."

**http://www.fcc.gov**

## Report: VoIP, Cost Reduction Driving Fixed-Mobile Convergence

**By Ted Glanzer**

The VoIP industry and cost reduction are the two major drivers behind the fixed-mobile convergence (FMC) market, where a consumer has dual-use, home/mobile phones, according to a new ABI Research study.

The report, Fixed-Mobile Convergence: Comparative Business Plans, Implementation Scenarios and Capital Expenditure, includes "a forecast of the FMC market through 2010, outlines the technologies, explains the benefits for subscribers, and analyzes business scenarios that make FMC attractive to operators," according to an ABI Research press release.

"The case for FMC rests on the availability of low-cost, dual-use (cellular and WLAN) handsets," said ABI analyst Ian Cox in a prepared statement. "The first models are nearing commercial launch, and their prices should be competitive with conventional mobile handsets early in 2006. That will be the trigger for offering the service."

Two standards will lead the way: UMA for the consumer market and SIP for enterprises according to the release. FMC will most likely catch on overseas first.

"We expect FMC to take off in Europe and Asia rather than in North America because of the greater prevalence of GSM (define - news - alert) and 3G services in those regions," said Cox. "However, any operator using a suitable network can gain a competitive edge by early adoption of FMC, and . . . up to a fifth of all broadband subscribers will be taking advantage of FMC's convenience and lower costs by 2010."

**http://www.abiresearch.com**

## XO Joins Stealth's Voice Peering Fabric

**By Johanne Torres**

XO Communications, Inc. (news - alert) has interconnected with Stealth Communications' (news - alert) Voice Peering Fabric (VPF), a peering network that allows members to exchange VoIP traffic without relying on traditional telephone networks, the companies jointly announced on Tuesday.

The VPF allows the trading of calls on different VoIP networks without the need to be traversed through the public switched telephone network (PSTN). "The VPF is a distributed Ethernet network that functions as an exchange or meet-point for VoIP traffic by allowing enterprises and service providers to establish peer-to-peer connections in a secure, quality-of-service environment," notes the companies' news release.

"VoIP providers need access to the traditional phone networks and, through the VPF, XO can offer members a cost-effective way to deliver their VoIP calls to the public switched telephone network," says Ernie Ortega, president of carrier sales at XO Communications. "With our national IP network and VoIP infrastructure, XO can deliver VPF members' calls to any domestic location on the public switched telephone network through a single connection to our network."

The agreement will enable XO to belong to the VPF ENUM Registry and exchange VoIP traffic directly with other VPF members. XO will also be able to offer VoIP Origination and Termination services nationwide through the VPF to its members needing to deliver VoIP calls to the PSTN.

"We are very excited to have XO to join our community. Their participation in the VPF Minutes Market will no doubt provide competitive services and add value to our marketplace. XO's use of the VPF ENUM Registry will decrease their operational costs and benefit our members by allowing VoIP calls to remain within the IP domain." says Shrihari Pandit, president and CEO of Stealth Communications.

XO made news earlier, when it was picked by SunRocket for VoIP termination services.

By selecting XO to terminate VoIP traffic, companies virtually hand over their VoIP calls directly to XO for termination to the public switched telephone network (PSTN).

That agreement will allow SunRocket's voice traffic to be routed across the XO OC-192 class of service-enabled networks at a higher priority level. The traffic will pass through an XO softswitch gateway and will be carried across the XO IP network to an XO softswitch gateway nearest the final destination of the call. The process will convert the traffic by a XO softswitch to a format accepted by the terminating local service provider.

**http://www.xo.com**
**http://www.stealth.net**

## AOL Rolls Out TotalTalk VoIP Service

AOL (quote - news - alert) announced that it will begin its roll-out of a new VoIP service, TotalTalk. The TotalTalk service transforms any high-speed Internet connection into a quality phone service that is simple to use and affordable, with savings of up to 40 percent off monthly phone bills, compared to traditional landline service. No AOL subscription is required and current AOL Internet Phone subscribers will receive this upgrade to the new and enhanced service automatically.

Combining the savings and convenience of a VoIP phone with a suite of advanced communications features, available at no additional charge, TotalTalk includes: a "soft phone" (free PC-to-PC calls and PC-to-Phone calling capabilities via a Preview Edition of the AIM Triton client), unified voice, e-mail and instant messaging, enhanced voicemail and call management capabilities, and the ability to make and receive calls on a home phone line from anywhere users have access to AIM.

The service includes, at no additional cost, many phone features such as: Call Waiting, Caller ID, 911 emergency calling, Star Codes, voice mail integrated with e-mail, and Three-Way calling.

A Web-based Dashboard provides flexible, easy to use call management features such as: on-screen call alerts with call handling, call forwarding preferences, at-a-glance logs of incoming and outgoing calls, a frequently called number list, click-to-call features, and integration with address books.

TotalTalk makes it possible to manage and retrieve both voice and e-mail messages, anytime, anywhere from any touchtone phone or Web browser. Voicemail messages appear together with e-mail messages, and SMS alerts can be sent to mobile devices when a voicemail is left.

AOL also announced three new partnerships to bring industry-leading audio and video conferencing to its suite of voice and instant messaging products. AOL has worked closely with open source solution provider Pingtel, Global IP Sound, and On2 on custom-developed solutions designed to deliver state-of-the-art audio and video quality, simplify connections behind firewalls, and provide SIP compliance and more to TotalTalk and AIM.

Pricing is as follows:

- *Local Plan:* $18.99 per month (plus taxes and additional fees) for unlimited local calling, plus $.039/min for domestic long distance.
- *Unlimited Calling Plan:* $29.99 per month (plus taxes and additional fees) includes unlimited domestic long distance and Canada.
- *Global Calling Plan:* $34.99 per month (plus taxes and additional fees) includes unlimited domestic long distance plus low international rates.

**http://www.totaltalk.com**

## Aruba: The Largest WLAN Installation In The UK?

**By Johanne Torres**

Aruba Networks (news - alert) of Sunnyvale, CA claims to have created what is believed to be the United Kingdom's largest single wireless network.

The project was part of a major overhaul of IT infrastructure at the University College London Hospitals (UCLH) NHS Trust. The network covers 7,000 users in eight hospitals, including UCLH's new flagship hospital on Euston Road, London. It comprises three Aruba 5000 mobility controllers and more than 300 centrally managed access points (APs).

According to the company's news release, it was installed primarily to "accommodate UCLH's move to a completely paperless, Electronic Patient Record (EPR) system over the next few years, and will enable doctors and nurses to access patient information on the move anywhere within the hospitals, from bedside to operating theater."

"This is a great illustration of the vital role that wireless technology has to play in a modern hospital environment," said Kevin Jarrold, director of information management at UCLH. "Moving away from paper-based patient records to a wireless-accessible electronic system really frees up the way that medical staff work. Not only is information more accurate, but notes can be referred to and updated in real time from anywhere in the hospital. This is particularly useful when a patient is being moved and the doctor requires access to his or her record en route."

The network is also supposed to be able to protect patient info through a centralized multi-layer security protection system, which requires any user or device attempting to connect to a corporate system to obtain authentication before being allowed onto the UCLH network. Additionally, all encryption is implemented by a centralized hardware-accelerated encryption engine rather than in individual access points.

"UCLH is at the forefront of both technology innovation and clinical care," says Bob Vickers, head of sales UK, Aruba Networks. "Not only will the wireless network immediately increase the efficiency of information access, but it also represents the foundation of future converged communications advances. For instance, the network is fully ready for VoIP over wireless."

Aruba made news back in June, when it was picked by Microsoft to deploy next-gen networking equipment serving over 25,000 simultaneous users each day across more than 60 countries around the world.

The companies' ambitious project will be deployed in 277 buildings covering more than 17 million square feet. The deal called for Microsoft's new WLAN to be constructed using Aruba's mobility controllers, mobility software, and some 5,000 ultra-thin Aruba wireless access points.

**http://www.arubanetworks.com**

## QUALCOMM Bundles WLAN Into Mobile Headsets

**By Johanne Torres**

Wireless technology provider QUALCOMM Incorporated (quote - news - alert) will support Royal Philips Electronics' (news - alert) wireless local area network (WLAN) module, the companies announced. The integrated system will connect to WLAN networks as well as to existing cellular networks, featuring compatibility with 802.11b and 802.11g protocols on CDMA2000 and WCDMA (UMTS) networks.

"Our work with Philips helps us to address the demand for additional functionality in mobile devices," notes Mike Concannon, vice president of strategic products for QUALCOMM CDMA Technologies. "This strategic relationship delivers a number of new possibilities, which we look forward to making a reality."

The WLAN system will enable mobile devices to use high-speed wireless local area networks when available for connections of up to 54 Mbps. Initially supported on the MSM6550 chipset, the 802.11b/g-compatible broadband capabilities enabled by Philips' WLAN module are scheduled to be commercially available by the end of this year. WLAN technology supports VoIP apps, voice calls with simultaneous data transfer, and other data-intensive apps.

"We are pleased that QUALCOMM has chosen Philips' WLAN solutions to help meet the growing demand for greater connectivity in mobile devices," says Paul Marino, vice president and general manager, Connectivity, Philips Semiconductors. "Combining our highly-integrated, low-power WLAN solutions with the industry-leading QUALCOMM MSM chipsets brings a new level of mobile functionality to wireless users."

QUALCOMM made news back in May when it was hit by a lawsuit filed by Broadcom for alleged patent infringement. Broadcom claimed that QUALCOMM infringed the company's patents related to wired and wireless communications and multimedia processing technologies.

QUALCOMM then hit Broadcom back with a lawsuit in July for infringement of seven QUALCOMM patents. QUALCOMM noted in a news release that it was suing the company because "Broadcom is infringing six of the patents by the manufacture and sale of integrated circuits for use in GSM Standards handsets and is infringing the remaining patent by the manufacture and sale of semiconductors for WiFi devices."

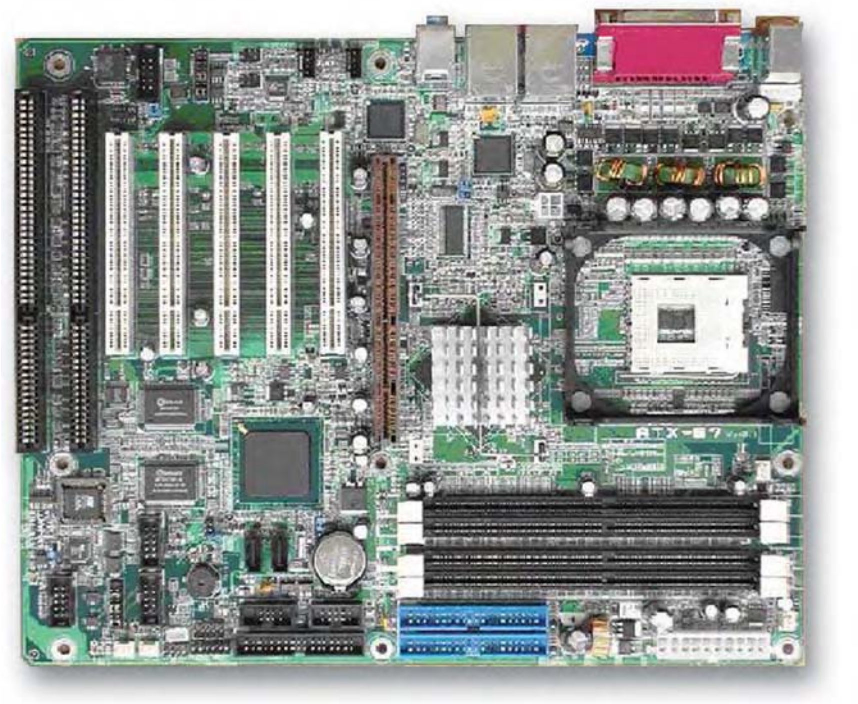
QUALCOMM is in the process of seeking an injunction against Broadcom's continued manufacture and sale of these products as well as monetary damages.

**http://www.qualcomm.com**
**http://www.semiconductors.philips.com**




Go To Table of Contents | Go To Ad Index

## Arista Unveils Industrial Motherboard



Arista Corporation (news - alert) recently introduced its new ATX-865G Industrial Motherboard. The motherboard is specifically designed for Industrial computing applications. The computer features a Pentium 4 Prescott (90nm) CPU with 400/533/800M FSB selectable speed and is Intel VRD 10.1 compliant to support future advanced processors. Ideal applications include computing functions for factory automation process, computer automation, medical devices, and custom integration solutions.

Arista's new ATX-807 CPU comprises an Intel Pentium 4 processor and four DDR DIMM Sockets supporting DDR 266/333/400 unregistered non-ECC Memory up to 4.0 GB. The ATX-865G has an integrated Intel Extreme Graphic Engine with 266MHz Core Frequency and a 100/10 Intel 82547GI and 82541GI Ethernet controller on-boards with two LEDs to display speed, link, and activity. The motherboard is also equipped with eight USB ports compliant with USB Specification Rev. 2.0 and support USB Hot-Plug function. The ATX-807 also provides two ISA slots for applications that need to use ISA legacy cards.

"The ATX-865G is a state-of-the-art motherboard with the latest computing technology and the ability to upgrade to future technologies," says Stewart Austin, director of sales for Arista Corporation. "Backed by Arista's two-year worry-free warranty, Pentium 4 products, and one of the industries highest MTBF, the high-quality ATX-865G is ideal for industrial, medical, and a variety of different computing applications."

Pricing for the ATX-865G starts at $365 for North American sales and is available for immediate delivery.

**http://www.aristaipc.com**

## Sangoma Seeks Patent For 'Echo Canceller Controller'

**By Ted Glanzer**

Telecommunications and routing equipment provider Sangoma Technologies Corp. (news - alert) announced that it filed a patent application for enhancing the efficiency of echo cancellation hardware and software.

Specifically, the application — titled "Echo Canceller Controller" — covers the techniques of measuring echo on incoming voice streams and controlling the echo canceller.

According to a press statement, Sangoma's Echo Detection and Control (EDAC) "is an algorithm that examines each call as it is connected and, within about one second, determines whether the call has echo or not. It then enables or disables the echo canceller as necessary."

The end result, according to the statement, is that echo cancellation loads on HMP systems can be reduced by 80 percent or more.

"EDAC will be implemented as a standard application in our on-board firmware at no cost," says David Mandelstam, Sangoma's CEO. "We expect EDAC to become a must-have feature in the delivery of practical HMP and low-cost hybrid HMP/hardware-based VoIP gateways, PBX and IVR systems, and other telephony systems."

**http://www.sangoma.com**

## Uniqall Offers New Gridborg HMP Release

**By David Sims**

Uniquall (news - alert) is announcing the fourth beta of its upcoming Gridborg HMP Server 1.1 software available for free evaluation and download. With T.38 FoIP (Fax over IP) working, the Gridborg Host Media Processing Server targets vendors of fax board-based applications who are looking for products advertised to both cut prices and enable their products for the boardless VoIP world.

According to a recent New York Times article cited in the Uniqall news advisory, the Consumer Electronics Association says there were 1.5 million stand-alone fax machines sold in the US alone in 2004.

The popularity of VoIP brings a problem to light, according to Uniquall, based in Zagreb, Croatia: "The need to attach those fax boards directly to analog or T1/E1 digital telephone lines made VoIP deployment within an enterprise more complicated and expensive than it should be."

Here's where their T.38 Fax over IP enabled HMP product, Gridborg HMP Server, comes in. By upgrading their products to support HMP in addition to fax boards that are already supported, Uniqall officials maintain, "fax server software vendors will ensure that their existing customers have a smooth and simple transition path once when they start their VoIP deployment.

"Once HMP technology is incorporated there is very little, if anything at all, that prevents vendors from releasing stripped-down, software-only, versions of their products. Such versions will allow them to conquer market segments that were previously completely out of reach due to the disparity between what potential customers were willing to spend and the total price of expensive fax board-based solutions."

Drazen Dimoti, lead developer on Uniqall's Fax over IP effort says, "it should also be noted that vendors of T.38 building blocks have to put in additional work to ensure standard compliance in order to avoid spoiling their own party."

Other new and upcoming features in the Gridborg HMP Server 1.1 include support for plugable codecs, Redirect Number, and dramatically improved conferencing capabilities. Gridborg HMP 1.1 works on x86 processors within both Windows and Linux environments. Support for additional operating systems and processor architectures will follow in subsequent releases.

**http://www.uniqall.com**

## GIPS Launches ATA Voice Quality Enhancer

**By Johanne Torres**

Audio processing vendor Global IP Sound (GIPS) (news - alert) introduced Voice Quality Enhancement (VQE) for ATA recently. The module cancels echo and noise by adjusting speech levels, thereby improving voice quality in low-complexity ATA box systems.

"The demand for VoIP equipment is poised to explode over the next several years as home users adopt the technology and chip makers and ATA manufacturers are anxious to bring their products to market quickly," says Jan Linden, vice president of engineering at Global IP Sound.

"In the same way that our earlier VQE modules empowered applications vendors, VQE for ATA utilizes GIPS unmatched, field-proven technology to put the highest-grade echo and noise cancellation technology available today into the hands of the ATA market," adds Linden.

VQE for ATA technologies include Network Echo Cancellation Integrated Access Device (NEC IAD), Noise Cancellation (NC), Noise Suppression (NS), Automatic Gain Control (AGC), Voice Activity Detection (VAD), and Comfort Noise Generation (CNG). The NEC-IAD in VQE also complies with the G.168 standard for echo cancellation.

Global IP Sound has also inked a deal with Skype recently whereby GIPS will provide Skype with components of its voice processing technology not only for use in the company's software for PCs but also for pre-installed versions of Skype on certain hardware devices.

The company was also tapped by America Online Inc. (AOL) to work "behind the scenes" with the dial-up ISP as it launched an enhanced version of its Internet Phone service.

GIPS had also announced earlier this year it teamed with Logitech to jointly develop and launch the Logitech ViewPort AV 100. GIPS integrated its Acoustic Echo Cancellation (AEC) technology to equip the device with audio and video, making it the foundation for a desktop video communications system.

**http://www.globalipsound.com**

## RADCOM Adds Software Development Kit To Its SIPsim

**By Johanne Torres**

RADCOM Ltd. (news - alert) just added a Software Development Kit (SDK) to the company's SIPSim, an SIP services simulation tool.

The SDK-enhanced system allows customers to define call flows, thereby simulating conditions of extreme stress. The system's Call Flow Library feature can be deployed as is or with customer-defined modifications to stress-test any SIP entity. Customers can also control call flow run-times and other relevant parameters via GUI commands.

"Because of its flexibility, developers and service providers use SIP to create a variety of devices, services, and applications. The diversity of these SIP-based products makes testing SIP architectures increasingly more complex. RADCOM's solution offers a variety of testing functionalities for these many and varied SIP implementations, from VoIP calls, to instant messaging to multimedia applications. This solution is in line with our strategy of providing customers with a high performance, carrier-grade solution that also offers maximum session-tuning flexibility," says Ofir Michael, director, Product Management, VoIP and IPTV solutions, RADCOM.

RADCOM's SIPSim helps vendors to develop and service providers to evaluate SIP-based network elements. "It enables them to conduct benchmark and performance testing in a real-world, high-volume call environment. By discovering the device's 'breaking point' and revealing what happens at that point, the solution reliably predicts the behavior of network elements under extreme stress conditions," notes the company's news release.

**http://www.radcom.com**

## AudioCodes Intros ATA VoIP Chip

**By Johanne Torres**

AudioCodes (news - alert) announced that it had teamed up with Jungo Ltd., (news - alert) a gateway software developer, and Legerity, (news - alert) in order to introduce the Tulip AC494 ATA, a VoIP chip with fully integrated software and hardware reference design.

The developer trio plans to market the product to original equipment manufacturers (OEMs), original design manufacturers (ODMs), and designers of analog telephony adapters (ATAs) and broadband access devices (xDSL, cable, wireless, FTTH) seeking to add VoIP capabilities to their products.

"With the rapid deployment of Voice over IP in the world, more and more companies are looking to add voice capabilities to their products, while maintaining and improving their data capabilities," says Shaul Weissman, vice president, VoIP processors and modules business line manager at AudioCodes. "Combining our VoIP chip and proven voice quality with Jungo's excellent data solutions and Legerity's high-performance VE880 VoicePort Series gives us a winning combination for the market."

The Tulip AC494 ATA integrates two 10/100 baseT Ethernet ports and one to four VoIP ports with two FXS analog interfaces. The product can support various configurations of FXS and FXO ports using pin and software-compatible devices from Legerity's VE880 VoicePort Series.

"Integrating Jungo's market-leading OpenRG software platform with the award-winning VoIP solutions from AudioCodes creates a compelling offering for OEMs," says Yaal Eshel, Jungo's director of sales and business development, Europe and the Middle East. "The highly integrated design will help OEMs bring to market high-quality ATAs and voice gateways in the shortest time possible."

"Legerity's VE880 VoicePort devices are revolutionizing the VoIP industry. They allow designers to create reference designs with groundbreaking performance at the industry's lowest total system BOMs," says Mike Stibila, vice president of marketing and applications at Legerity. "The Tulip AC494 ATA design maximizes the features of the VE880 Series and is propelling the VoIP market forward with the high performance/low cost design necessary for residential and SOHO VoIP market saturation."

The Tulip AC494 ATA is currently available for under $25.

**http://www.audiocodes.com**
**http://www.legerity.com**
**http://www.jungo.com**




Go To Table of Contents | Go To Ad Index

# BEA Sets SIP Routing Benchmark With Intel Hardware

By Robert Liu

Eight months after coming to market with a standards-based suite of carrier-grade infrastructure software for the telecommunications industry, BEA Systems (news - alert) recently touted its capabilities when applied to Intel's (quote - news - alert) Pentium architecture.

At its annual user's conference dubbed BEAWorld, the middleware vendor said its BEA WebLogic SIP (Session Initiation Protocol) Server using Pentium hardware was able to process and route SIP-based voice messages at breakneck speeds with an extremely small amount of network latency.

"As voice networks begin to evolve and assimilate Internet technologies, applications server performance and scalability become crucial factors in the design of network architectures," said Michael Stanford, director of VoIP strategy, Digital Enterprise Group at Intel.

BEA WebLogic SIP Server is a component of the BEA WebLogic Communications Platform, formerly code-named Project Da Vinci. The BEAWorld event and a batch of related announcements come at a time when middleware vendors are hoping to capitalize on the rejuvenated activity in the telecom sector. Due largely to the widespread proliferation of SIP as an industry standard, BEA is hoping the migration to packet-switched technologies from a circuit-switched world will lead to new avenues for applications and hence new markets for its applications server.

But on top of the likes of WebMethods or IBM, BEA also faces new competition in the form of Microsoft, which has also announced plans to explore interoperability with JBoss Enterprise Middleware System, an open source Java-based application server. Microsoft has its own Communications Sector practice targeting wireline and wireless telecommunications providers, hosting companies, cable operators, and media and entertainment companies.

In a press statement, BEA said the industry standard for voice calls is often measured in "Busy Hour Call Attempts" (BHCA), or the number of completed calls during a "Busy Hour." The BEA/Intel SIP application server solution clocked in at 10 million BHCA "with bandwidth to spare," exceeding industry expectations by 9 million calls. With regard to message latency, the combined BEA/Intel SIP application platform processed messages in less than 40 milliseconds (ms), equal to one thousandth of a second. The "performance breakthrough" (as the company describes) beats latency standards by 160 ms.

No competitive data were provided.

"The BEA WebLogic SIP Server and Intel benchmarks represent a new and exciting high-performance and scalable platform for next-generation VoIP and IP Multimedia Subsystem (IMS) communication and collaboration services," said Mike McHugh, vice president and general manager, BEA WebLogic Communications Platform, BEA Systems. "This standards-based, open platform — combining the rich programming environments of J2EE, SIP and IMS — addresses the key challenge operators face in moving to their next generation networks: dramatically lower costs, unprecedented performance, and a tremendously simplified services development and deployment model."

http://www.bea.com
http://www.intel.com


Getting rid of the old phone system was easy. Dead easy.
THE RINGING
SWITCHING TO COVAD VoIP will make your company more PRODUCTIVE, easier to MANAGE and save as MUCH AS 40%.
BUT what are the DARK consequences if your OLD SYSTEM doesn't want to be SWITCHED?
SEE IT AT THERINGINGMOVIE.COM
Cisco Powered
COVAD voip
The New Voice of Business
©2005 Covad Communications Group, Inc. All rights reserved. Covad, Covad Communications and the Covad logo are registered trademarks of Covad. Services not available in all areas. See covad.com/voip for service details. Up to 40% savings based on case studies by Covad comparing total cost of ownership of Covad's vPBX product vs. PBX premises-based solutions. Cisco, Cisco Systems, the Cisco Systems logo, and the Cisco Square Bridge logo are registered trademarks or trademarks of Cisco Systems, Inc. and/or its affiliates in the U.S. and certain other countries.

## Persona Software Introduces Solution For FMC

Persona Software, Inc., (news - alert) has announced a new version of its Persona OnePhone solution for Fixed Mobile Convergence (FMC). Persona OnePhone 2.0 provides support for an expanded range of dual- mode (WiFi/cellular) handsets and operating systems including Symbian OS. In addition, this latest version of Persona OnePhone adds several new end user features, increases security capabilities, and integrates seamlessly with IP Multimedia Subsystem (IMS), which is the basis for FMC and future converged services.

Persona OnePhone 2.0 further enhances Persona Software's leadership in the FMC market and gives carriers a robust FMC solution based on mature, proven SIP-based technology that they can deploy in their networks now.

"The FMC market is ready to explode. For service providers to attack the FMC opportunity now, they need a carrier-grade FMC solution that meets their requirements for reliability and scalability, offers differentiated mobility features; and is based on the open SIP standard that is the heart of next-generation IMS/3G networks," says Rob Fuggetta, Vice President of Marketing and Chief Marketing Officer for Persona Software. "Persona OnePhone 2.0 meets and exceeds these needs, giving service providers a unique and truly commercial-grade FMC solution they can go to market with now."

Delivered on Persona Software's SIP Application Server, Persona OnePhone enables 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 multimedia "Personal Mobility Applications" that provide users with unprecedented mobility choice, control, and savings.

**Expanded OS Support**

Persona OnePhone 2.0 adds support for three additional operating systems, including Symbian OS, Windows Mobile 5.0, and Micro Linux Mobile. Persona OnePhone will continue to provide support for Microsoft Windows 2003.

Other important new features:

- Adds new business, consumer, and regulatory features such as emergency service support, call transfer, ring-back when free, click-to-dial, and many other features that expand the mainstream market appeal of FMC services.
- Strengthens security capabilities, a key requirement for mobile enterprises, by expanding support for IP Sec over both WiFi and GPRS networks.
- Integrates seamlessly with the IP Multimedia System (IMS), enabling service providers to start delivering IMS/3G applications now while migrating to IMS at their own pace.
- Meets the key requirements stated recently in the product requirements document issued recently by the Fixed-Mobile Convergence Alliance (FMCA), a group of approximately 30 leading carriers worldwide that are promoting adoption of FMC. These carriers serve more than 440 million customers worldwide.

**http://www.personasoft.com**

## Nuera Intros Small SIP VoIP Media Gateway

Nuera Communications, Inc. (news - alert) introduced the SGX-100 SIP (Session Initiation Protocol) media gateway with an eye on helping small-scale service providers and large enterprises establish cost effective points of presence and connect legacy systems to the public telephone network. The SGX-100 is a carrier-ready small SIP media gateway that distinguishes itself with low price and redundancy. With list pricing starting at $5,000, the SGX-100 scales in capacity from one to four E1 or T1 ISDN circuits. The hot-swappable redundant power supplies of the SGX-100 are designed to provide high system availability typically found only in medium and large media gateways.

The SGX-100 is the first member of Nuera's SIP media gateway product line, which will also include gateways with higher capacities.

"Nuera has a long history of helping service providers and large enterprises take advantage of advancements in VoIP technologies" says Bill Ingram, president and chief executive officer of Nuera Communications. "The success of SIP in enabling both advanced subscriber features and cost-effective PSTN connectivity marks an exciting stage in our industry. We expect the SGX-100 and our upcoming SIP gateways to help build the excitement and grow the industry."

The SGX-100 is available as of November 1, 2005.

**http://www.nuera.com**

# TARGET DISTRIBUTING

YOUR 1-STOP TELECOM, VoIP AND SECURITY SUPPLY HOUSE

# YOUR VOICE AND VIDEO IP SOURCE!

POLYCOM®

Certified Dealers Only!

**SoundPoint® IP300**
**SIP Protocol 2-Line IP Phone**
**2200-11300-001**
- High quality entry level IP phone
- Up to 2-line capability
- Dual 10/100 Mbit/s switched Ethernet ports
- Handset/headset mode
- 4 line x 20 character wide LCD
- 802.3AF POE Compatible

**CALL FOR PRICING**

Certified Dealers Only!

**SoundPoint® IP500**
**SIP Protocol 3-Line IP Phone**
**2200-11530-001**
- Full duplex speakerphone
- Large LCD display
- Auto IP Address Configuration
- Headset capable
- 12 configurable feature keys
- Predefined hold key
- Stutter tone MWI MORE!
- 802.3AF POE Compatible

**CALL FOR PRICING**

Certified Dealers Only!

**SoundPoint® IP600**
**SIP Protocol 6-Line IP Phone**
**2200-11630-001**
- Acoustic Clarity Technology
- 320 x 160 pixel graphical LCD
- Two-port 10/100 Ethernet switch
- Flexible powering options
- Ideal for hosted IP, e.g. BroadSoft
- 802.3AF POE Compatible

**CALL FOR PRICING**

**2200-11611-001**
- Wall Mount Kit

Certified Dealers Only!

**SoundStation® IP4000**
**SIP Protocol**
**IP Conference Phone**
**2200-06640-001**
- Industry leading speakerphone technology
- Fully duplex w/echo cancellation
- Automatically adjusts to room
- Software is field upgradable
- 3 context-sensitive soft keys
- Handles 2 calls simultaneously
- Includes AC power supply

**CALL FOR PRICING**

POLYCOM®

**PVX™**
**Video Conferencing Software**
**5151-22019-001**

- Supports Polycom Siren™14 kHz audio
- Codes and transmits 14kHz - Requires 14kHz capable headset
- Supports G.722.1 and other industry audio standards
- Advanced, full-screen, full-motion video up to 30 fps
- Additional Annex support for better H.263+ video
- Show your desktop as high-resolution, XGA graphics to the far side while having high quality video simultaneously
- Receive high-resolution content from remote participants without losing their video
- Full collaboration with other T.120 endpoints
- Supports all Microsoft® NetMeeting® functionalities – Chat, File Transfer, Whiteboard, and Application Sharing

**CALL FOR PRICING**

POLYCOM®

**VSX™7000**
**Video Conferencer**
**2200-10800-001**

- H.264
- Delivers up to 30 fps video at 512Kbps
- Full-duplex digital audio with noise suppression and echo cancellation
- Integrated speaker with subwoofer for enhanced audio (80Hz to 22 kHz)
- Treble and bass controls
- Mic and VCR input audio mixing
- Remote control
- 360º voice pickup with 3 cardioid elements per microphone array

**CALL FOR PRICING**

**VSX™7000 Quad BRI**
**2215-20523-001**
- Quad BRI module for the VXS 7000 **CALL FOR PRICING**

BICSI® CORPORATE MEMBER

# 1-800-873-5528

www.targetd.com • TGSales@targetdist.com
Monday thru Friday 8:00 AM – 5:30 PM EST

CEDIA CUSTOM ELECTRONIC DESIGN & INSTALLATION ASSOCIATION Member

**ORDERS RECEIVED BY 4:30 PM EST SHIP SAME DAY! WEST COAST SHIPPING IN ONLY 3 DAYS AT GROUND RATES!**

## NetSuite Adds CRM+ To Portfolio

**By Tracey Schelmetic**

NetSuite (news - alert) states that the new CRM+ forecasting capabilities are not possible with many other SFA applications since competitors' solutions are not built to handle sales transaction-related processes such as booked orders or incentive compensation. CRM+ is built on the foundation of NetSuite's Order Management & Commissions platforms, and the new capabilities allow companies to measure sales teams using more than just gross metrics of total sales or total bookings. NetSuite provides examples of how this functionality is useful: many companies desire to have one quota for product sales and another for services sales. Companies that offer an ongoing subscription service may wish to incent their sales force on the revenue recognized from a single year of a contract, as well as the total value of the booked order.

"This announcement marks the death of the Excel forecast. Simple, single metric quotas and forecasts are rare in any size business, but the SFA tools offered — both on premise and hosted — can't handle this basic requirement. As a result, most forecasting and commissions happen in Excel, not the SFA tool," says Zach Nelson, CEO of NetSuite. "Because NetSuite CRM+ captures the actual sales transaction in the form of an order, that order can now be sliced and applied against multiple quotas, forecasts and commissions schedules automatically. No more Excel."

**http://www.netsuite.com**

## Microsoft Absorbs Unveil Speech Solution

**By Tracey E. Schelmetic**

Speech marches on at Microsoft. (quote - news - alert)

The Redmond, Washington-based behemoth today announced it has acquired "certain intellectual property assets" from Unveil Technologies, Inc. Unveil (news - alert) was, until recently, a builder of a product called Unveil Conversation Suite, a speech application environment manager. Conversation Suite allows the 99.9999 percent of business people who are not speech software engineers to build and tune speech applications like pros.

Using the Unveil product, call centers can quickly build customized "virtual agents" to help offload common transactions from real agents.

With the acquisition, Microsoft plans to integrate the Adaptive Learning and Conversation Assist components of the Unveil product into future versions of Microsoft Speech Server. According to Microsoft, this is a key step in its path to delivering "the most flexible and integrated speech platform at the lowest total cost of ownership."

Microsoft believes that, when integrated into the Speech Server platform, Unveil's innovative learn-by-example and agent assist technologies will make it easier, faster, and less expensive for customers to develop and optimize speech applications.

"Unveil's technology is a natural fit with Speech Server because it is in line with our vision of making speech applications affordable and easy to develop and maintain," said Rich Bray, general manager of the Speech Server group at Microsoft. "By supplementing our toolset with Conversation Suite technologies, even a non-technical call center manager can prototype and develop robust speech applications."

As part of the acquisition, Microsoft will retain some Unveil development employees. Unveil's existing customers will have the opportunity to participate in future programs that include the acquired technology.

**http://www.microsoft.com**
**http://www.unveil.com**





Go To Table of Contents | Go To Ad Index

### Telephony@Work Inks Deal With Marconi

Telephony@Work (news - alert) announced a partnership with Marconi Corporation plc., (news - alert) whereby the latter will deploy the former's hosted contact center application, CallCenterAnywhere, in conjunction with Marconi's Impact SoftSwitch, to deliver Impact UltraContact, which will enable service providers to generate new revenue streams by offering a hosted contact center solution to small to medium enterprises.

Marconi Impact UltraContact is a hosted software application that handles all interaction between customers and a business, including telephone, Web chat, and e-mail. It can deliver all the features of a contact center, including automatic call distribution, computer telephony interaction, and interactive voice response. As a hosted solution, it does not require SMEs to make the investment in time, money, and expertise associated with rolling out a premises-based contact center.

Impact UltraContact is hosted by the service provider and is essentially rented to SMEs, which are charged a usage-based fee or a monthly rental amount, ensuring that the service provider has an ongoing source of revenue.

"The partnership with Telephony@Work forms part of Marconi's ongoing strategy to partner with best-in-class companies to deliver applications that will help service providers better serve their customers," says Kevin Hourigan, vice president of Marconi's SoftSwitch product group. "Impact UltraContact will help service providers fulfill the demands from SMEs to provide hosted voice applications, which have the same functionality as the most comprehensive contact center system, but do not require a costly outlay, ongoing maintenance, or upgrades."

Telephony@Work, Inc. is a provider of multi-tenant and adaptive IP contact center technology for enterprises and service providers.

"The partnership with Marconi will enable Telephony@Work to expand its market reach to new service providers, who are becoming increasingly important as both SMEs and larger organizations look to their service providers to supply value-added voice services," says Eli Borodow, CEO of Telephony@Work. "Marconi has expertise and relationships in this market and their customers are looking for exciting new applications that can provide customer value and generate additional income in managed and hosted services."

Marconi's SoftSwitch offers carrier-class reliability in an open-interfaced IP platform that meets the increasing needs of network operators and large enterprises.

**http://www.marconi.com**
**http://www.telephonyatwork.com**

## Pandora Networks Deploys IP Communications Service Globally

Pandora Networks (news - alert) announced that VCOMM UK Ltd., a distributor of IP Telephony solutions to the UK and Irish channel, has partnered with Pandora to distribute Worksmart services, an integrated VoIP, video, messaging, and collaboration service for small and medium size businesses ("SMB").

Under the terms of the agreement, VCOMM UK will act as the Master distributor to resellers and service providers throughout the United Kingdom and Ireland. Worksmart is a complete IP communications solution designed to enable SMB IT managers to immediately self-provision and manage a wide variety of integrated communications and collaboration services including a VoIP-based telephone network and hosted PBX service.

"We recognize that the UK market is an important step in our objective towards making Worksmart a global business communications service. VCOMM was chosen to distribute our service in the UK market because they have the technical and marketing expertise that we require when expanding into new markets," comments Walter Snell, chief executive officer of Pandora Networks. "Its customer base, coupled with their extensive VoIP knowledge and partnerships with carriers, resellers, and ISP network partners, make VCOMM an ideal European partner for Pandora.

"Worksmart is not just another IP Centrex platform," claims Scott Dobson, managing director of VCOMM UK Ltd. "It incorporates innovative value added hosted services such as instant messaging, presence, desktop share, and Web contact center capabilities in concert with all of the standard IP PBX services such as IVR, ACD, and Follow Me call routing." Dobson adds, "VoIP service alone is not what small and medium sized business customers in the UK are looking for. They want a supplier to solve all of their communications needs and Worksmart uniquely delivers that in one service without a capital equipment expense."

**http://www.pandoranetworks.com**

## VoIP Inc. Intros VoiceOne Carrier Direct Program, Expands Net

**By Johanne Torres**

VoIP Inc. (news - alert) of Fort Lauderdale, Florida is hailing the launch of its VoiceOne Carrier Direct Program. The new program is designed to enable carriers to provide IP-based services with no CAPEX or other requirements.

"Through our new Carrier Direct Program, VoiceOne provides carriers with media gateways at no charge, and installs and connects them to the VoiceOne network. This enables carriers to immediately begin selling IP-based services, such as hosted IP centrex, broadband voice, IP origination/termination, 800 service, CNAM, 911 services, etc., over the VoiceOne network," explains VoIP Inc.'s CTO Shawn Lewis.

The VoiceOne is a protocol agnostic network that supports SS7, PRI, H.323, SIP, and MGCP protocols utilizing both circuit and packet switched technology.

"Our Carrier Direct Program enables carriers to increase their level of service offerings and recognize an immediate ROI, utilizing our technical expertise to support them as they quickly gain entry into this fast-paced market," said VoIP Inc.'s CEO Steven Ivester. "The best part of this service for carriers is that it is integrated back into their existing TDM infrastructure with no changes and no CAPEX requirements," Ivester added.

The company also announced the expansion of its VoiceOne direct network to an additional 48 CLEC customers and cities. The company's network, which has been running for more than two-and-a-half years, currently interoperates with Netrake, Sonus, Cisco, Asterisk, Quintum Technologies, and others.

"Through this expansion, we are increasing our direct network footprint providing our customers with the necessary tools and network services to begin servicing the exploding IP services marketplace. Not only does the expansion benefit new and existing customers of VoiceOne, but it further builds upon and strengthens the quality of service aspects of IP communications in general," says Lewis.

**http://www.voipinc.com**





Go To Table of Contents | Go To Ad Index

## AudioCodes Launches Partner Portal

**By Johanne Torres**

Voice over packet equipment maker AudioCodes (news - alert) has launched a new channel partner Web-based portal. The new portal is an extension to the company's current Marcom portal, and includes sales and marketing tools for AudioCodes' channel and application partners worldwide.

Channel partners will now be able to use the new portal to locate info required in order to sell and support AudioCodes technology and networking products. Information available through the portal includes product and solution presentations, data sheets, white papers, case studies, and many other sales and marketing tools for the channel.

"AudioCodes' special position in the market, providing both technology products to OEMs and ISVs and networking products to systems integrators, enterprises, and service providers, allows us to play a key role in the new VoIP ecosystem," says Haim Melamed, AudioCodes' director of Channel Marketing. "Using our key position in the market, we can act as the point of contact between an expanding network of channel partners and a large number of application partners."

**http://www.audiocodes.com**

## IPN Resells ShoreTel's IP PBX In The UK

**By Johanne Torres**

IPN Consultants, (news - alert) a vendor-independent IP Centric telecom company, announced that has it become an authorized ShoreTel (news - alert) Channel Partner in the UK.

The new partnership will enable IPN to now offer a combination of products and services including ShoreTel's IP PBX Telephony services, WAN infrastructure, an OEM VoIP offering, and IP telephony.

"Our focus is to deliver IP centric global networking solutions that add real value to the corporate marketplace," says Stuart Smitherman CEO of IPN Consultants. "We believe IP-based converged solutions should not only offer a definable cost benefit but one that makes life easier for the organization. Better controls, easier management, improved service, and the like. ShoreTel's IP PBX fits into this category and greatly supports and enhances our unique WAN and VoIP offerings in this respect."

ShoreTel made news recently when it announced it had unveiled ShoreTel 6, the sixth generation of its distributed IP PBX voice system. The company introduced a new Office Anywhere feature, which supports mobile users regardless of location or device used. The company also delivered two new telephone devices, a low-end IP phone and a 24-button programmable button box for operators and assistants.

ShoreTel 6 is designed to enable users, system administrators, and managers to have a work identity and profile. It can be put on any device, including cell phones and PDAs, which eliminates the need for multiple phone numbers and voicemail accounts. The ShoreTel 6 has a native SIP interface that can support a variety of third-party applications.

**http://www.ipnconsultants.co.uk**
**http://www.shoretel.com**

# Black Box To Sell Siemens VoIP Products

By Johanne Torres

A new multi-year distribution agreement will enable Black Box Network Services Corp. (news - alert) to sell, deploy, and support the Siemens Communications, Inc., (news - alert) portfolio of HiPath Real-Time VoIP systems, devices, and applications, the companies recently jointly announced.

The new partnership will make Black Box a Siemens Authorized Reseller and, therefore, one of Siemens' largest distributors. Black Box completed the acquisition of Norstan, Inc., a 2005 Siemens' channel reseller.

"Siemens is an integral part of our voice services business," says Fred Young, Black Box's CEO. "With our new agreement, Black Box will continue to expand its installed Siemens customer base throughout the United States and Canada."

The agreement will allow Black Box to continue selling and supporting Siemens' complete communications systems including the HiPath Real-Time IP systems, which provide centralized control of features, access, configuration, and call-processing functions. The Siemens portfolio also includes HiPath Wireless, a secure and mobile management system for voice over wireless LAN networks. The HiPath portfolio also includes the DirX family of access and data management systems, the optiPoint family of IP and wireless phones, the ProCenter portfolio of contact center systems, the Xpressions unified messaging family, and the HiPath OpenScape, a presence-enabled and permission-based unified communication portal.

"Black Box, with its acquisition of Norstan, continues to expand its operational footprint and is a leader in providing enterprises with the next generation of converged voice, data and multimedia solutions," says Rick Fitzgerald, vice president and general manager of the Siemens Channel Solutions Group at Siemens. "Black Box is a strong technical services partner. Their sales and operational teams, coupled with a strong commitment to customer service, also make them a valuable enterprise partner."

This channel news follows Siemens' recently inked deal with Genesys Telecommunications Laboratories Inc. The companies joined forces in order bring a single, centralized platform of rich contact center and communication systems to large enterprises. Siemens and Genesys will do this by integrating the SIP-based Siemens HiPath 8000 Real-Time IP System and the SIP-based Genesys 7 portfolio of contact center management applications.

"The joint solution consolidates contact center applications on a SIP infrastructure and can help deliver key customer benefits such as centralized management operations and reduced total cost of ownership," says Nicolas De Kouchkovsky, senior vice president of marketing and business development for Genesys." This combination of SIP-based solutions means that all the resources of an enterprise, regardless of location, can be enabled with applications that produce high-quality contact center service and customer care."

**http://www.blackbox.com**
**http://usa.siemens.com/communications**

By Marc Robins

# Welcome To The Golden Age Of Web Telephony

It seems lately, at least when you read about VoIP (define - news - alert) in the mainstream press, it's all Skype, Skype, and more Skype. And no wonder: The outsize value of the eBay/Skype deal is certainly breathtaking, and it's exciting to see some froth in the market again (although with all the hullabaloo, I've seen scant discussion about how the two companies will go about integrating the Skype service with the eBay user community).

Of course, peer-to-peer (PC-to-PC) VoIP communications is hardly a new phenomenon. After all, the first iteration of VoIP in the marketplace 10+ years ago was PC-to-PC software applications like Internet Phone. Yes, problems abounded — which seriously hampered widespread adoption — and included pokey dial-up connections, clunky software, and underpowered PCs, and a multitude of Internet QoS impairments that resulted in very poor call and voice quality. And voice chat has been an option inside most instant-messenger applications for some time, but again, poor voice quality made it difficult to use.

The early technology, however, was truly pioneering and very much set the stage for where we are today. With today's widespread broadband connections; improved standards and protocols; robust softswitches and Web infrastructures; rapid growth of e-commerce and Web-based user communities; and PC hardware, operating systems and software light-years ahead of their predecessors, all the ingredients are in place, I believe, to bring on a golden age of Web telephony.

With the eBay/Skype combination, it's a natch that all eBay buyers and sellers will be provided with a free Skype (news - alert) account, tied-in with their existing user names. The eBay community will be able to conduct voice communications with each other at no charge, and eBay (quote - news - alert) will have the means to offer (and sell) more value-added communications services to its members, such as the Skype-Out offering. eBay will also bring PayPal into the mix, and allow Skype users to use their PayPal wallet to pay for call services. In fact, when eBay talks about the purchase, they talk about the "Power of Three" and the inherent synergies between eBay, Skype, and PayPal. It's a powerful and quite compelling rationale.

According to eBay, the acquisition will "expand eBay's share of e-commerce, accelerate commerce on eBay, open up new lines of business, new monetization models and new geographies, and serve as a great standalone communications business." The central notion here is that communication is essential for e-commerce to work, and that the online transaction process between buyers and sellers requires various communications points along the way, including Q&A, and transaction and order-related communications. Apparently, more than five million e-mail messages are initiated each day between buyers and sellers, and with Skype, eBay hopes to remove a key source of "friction" between buyers and sellers, creating an environment even more conducive to e-commerce and higher-value, more complex types of transactions.

So what lies beyond the obvious? Word has it that eBay/Skype will soon release an Internet Explorer toolbar that will let you effortlessly dial numbers that appear on search results or Web pages, and provide integration with Microsoft Outlook that will allow you to initiate Web calls to your Outlook contacts. Other applications apparently in development include "Web presence" integration, which creates click-to-call "Skype-Me" links in member profile areas of social networking Web sites, and 'Voice Marketplace" applications that power innovative voice service such as language learning tools, homework tutoring, and interactive voice response services such as traffic reports and the like.

Another deal that got widely picked up, but then got lost quickly in the shuffle, was the acquisition of Teleo by Microsoft this past August. Teleo has many of the peer-to-peer features of Skype, but is more focused on allowing users to use their PC to make VoIP calls to cell phones and regular phones. It is also ahead of Skype in coming up with a very slick integration with Microsoft Outlook and Internet Explorer that enables "click to call" dialing of any telephone number that appears on screen — such as within a Web site, search results page or e-mail message. The acquisition also underlines the growing importance of voice services to the instant-message platforms in use, such as MSN Messenger, as it will allow Microsoft (quote - news - alert) to build click-to-call links into the MSN service so that local search results — for a pizza parlor or flower shop, for example — can include a link to let a Web surfer's PC call directly to the business

Of the two deals, it seems to me, the Microsoft /Teleo one is actually far more interesting in terms of the implications for the marketplace for Web telephony. Microsoft has indicated that the "the Teleo technology would be used primarily within MSN applications, rather than being integrated into the Windows operating system." This remains to be seen. But just imagine the implications for the market if Microsoft decides to bundle Teleo's itty-bitty click-to-call app into Internet Explorer. IT

*Marc Robins is Chief Evangelism Officer of Robins Consulting Group, which offers an array of services to the IP telephony industry. He has been involved in the telecommunications industry as a reporter and analyst, trade show producer and publisher, and marketing executive and consultant for more than 25 years. For more information, call RCG at 718-548-7245 or e-mail robinsconsult@optonline.net.*


Your Network is Complicated Enough
Upgrading to Power over Ethernet
Shouldn't Make you Lose Sleep

By Tony Rybczynski

# Security Checks On Users

According to the FBI, the majority of security breaches are inside jobs. So how does the firewall you deployed at the edge of the Internet help you do this? It doesn't! What you need is end point security. End point security, whether local or remote and whether wired or wireless, ensures that only authenticated users and compliant devices can connect to the network and that these are authorized to access certain application and networking resources (all based on the enterprise security policy). This will better control who has access to which applications, protect people's productivity against the "worm du jour" and ensure security audits associated with regulatory requirements can be met.

## End Point Security And Layered Defense

Security needs to be implemented in a similar way to which we have built highly reliable networks, that is by removing single points of failure. This philosophy leads to a layered defense approach to security that places different forms of security at different layers or places in the network. Overall security is increased because threats that may pass one layer can be caught by another layer. This layered defense approach to security provides four basic functions: end point security, perimeter security, communications security, and core network security, incorporating many different approaches to enforcement ranging from access lists to behavior anomaly detection.

The first step in providing end point security is authenticating the user. While device-based authentication may be adequate in certain environments, user authentication enables role-based policy management that restricts user access to applications and network resources, and creates an environment in which users and devices are managed separately, an important factor in virtualization for increased mobility. User authentication consists of secure exchange of one, two or three identifiers (who you are, what you know and what you have) using for example hardware and software tokens, smartcards, and/or biometrics. User authentication can be done in a number of ways. Some options include: port-based authentication controls based on IEEE802.1x and the Extended Authentication Protocol (EAP); IPSec VPN (define - news - alert) authentication; submission of username and credentials via a SSL VPN. In all cases, inline gateways are required for the authentication and authorization session.

**The majority of security breaches are inside jobs.**

After user authentication, you need proactive checks to allow network access only by compliant devices, and reactive checks to detect and isolate non-compliant devices. End point security ensures that individual end points, whether wired or wireless, and at the desktop or mobile, are secured at the operating system, network, Web browser, and application levels.

## End Point Security Under The Covers

End point security verifies that current security software (e.g., antivirus and personal firewalls) is running and totally reflects current security policies. It also detects device configuration errors that may compromise security, missing operating system patches and expired intrusion detection and prevention signature files that may make security mechanisms ineffective. Operating system security settings can also be checked via scans launched from the server or portal at the time the endpoint device comes online. Custom checks, which allow for monitoring of registry keys, files and processes, can also be defined.

Once users are authenticated and the devices they are using checked against the security policy, centralized access controls kick in. In this way, only authenticated users connect to the network and when connected only have access to authorized applications. Management can issue, revoke, and change user access privileges. A number of remediation or enforcement policies can be instigated based on status such as authenticated user, unauthenticated user, vulnerabilities in scan results and failed compliance checks. If a user is not authenticated to the network, this can result in limited access to specific areas of the network, while authenticated users can undergo more strenuous checks and be granted wider access to network resources. If a problem is identified, the out-of-compliance device can be sent an installation file, receive an alert message or be sent to a URL.

## Client Versus Clientless Approaches

Both clientless and client-based end point security approaches check five parameters for assigning network context, and granting role-based access: Who the user is? Where the user is? What is the time of day? What is the level of compliance? Where can the user go?



Go To Table of Contents | Go To Ad Index

Client-based approaches have been available for some time for remote access policy enforcement integrated into IPsec VPN clients. The evolution of SSL VPNs and the recognition that end point security is equally important on wired and wireless LANs has resulted in both client-based and clientless end point security solutions being made available to enterprises.

A client-based approach requires client-side code that monitors the user device for malicious activity under control of an end point security server; in contrast, a clientless approach relies on the device being able to support common browser functionality with all monitoring being performed by an end point security portal. The big disadvantage of client-based approaches is that software has to be available for every wired and wireless, fixed or mobile device in your network, including a growing list of networked devices such as smart phones, PDAs and security cameras. Client software has to be downloaded to each device and upgraded periodically. All this translates into higher life cycle costs compared to clientless approaches, and either holes in the end point security device coverage or restrictions on device connectivity pending availability of client software for a particular type of device.

In contrast, a clientless end point security framework avoids these operational requirements and costs, while securing the network from endpoint vulnerability in the most effective manner. A security portal is central to such an approach and needs to be highly reliable, scalable, and have the flexibility to work into a broad variety of back-end authentication and security policy management systems in determining policy compliance, policy based routing and policy definition for network optimization. It also interacts with the network to control application and network accessibility. Clientless endpoint security provides a simple end point security solution for local LAN/desktop users, for mobile campus workers as well as remote users and teleworkers (whether using IPSec or SSL-based VPNs), using fixed and wireless options.

Going forward, a combination of client-based and clientless approaches may be used to address the variety of devices that need to be supported. However, approaches that require that all packets be inspected by an in-line gateway are clearly less scalable than those in which the gateway makes the admission decision and then is out of the data path until re-authentication is required. **IT**

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

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

By John Cimko

# S. 1504 — Senator Ensign's Big Gamble

S. 1504, introduced in July by Sen. John Ensign (R-Nev.), aims to overhaul the Telecommunications Act of 1996 to restore America's leadership status in global telecommunications technology and to ensure that federal law and regulations account for sweeping developments in technology.

Other proposals also are beginning to surface. The House Energy and Commerce Committee released a draft rewrite bill in September, proposing the same regulatory treatment for all broadband providers. Although action on telecommunications legislation before the end of the current Congress appears unlikely, some of the key players are starting to frame the issues.

S. 1504 takes a big gamble. The proposed legislation rolls the dice on the proposition that deregulating essentially the entire telecommunications industry (other than the provision of basic service) will enhance consumer choice, bring down prices, and serve President Bush's goal of universal broadband by 2007. As one economist (John Rutledge) claimed in a recent press report, S. 1504 is "the answer to outsourcing of U.S. jobs abroad. [The bill] will trigger massive . . . capital spending on fiber optics and high-speed networks . . . . In addition to clearing away the regulatory underbrush, this legislation could add as many as 200,000 jobs to the American workforce and generate upwards of $600 billion of GDP, resulting in higher productivity and lower inflation and interest rates over the course of the next five years."

These predictions about the Ensign bill assume, of course, that deregulation is the best way to "incentivize" telecommunications service providers to invest and compete. But two issues may raise questions about whether Senator Ensign's gamble will work.

First, the bill assumes there is enough competition right now in the telecommunications marketplace to warrant the wholesale removal of regulatory oversight. The argument goes that, in the case of broadband (which the bill defines, rather broadly, as any service at 64 kbps or above), cable companies and the RBOCs are vigorously competing today, and maintaining a level playing field for these competitors (without requirements for non-discriminatory access to their broadband facilities) will strengthen competition, spur investment, and benefit consumers. The soundness of the bill's assumptions hinges on whether what amounts to a duopoly in the telecommunications marketplace will actually produce the desired results. Will there be enough competitive pressure to drive RBOC investment in innovative broadband technologies that will boost speed and capacity? Will the ongoing march of mergers and acquisitions in the telecommunications sector undermine the assumptions about competition made by Senator Ensign's legislation?

A second issue involves universal broadband. The arguments of the bill's supporters don't address the issue of whether unregulated RBOCs and cable companies will have sufficient incentives to bring broadband services to rural and low-income areas if there are no regulatory requirements or incentives in place to ensure "universal" offerings of broadband services. In fact, when we look at the scorecard of "good news" and "bad news" in S. 1504, we see that the bill might actually deepen the digital divide.

This is because the bill would hamstring efforts by municipal governments to provide broadband infrastructure. Specifically, any state or local government planning to provide broadband services would be required to allow private companies to bid on the project and to get the same perceived advantages (such as free or below cost rights-of-way or any preferential tax treatment) available to the government. If a private company makes a bid that is "identical" to the government's plan, preference would go to the private bidder. If there are no private bids, or if the government wins the bidding process, and the government broadband facilities are constructed, then S. 1504 would permit private companies to use the facilities subject to the same conditions as the government.

This is bad news for local governments. The National League of Cities is concerned that the Ensign proposal would block efforts "to ensure that communications services are available to anyone, not just the chosen few." A growing number of local governments has reached the conclusion that one way to help President Bush meet his universal broadband goal is for local governments to bring broadband benefits directly to their citizens. The proposals in S. 1504, in erecting roadblocks against this municipal involvement, run counter to the President's broadband goals and seem indifferent to the digital divide between poor and rural areas and the rest of the country.

On the other side of the ledger, there is also some good news in S. 1504. For example, the bill prohibits network providers from blocking their subscribers' access to competing VoIP services, and also promotes "network neutrality" by barring network providers from blocking their subscribers from accessing any lawful Web content or from using any lawful Web application. The bill potentially undercuts this latter protection, however, by providing that a subscriber's access to Web content and applications is not guaranteed if this access would be inconsistent with the network provider's service plan.

S. 1504 is an important step in framing the debate for rewriting the country's telecommunications laws. As the debate proceeds, lawmakers and affected parties should focus on the gambles and trade-offs that have shaped Senator Ensign's proposal. **IT**

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

By Hunter Newby

# There's No Turning Back Now

GoogleTalk is all the talk. Hats off to them too. It's wonderful to know that the entire international calling world can be forever changed by ad revenue subsidies. In a strange sense, revenue from static text words will replace revenue from full duplex, real-time words. The truth is, the minute is dead and distance is gone. It's all about access. Vonage (news - alert), Skype (news - alert), and now GoogleTalk (news - alert) have taught the world a harsh lesson about radical change in a short period of time. VoIP (define - news - alert) as a technology within a standalone business model has been introduced to the mainstream, matured, and burned out in three years! VoBB, Voice on the Internet (that's free and for a fee) and now an open source, global free service. Where's the value in voice when the money from minutes is gone? For the end user it's become a give-away service, like e-mail. It isn't going to all happen overnight, but all of the pieces are in place and big names are starting to show their cards.

Microsoft also recently announced their entry into the VoIP business with the acquisition of Teleo. This marks a significant change in the industry and a point from which we will never go back. The VoIP peering significance of these public Internet VoIP application services being launched is that some of the operators (Google, Microsoft and potentially Yahoo!) may in fact peer with each other via Layer 2 Ethernet. In that case the public Internet portion of the call would only be on the access tails.

The relationship between these events and VoIP peering is an interesting one. These VoIP announcements are coming from providers of Internet-based VoIP services whereas what the enterprise and carrier voice network operators are typically building for their own internal purposes are Layer 2 networks to carry their VoIP traffic. The Fortune 500 will most likely never rely on the public Internet for their own internal voice network and will also most likely require their carrier service providers to do the same (that is if they know enough to ask). There will be some overlap in the SOHO business segment as they don't typically have the size, or savvy to build their own VoIP network, but the distinction between the two groups (end users versus corporate and carrier) and their preferred media (public Internet VoIP versus private Internet VoIP) will become much clearer.

This is an important part of the evolution of VoIP as it will help push the issue with all people, businesses, and carriers and create another information and education renaissance similar to when the e-mail phenomenon first occurred. Many people use VoIP today and don't even know it, but that is due to the carriers changing their own internal network operations and not making a big announcement about it. The renaissance occurs when people realize that if they're just a little bit smarter about how they do what they're already doing, they can do it better and at a lower cost. The most successful process for this realization is viral dissemination. Most people found out about e-mail from a friend. Most MIS and IT directors will probably find out about enterprise VoIP peering from their MIS and IT peers at other companies.

E-mail as an application over the Internet was a service evolution from the telegram as an application over telegraph networks. The time lapse from when the masses stopped sending telegrams to when e-mail was commercially available was too long for anyone to realize the relationship between the two, or to understand what improvements e-mail made to the telegram. First, telegrams were billed to the sender by the word. E-mail is free no matter how many words it is. E-mail is a service that for many is bundled in to the price of access. Second, to send, or receive a telegram you had to go to a telegram office, or station. E-mail, when it became available, was brought in to your home. It makes a big difference on rainy days and saves lots of time. Today with your PDA, e-mail is wherever you are. Since voice calling has been with us every day since it was introduced to the masses all the way from fixed-line to mobility we won't see VoIP as a "new" service like e-mail, but rather an improvement (technically, economically, or otherwise) to what we already have. Just like e-mail, voice calls, as VoIP, are quickly becoming a service that is bundled in to the price of access. Video will be next.

The correlation here is that all of our original "networks" — telegraph, telephone, and television — all have the same root: 'tele.' Tele is Latin for distance, or far-off. This is the single, fundamental Achilles heel to all of these networks as their business models were based on the concept. In fact, it was not a concept at all at the time, but a physical limitation, a reality. Today that is no longer the case. Internet Protocol is one common language that encompasses telegrams, telephone, and television as e-mail, VoIP, and video over IP. IP enables peering and IP Peering (including VoIP Peering) is translated into English as "local-area." Well, not exactly, but the point is that peering brings the world to you rather than you having to pay to get to every part of it. The world has become a much smaller place as a result. Now if we could only put DNA over IP I could travel over a Google Earth-like application, get there in three seconds and not have to fly on a plane again. **IT**

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

# OFFER YOUR CUSTOM CONVERGED IP NETWORK.




Today, more of your customers want networks with managed IP services, while also demanding those networks be faster, more efficient, more secure and handle more IP applications. All this, while keeping their legacy infrastructure. It's a puzzle. But it's one that Global Crossing's single, global, converged IP network can put together. It offers simple, seamless interoperability between the legacy infrastructures your customers

By Max Schroeder

# Of Old Dogs And New Tricks

## FoIP + VoIP + Data = Converged Communications

Despite the enormous growth of the Internet and e-mail, fax remains a viable communications medium. Not bad for a technology that was invented in by Alexander Bain, a Scottish mechanic in 1843. According to industry estimates, there are over 100 million fax machines in use in the world today. Surprising numbers considering the following scenarios:

• Employees sending or receiving an e-mail would be forced to walk to the company "e-mail machine" (SMEM – Snail Mail E-Mail).

• The same procedure would apply for Web access and instant messaging.

If the above conditions were implemented in your office, messaging and research productivity would fall off dramatically yet fax machine sales and fax traffic remain vigorous — why?

A key reason is that faxed documents unlike e-mails are considered legal documents in countries around the globe. Another major differentiator with e-mail is that it is text-based and fax is image-based, so the scanning and sending functions are combined into a single simple procedure. A plus factor is that an image is much harder to alter or modify than a text-based document. And, the last of the "Big 3" reasons is that standard fax using the T.30 fax protocol delivers real-time delivery confirmation. There are other reasons but the above three definitely make fax an enterprise essential.

OK, we have determined that fax is critical to the enterprise and widely used, so how do we make it more efficient and productive. The answer is to install an enterprise fax server to achieve the following:

1. Legacy PSTN (define - news - alert) analog or FoIP-capable fax server technology provides desktop access and full integration with CRM, ERP, accounting and other office applications. This technology has existed for some time so it is proven and very easy to implement.

2. For the purposes of brevity, think of an enterprise fax server similar to enterprise e-mail like Outlook/Exchange. In fact, you can send, receive and manage documents from within Outlook, for example. By integrating the e-mail and fax servers, users can consolidate e-mail and fax correspondence within a single location. A key differentiator is that the top-of-the line fax solutions are simpler to install and maintain than e-mail servers.

3. Some fax servers also have true Web clients so you can send and receive faxes using any Internet connection regardless of location, such as a WiFi café or via VoIP connection.

4. Lastly, if whether you are using fax machines or legacy fax servers, you can migrate to Fax over IP (FoIP) as your business migrates from standard telephone systems to VoIP solutions thus consolidating voice and data traffic.

5. The technology to fully integrate fax into the IP traffic stream has also existed for some time. The two protocols for sending faxes over an IP network — T.37 and T.38 were both ratified in 1998, following joint development by the International Telecommunications Union (ITU) and the Internet Engineering Task Force. Again, proven technology.

T.37 is based on a "store-and-forward" asynchronous architecture like e-mail. There is no direct connection between the sender and the receiver so issues like paper out failures or busy signals cannot be resolved in real-time. Of course, without real-time connectivity, the sender does not receive an immediate notification of delivery success or failure.

The second standard, T.38, is similar to legacy PSTN transmissions since faxes are sent and received in real-time. T.38 is the fax equivalent of VoIP and the standard essential to true convergence. A fax is transferred via the Internet similar to the way VoIP call is made across the Web.

Sound simple? Well that's because it is simple. The next step is to select the appropriate fax application for your enterprise. Assuming the goal is for a fully converged IP network, the following features should be considered as basic and essential:

1. Sending or receiving a fax must be as intuitive and natural an activity as accessing any other document via the Web or e-mail.

2. Select a fax server solution that offers a thin-client interface that only requires a browser such as Internet Explorer.

3. Fax should be handled like any other form of messaging so integration with the corporate e-mail system such as Exchange is mandatory.

4. Evaluate your back office needs and determine if integration with back office application(s) is required. If so, then a .NET API and other integration tools are essential requirements.

5. If fax is critical to your operation look for an architecture that provides for fault tolerance and distributed deployment, providing redundancy and fail-over capabilities.

Lastly, I want to emphasize that all of the above technologies exist today and if you are in the planning stages of migrating to VoIP and a converged architecture, you do not have to wait to install a fax server. This is particularly relevant if you are still using standard fax machines. There are several products on the market that provide for standard analog PSTN deployment and also provide a simple migration path to FoIP **IT**.

*Max Schroeder is the senior vice president of FaxCore Inc. (news - alert) I(http://www.faxcore.com) and a Member of the Board of the Enterprise Communications Association (news - alert) (ECA; http://www.encomm.org), an industry forum promoting the deployment of voice, video, and data communications solutions in the enterprise. Schroeder is Chair of the Media Relations Committee and, in that capacity, is the ECA liaison for TMC.*





Go To Table of Contents | Go To Ad Index

# Homestar Direct: Using IP Telephony To Gain A Competitive Edge

As nearly everyone in America can attest, the residential real estate market continues to defy the odds by growing at a near exponential pace. All over the country, buyers are in hot pursuit of their "dream home," whether it's a sprawling home in the suburbs, a penthouse with a city view, or a beachside vacation retreat.

Keeping pace with this hot real estate market is the highly competitive mortgage lending arena, where literally thousands of financial institutions battle one another in search of customers looking for new mortgages, or to refinance their existing loans. This frenetic activity has resulted in hundreds of new entrants into the marketplace, resulting in intense competition and compressed windows of opportunity for lenders to attract and secure new business.

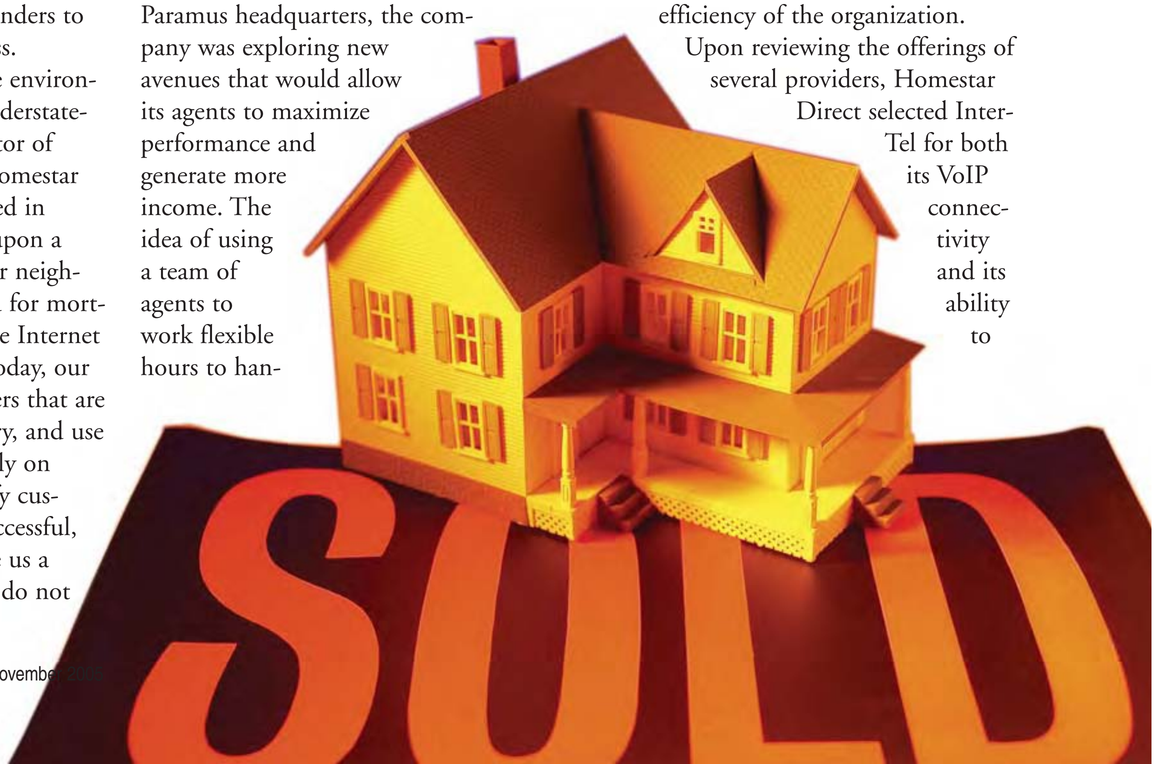
"To say this is a competitive environment would be an extreme understatement," says Jamie Troia, director of information technology for Homestar Direct, a mortgage lender based in Paramus, New Jersey. "Once upon a time, customers met with their neighborhood bankers and shopped for mortgages in person. Obviously, the Internet has changed that paradigm. Today, our company competes with lenders that are located throughout the country, and use go-to-market strategies that rely on speed and efficiency to identify customers. In order to remain successful, we leverage technology to give us a competitive edge. We literally do not have a second to waste."

Success is a subject very familiar to Homestar Direct. A division of Opteum Financial Services, Homestar Direct originates some $30 million in mortgages per month, serving a customer base that's primarily located in the Northeastern and Mid-Atlantic regions of the country.

While most of the company's 65 employees work out of the company's Paramus headquarters, the company was exploring new avenues that would allow its agents to maximize performance and generate more income. The idea of using a team of agents to work flexible hours to handle periods with high call volume made strategic sense. The challenge was to identify a technology solution that would allow Homestar Direct to implement this strategy without adversely affecting the company's customer service or productivity.

As a potential solution, Homestar Direct met with several vendors to explore voice over IP as a means to address their challenge, as well as to improve the overall productivity and efficiency of the organization.

Upon reviewing the offerings of several providers, Homestar Direct selected Inter-Tel for both its VoIP connectivity and its ability to




Finally.

deliver applications that tangibly improve business processes.

"After meeting with Homestar Direct and understanding their challenges and business direction, we felt comfortable that we could offer a solution that would provide the flexibility they need as their business grows," said Joseph Denise, chief operating officer with Norcom of New York Telecommunications Solutions, LLC, an authorized Inter-Tel distributor. "Our approach was to deliver an IP-based system that would certainly help Homestar Direct better utilize its personnel, as well as serve as a foundation that would positively impact the company's business processes by improving productivity and enhancing customer service."

By using Inter-Tel's Axxess communications platform, along with Inter-Tel's (news - alert) IP endpoints, Homestar Direct is able to maintain efficient operations seven days per week. The system's IP capability enables agents to work from their homes while allowing them to access the company's database resources and other tools to conduct business. The system is transparent to the customer, who receives the identical customer care as if they were calling into a centrally-located call center. Most importantly, the IP deployment allows Homestar Direct to remain competitive while giving it the flexibility to meet customer demand during peak hours.

"While Inter-Tel IP's system has exceeded our expectations for overcoming our local staffing issues, we're just as pleased with the reliability and elasticity of the platform," explained Troia. "The ability to add and change agents without incurring additional expenses, along with the integration of powerful applications increases our capacity to serve more customers in an expeditious and efficient manner."

According to Craig W. Rauchle, president and chief operating officer of Inter-Tel, a growing number of companies are mirroring Homestar Direct's deployment of voice over IP.

"One of the most compelling aspects of IP communications is its ease and efficiency in facilitating remote communications," he remarked. "Businesses like Homestar Direct, which rely heavily on knowledge-based workers and agents, are coming to the realization that these individuals can use IP technology to work from home offices, customer sites, or virtually any location that has broadband access. As these customers are finding out, the impact on productivity, customer service, and efficiency is substantial."

In addition to the core IP communications system, the company also elected to leverage a power dialer provided by Inter-Tel to build its outbound marketing campaigns — crucial components in the company's ability to generate revenue. Using this technology, Homestar Direct can instantly import fresh leads it receives from a variety of sources, such as online repositories and other databases directly into the power dialer, which immediately contacts these prospects. This gives Homestar Direct a substantial competitive edge by touching these potential customers in near real-time, ensuring that they reach these individuals before any of their competitors do.

"The power dialer has been a tremendous asset in our ability to turn leads into mortgages," said Troia. "In our business, speed-to-market is incredibly important. If we're not the first lender to touch a customer, chances are that individual will choose another lender. The power dialer puts us where we need to be — in the position to reach customers well before other lenders."

The last piece of the puzzle in Homestar Direct's communications system was the integration of Inter-Tel's Call Center Suite, a powerful application that provides businesses with comprehensive tools to gauge and manage both incoming and outgoing call activity. Call Center Suite provides companies with myriad real-time statistics, such as the number of agents on duty, the amount of calls in a particular queue, calls on hold, and a wide range of per-

"One of the most compelling aspects of IP communications is its ease and efficiency in facilitating remote communications.
– Rauchle

formance-based statistics for individual agents. Call Center Suite provides this detailed information to managers in a variety of graphical formats, or dashboards that can be customized to provide both complex statistical information as well as performance snapshots.

As Homestar Direct has learned, the power of Call Center Suite becomes even more pronounced when it can be directly linked to the company's customer relationship management (CRM) database. For example, Homestar Direct can cross-reference both inbound and outbound call activity with a wide range of data, such as the number of mortgage applications that are generated, and the volume of loans that are originated. This heightened functionality allows the company's managers to have immediate insight into the productivity of its agents, the quality of its leads, and the effectiveness of its various marketing campaigns.

These measurement tools are equally accurate and efficient with employees who are working from their homes over the IP system as it is with agents that are located at the company's headquarters.

"We have been completely satisfied with the way our Inter-Tel solution has performed," said Troia. "The flexibility and reliability of the IP-enabled Axxess system provides a cost-effective foundation to that will meet our business needs as we continue to grow."

"Additionally, Inter-Tel's business applications are equally integral to our success," he concluded. "They have made a substantial impact in our productivity and organizational efficiency. The entire solution continues to pay for itself many times over." IT

# Contents

Here are several articles currently on our site. For more insight, visit ipcommunications.com.

## Don't Bother With SPIT - Police Your Borders

**Examines two opposing trends:**

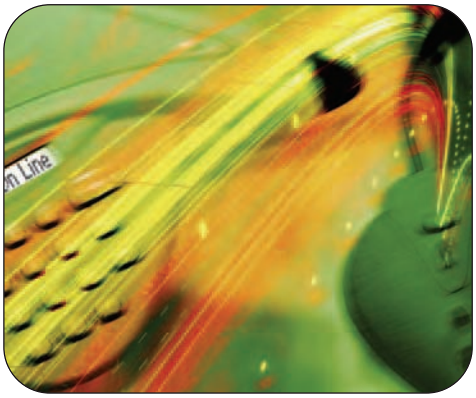
- The use of session border controllers (SBC) to provide the additional layer of security and bandwidth management.
- Methods of protecting intra-network devices via network security appliances and/or new technology that hardens the client device itself and provides authenticated user identity.

www.ipcommunications.com/dbws

## The Pro's and Cons of Interoperability

SIP-enabled interoperability also introduces "best practices" safeguards to prevent voice packet intercepts, spoofing or even hijacking. What encryption and authentication schemas should be deployed for VoIP systems? And what are the different tools available for service providers and enterprise users?

www.ipcommunications.com/pc



## Desperately Seeking Video

Everyone knows the crown jewel of Triple Play services is video. But did you know HMP actually lends itself exceptionally well to IP video services as IT professionals throughout Asia have already discovered? The virtualization of media processing streamlines a converged solution for operators both fixed and mobile. And the simplicity of a general-purpose computing platform permits for a wider array of voice and video codecs.

www.ipcommunications.com/dsv

## "A Day in the Life" series

### 1) A Day in the Life of the Consumer:

Voice and video services are no longer a reality of just the wired, landline realm. Wireless VoIP and video services have already begun to reach critical-mass adoption in pockets around the globe. How exactly can an integrated voice, video and data offering improve an individual's quality of living?

www.ipcommunications.com/aditl-v1

### 2) A Day in the Life of the Working Professional:

Real-time communications have increasingly enabled entire business entities to break down their walls and operate as virtual companies, placing more of an emphasis on the capabilities of general purpose computing platform than ever before. So what happens when a manager wants to speech-enable their IP solutions?

www.ipcommunications.com/aditl-v2

### 3) A Day in the Life of the Enterprise:

"Collaboration" has become the corporate axiom for the new millennium much to the degree "globalization" was back in its heyday during the 1980s. But how do companies of all size from SMB's to large-scale enterprises capitalize on their networking infrastructure and corporate computing platforms to facilitate interpersonal cooperation and form truly global partnerships?

www.ipcommunications.com/aditl-v3

## QoS and Beyond

A popular misconception of switching to VoIP-based architecture is an IT manager will be forced to sacrifice performance in order to realize any cost advantages. In reality, though, the abundance of diverse offerings available for virtualized Intel-based platforms addresses all the mission-critical issues like echo cancellation, latency, jitter, etc., in order to help you achieve 99.999% reliability.

www.ipcommunications.com/qos

## Multi-Threading, Multi-Core

The expected ramping of dual-core (and multi-core down the road) technology combined with the ability to process multiple threads of software lends further credence to the ability of general purpose hardware to tackle the resource-intensive tasks of media processing. And by the number of dual-core CPU's on the horizon, the future for VoIP and IP-based multimedia services looks bright.

www.ipcommunications.com/mt

# Log On Today!

## www.ipcommunications.com

A new router upgrade was required to support 10 Gbit/s interfaces, and the existing core routers were not capable of supporting these speeds.

European player.

### TINet's Network Evolution

In each of the 15 countries that it interconnects (including the seven Tiscali national networks), TINet has two interconnected routers that interface with each country's national network. The traffic growth in some of these regional networks demanded an upgrade of the existing interconnection routers. Specifically, a new router upgrade was required to support 10 Gbit/s interfaces, and the existing core routers were not capable of supporting these speeds. Additionally, the existing equipment was nearing the end of its economic lifecycle, necessitating a technology refresh for the network.

## BUILDING A NEXT-GENERATION NETWORK

### Network Requirements Established

Being innovative, TINet didn't see its situation only as an opportunity to expand capacity. TINet also wanted to install network equipment that could support higher classes of service and lower capital expenditures over a 5- to 10-year period. In November 2003, vendors were invited to offer their solutions in line with a well-defined "wish list" that was captured in a RFP/RFQ. The RFP/RFQ responses provided Tiscali with an overview of the available technologies, features, and potential commercial frameworks for a new technology implementation during 2004.

Some key requirements listed in the RFP were:

- Ability to support 10 Gbit/s Interfaces today.
- Ability to scale beyond 10 Gbit/s in the future.
- Cluster capabilities of the system.
- High-availability feature set — allowing for in-service upgrades.
- IPv4 and IPv6 — dual stack.
- Virtual or logical routers.
- Interoperability (with existing equipment and with network management software).

### The Selection Process

Based upon the review of the RFPs, TINet learned that the big players' cluster technology was not ready for prime time; hence they decided to postpone the final vendor selection. Postponing the selection gave the Tiscali technical team some time to research each vendor's solutions and distinguish fact from fiction.

2004 was the year when Tiscali investigated in detail the solutions from the various vendors who had replied to the RFP. This investigation included Chiaro's Enstara IP/MPLS platform. Throughout the year, Tiscali performed an extensive evaluation of the Enstara platform that included a visit to Dallas to perform tests at Chiaro's lab as well as final stages of testing at Tiscali.

The last phase within the technical evaluation was a field trial in the Tiscali production network supporting connections between Frankfurt and Brussels. Protocols to support the live traffic included ISIS, BGP, IPv6 and MPLS. In February 2005, after forwarding 250 Terabytes of traffic over a three-week period, TINET chose the Enstara system for general network deployment as their core router of choice.

### Chiaro's Enstara Platform Shines

After an interactive year between Chiaro and Tiscali, the Tiscali management team reached some conclusions. All the "hard" items on the Tiscali checklist were addressed:

- Technology requirements: cluster, HA, interoperability, etc.
- Financially stable channel with ECI Telecom.
- 24/7 support structure and disaster recovery with ECI Telecom.
- Competitive price.

However, there was still a "soft" item to be resolved, which had to do with the selection of brand-new technology as well as a comparatively new supplier.

The Tiscali management team made some conclusions:

- Traditional vendors and their cluster technology were "as new" as the Enstara platform
- Only the Enstara platform was able to deliver, now, on the promise of real-world high availability.
- Responsiveness from the Enstara team was impressive and was not met by other suppliers.
- Tiscali's track record of having a bullish approach to launching new, innovative services in the market justified an innovative network with state-of-the-art technology from a new kid on the block — just as they did several years ago with their previous router technology.
- Tiscali wants to have a choice beyond the traditional duopoly (i.e., Cisco and Juniper).
- Tiscali planned a phased introduction of the Enstara platform in its live network, leaving room for evaluation before continued expansion.

Tiscali put two Enstara systems into production in April 2005 at two different locations in Amsterdam. The regional Tiscali network in the Netherlands is expected to reach the 10 Gbit/s traffic mark during 2005, and therefore this network required the first transition to the Enstara platform.

Installations performed jointly by the Chiaro and ECI teams went smoothly. The two Enstara systems now carry all traffic in and out of the Netherlands for TINet. The displaced routers are redeployed as edge devices in Tiscali's network.

The Enstara systems are running several protocols, including BGP, ISIS, LDP, RSVP, IP, IPv6 (define - news - alert) and MPLS. The systems provide for connectivity to external peers, transit providers, direct customers and other



Go To Table of Contents I Go To Ad Index

vendors' core site systems.

### Chiaro's Enstara Platforms Deliver Real-World HA

One of the main benefits for Tiscali of the Enstara platforms is their real-world high availability (proven in-service high-availability beyond 99.999).

Significantly, TINET has deployed several new Enstara features and enhancements by upgrading software — all during live operation, in service, without any downtime or network disruption. The key Real-world Enstara feature responsible for these "hitless" software upgrades is Chiaro's STateful Assured Routing (STAR) technology.

Chiaro's STAR technology is a unique embedded route processor protection mechanism that assures that router-to-router peering sessions are maintained during failure, switchover and software upgrades. This means that the Enstara router control plane is always up and running, under all circumstances, improving the reliability of the overall systems beyond 99.999 percent. STAR technology supports all routing protocols including unicast, multicast, and MPLS, and it moves well beyond the commonly used graceful restart paradigm in terms of effectiveness, speed, protocols supported and scale (i.e., number of routes supported).

### TINet Expands its Enstara Commitment

The reliable operation of the existing Enstara systems in Tiscali's Amsterdam facility, and the ongoing strong support of the Chiaro and ECI team, convinced Tiscali to increase its use of the Enstara platform. In addition, the ability to do hitless software upgrades has enabled Tiscali to "rethink" its network architecture and upgrade cycles.

Within a few months of its initial Enstara deployment, TINet expanded its Enstara technology in Amsterdam and in September 2005 ordered additional Enstara platforms for its operations.

The TINet upgrade has been triggered by a rapid increase of traffic handled by the national feeder networks. When these networks hit a certain threshold, it becomes clear that a transition to 10 Gbit/s oriented technology is required. The Enstara platform, after its successful introduction and operation at Tiscali, became the obvious choice for supporting this evolution. IT

# Tiscali International Networks: Experiencing Real-World High Availability

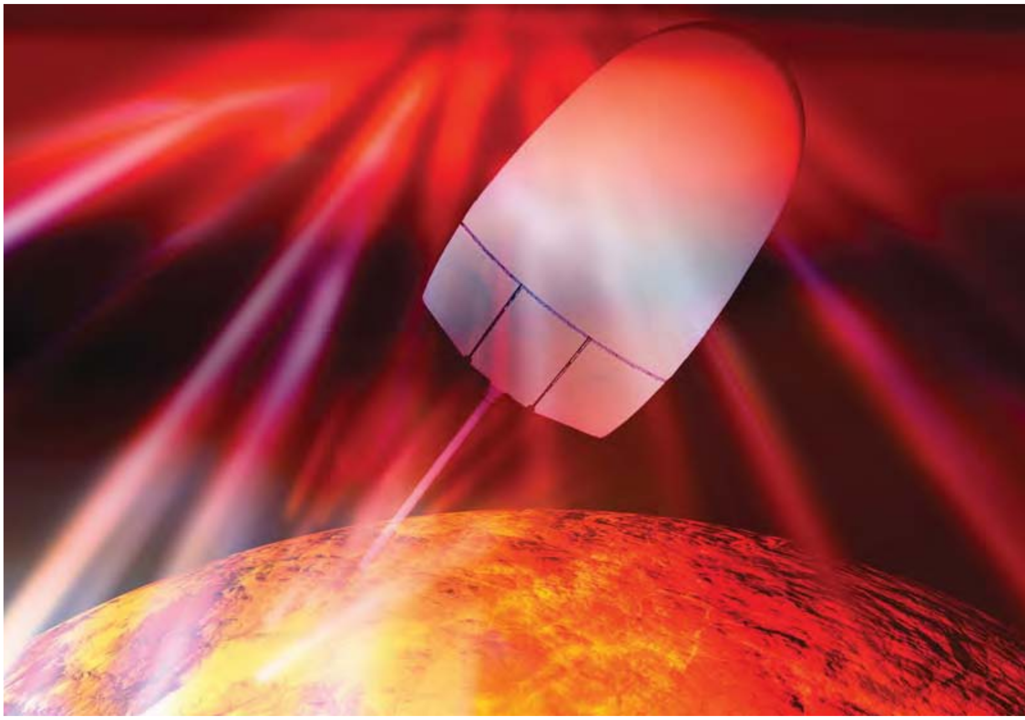
Since installing Chiaro Network's Enstara systems in its Amsterdam subsidiary in March 2005, Tiscali International Networks has experienced non-stop operation of the Enstara platform. Chiaro's Enstara platform has enabled Tiscali International Network's operations group to continuously maintain peering and live traffic during software upgrades without rebooting or interrupting network traffic. As a result, Tiscali has expanded the existing Enstara systems in Amsterdam and ordered additional Enstara systems in September 2005.

Tiscali International Network BV (TINet) is the backbone network subsidiary of Tiscali. (news - alert) It is a Tier-1 IP/MPLS network covering 15 European countries (more than 120 PoPs) plus the USA (six PoPs) as of Q4 2004, and it is still growing. TINet owns 12,000 km of fiber routes in Europe and controls an additional 40,000 km of fiber routes across the USA, along with two redundant transatlantic links. This large fiber network runs at 2.5 and 10 Gbps, on the top of which sits an IP/MPLS backbone (with AS3257).

TINet focuses on delivering leading-edge clear-channel, voice and IP-based turnkey solutions to carriers, xSPs (including Tiscali in its seven country networks) and multinational corporations. Supporting today's largest European VoIP (define - news - alert) implementation, TINet is becoming one of the largest and fastest-growing carriers throughout Europe and will likely become independent over time.

Key network assets of TINet include the following:

- Wholly owned fiber backbone running DWDM for scalability.
- End-to-end MPLS-enabled IP network delivering traffic engineering and QoS.
- 200+ peering agreements, the largest ISP community for a





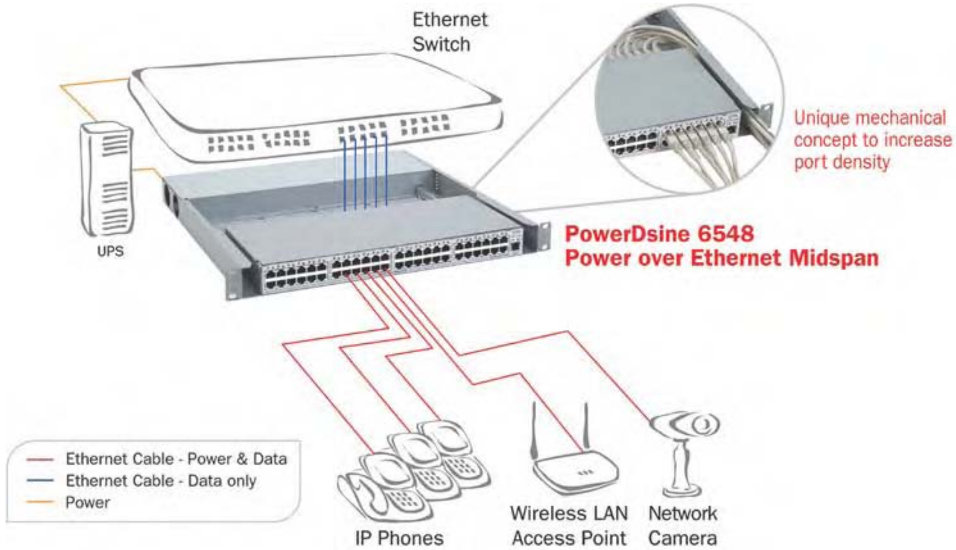


Go To Table of Contents | Go To Ad Index

## PowerDsine PD6024G & PD6548 midspans

**PowerDsine, Inc.**
**290 Broadhollow Road**
**Suite 305E**
**Melville, NY 11747**
**Tel.: 631-756-4680**
**Web: http://www.powerdsine.com**

*Price: 48-Port Managed — $1,790; Gigabit Midspan — $1,399*




PoE (Power over Ethernet) Midspan units are installed between standard Ethernet switches and IP phones, WiFi access points, and IP security cameras without modifying the existing network infrastructure. Midspans add 48V DC power to each Ethernet cable, to remotely power IP devices that support the Power over Ethernet IEEE 802.3af standard centrally from the switch closet. To enable use in legacy installations, a midspan adds PoE capabilities to existing hubs and switches by injecting power into the twisted-pair cabling. Additionally, you can provide battery backup to all of your dispersed IP phones, access points, and IP cameras using a single centrally located UPS system.

One of the leaders in the PoE midspan market is PowerDsine, (news - alert) who has shipped over 2 million Power over Ethernet Ports. They sent us a couple of their midspans to test, including the industry's first 24-port Gigabit midspan (PD6024G) (Figure 1) as well as their 48-port 10/100Base-T midspan (PD6548), which is the industry's largest and first 48-port midspan.

Each of the PowerDsine models lets you assign an IP address for SNMP management as well as for Web-based access to the unit. Also, the latest version defaults to IP address 192.168.0.50 so you can change the IP address without needing a serial cable along with HyperTerminal to first assign an IP address, which is always a pain. The Web interface is a nice addition since many SMBs don't have HP OpenView or other SNMP software to manage SNMP devices. Having a standard Web interface gives them another option for managing their network equipment. Additionally, the units sport a RS-232 serial interface which supports a very fast 38,400 baud rate perfect for uploading new software/firmware updates.

The real trick is to prevent the midspan from damaging equipment that's not PoE- compatible. To do this, the PowerDsine midspan performs a discovery process that runs at power-on, as well as every time the user plugs a device into a PoE port. It also detects when the user disconnects a device and disconnects the port within a few hundred milliseconds. According to PowerDsine, the discovery process was one of PoE's major challenges and in fact, PowerDsine helped write and ratify the IEEE 802.3af standard. PowerDsine actually leveraged their expertise in designing ring voltage generators and applied that knowledge from the telecom world to the datacom world. In any event, the method used to detect devices connected and disconnected uses "resistance discovery" looking for signature impedance by applying a current limited "test signal" to the cable and looking for return voltage.

Both the PD6024G (Gigabit 24-port midspan) and the PD6548 (48-port midspan) are constantly calculating power consumption in real time to ensure an overload condition does not occur. If it is near an overload condition it won't let a new device to be added. It assigns priority for the power by the port number, so if you reach an overload it will disconnect the lowest priority device. For instance, this lets you prioritize placing the CEO's IP phone at port #1. In

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





Go To Table of Contents | Go To Ad Index

**Figure 1. PowerDsine offers the industry's first 24-port Gigabit midspan (PD6024G).**

addition, both units keep a "power gap," a little bit of "wiggle room" for safety to make sure you never reach an actual overload condition. Each of these midspans supports up to the maximum standard 15.4W per port with 200W of total power from the power supply. Of course, if you take the either model and attempted to draw the maximum 200W you would only be able to connect 12 ports at the maximum before you'd exceed the 200W capacity with the 13th device. However, very few devices require a full 15.4W of power, so you wouldn't need a stronger power supply any way. In fact, most IP phones only use 5.6W, so you'd be able to power 24 IP phones with power to spare.

The midspans have temperature sensors both on a per port basis as well as system-wide to make sure you don't overheat. One really nice capability of independent sensors is that if a port overheats only that port is turned off. Obviously, if the system-wide sensor reaches a certain threshold then the entire system is shutdown. LEDs for each port indicate the status of the units including yellow to indicate an overload, as well as orange to indicate a system fault. Both models are compatible with MIB-based network management platforms (SNMP — Figure 2). The PowerView software monitors IP Phones for possible failure as well as IP phones status alerts: power fall, disappearance and malfunction.

TMC Labs performed extensive tests on each of the models and used a variety of PoE-compatible IP phones to see if the midspans would detect and properly power the IP phones. We connected Cisco phones, Pingtel xPressa phones, and snom phones. All of the phones were successfully powered by the PowerDsine models. PowerDsine also shipped us a special test unit that, when plugged into the midspans and using a special dial, we were able to select the amount of power drawn. As we increased the dial setting we were able to draw enough wattage to simulate an overload condition. The port would then go offline and the LED would indicate an overload condition. Once we turned the dial back down to draw less power  the port would then go back online.

We were quite impressed with both the PowerDsine 24-port Gigabit midspan (PD6024G) and the 48-port 10/100Base-T midspan (PD6548). With a built-in Web interface and SNMP capabilities, making sure your mission-critical IP phones or even PoE-aware WiFi access points and PoE-aware cameras are online and trouble-free is very easy to do. TMC Labs would not hesitate to recommend either of these PowerDsine models to suit your power over Ethernet needs. IT



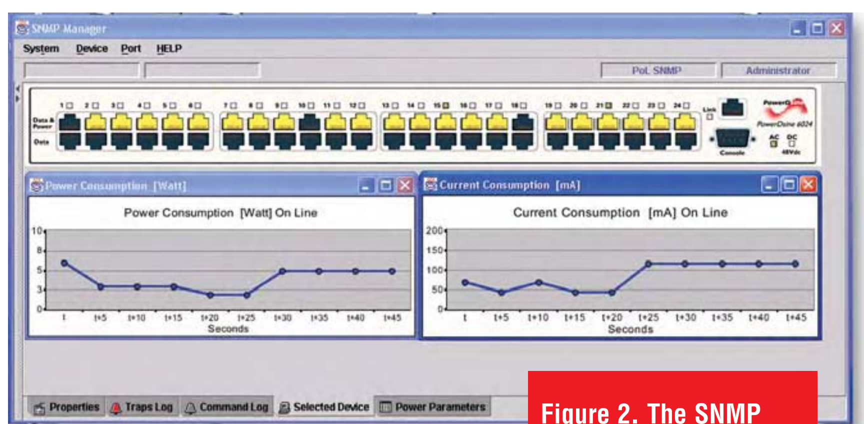



**Figure 2. The SNMP manager indicating power and current consumption levels.**

## SmartPro 2U Rack/Tower Extended Run UPS

**Tripp Lite**
**1111 W. 35th St.**
**Chicago IL 60609**
**Tel.: 773 869-1111**
**Web: http://www.tripplite.com**

*Price: $1,139.00*

Power protection products are a critical staple for any enterprise and they are continually becoming more advanced with more features to ensure critical equipment has power.

For instance, UPS manufacturers have developed tools to lengthen the life of the battery, warn of overloads, send e-mail alerts, manage via the Web or even a PDA, as well as warn the user that the battery needs to be replaced. TMC Labs examined a line interactive UPS from Tripp Lite (news - alert) called the SmartPro 2U Rack/Tower Extended Run UPS with models ranging from 500VA to 5000VA. We examined a model featuring 3000VA, 120V, and 60Hz sine wave. The SmartPro series informs you when it's time to change the battery, supports SNMP management, and has a plethora of other features. We should mention that as IP-PBXs are becoming the phone system of choice, power protection manufacturers are ensuring that their line of products can keep mission-critical IP-PBXs up and running. In fact, the SmartPro series has been tested compatible with Cisco CallManager, versions 3.3 and 4.0 and has earned the "Cisco Compatible" logo.

The SmartPro 2U is a line-interactive UPS that can be rack mounted either horizontally or vertically and comes with the screws, nuts, washers, and brackets necessary for mounting. It installs in standard 19 inch rack enclosures with an installed depth of only 18 inches. The SmartPro line supports 15-amp, 20-amp and 30-amp AC outlets. On the front of the UPS features several LEDs, including a Power LED to indicate it is on, a Voltage Correction LED indicating it is correcting high or low AC voltage (without the assistance of battery power), and an Output Load Level LED, which is multi-colored and will change from green, to yellow, to red (overload) to indicate the load condition. Two other LEDs are also on the front of the unit including a Battery Charge LED colored green, yellow, or red, depending on the amount of battery charge left and a Battery Warning LED, which indicates that the battery needs to be changed.

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

The SmartPro supports multiple interfaces, including 2 USB, 2 DB9 (serial) and SNMPWEBCARD (optional) giving you multiple management and monitoring options that work in conjunction with their PowerAlert software (included). PowerAlert UPS monitoring software supports safe unattended shutdown, monitoring, and control via local connected servers, plus any number of additional servers over IP. A remote shutdown server (Windows, NetWare, and Linux supported) allows for multiple servers to be shutdown via a network message from a PowerAlert UPS engine that is monitoring the UPS through a communication port.

The unit also features an Emergency Power Off (EPO) port that can be used to connect the UPS to a contact closure switch to enable emergency inverter shutdown. One really nice feature is the power sensitivity adjust-





Go To Table of Contents | Go To Ad Index


Minacom
PowerProbe 30
Testing
PocketDQ

# 2006: The Year Of VoIP Peering

**INTERNET TELEPHONY** Magazine's Rich Tehrani has proclaimed 2006 the year of VoIP Peering.

## 3rd VoIP Peering Summit at

## INTERNET TELEPHONY Conference & EXPO

## Thursday, January 26, 2006

The VoIP industry is abuzz with peering announcements and excitement as service providers, universities, government agencies and enterprises begin to peer directly with one another eliminating the fees, taxes and quality degradation associated with the PSTN.

Your first opportunity of the year to learn about this exciting new field will be at IT EXPO in Fort Lauderdale, Florida where the third standing-room only VoIP Peering Summit will take place.

**January 24-27, 2006**
**Fort Lauderdale-Broward County Convention Center**
**www.itexpo.com**

*The VoIP Authority Since 1998*

TMC Fort Lauderdale INTERNET TELEPHONY CONFERENCE & EXPO EAST ment dial which can be used to adjust how sensitive the UPS is to sine wave distortion. In area with poor power generation or chronic wave distortion you can reduce the sensitivity so that it doesn't switch to the battery too frequently draining its reserves. The SmartPro 2U also allows you to connect an external battery pack for extra runtime. The SmartPro features a dual boost AVR that corrects brownouts to 79V and overvoltages to 147V.

Importantly, it features current monitoring along with three controllable load banks. It also has nine outlets (4 5-15R/4 5-15/20R/1 L5-30R), as well as NEMA L5-30P input. Tripp Lite includes a 5-20P replacement plug for installations that have 5-20P wall outlets instead of L5-30P wall outlets.

We did some tests on the unit including hooking up a wide variety of TMC Labs equipment to see if we could force an overload condition. Eventually, after enough equipment was connected we did get an overload alert. Next, we pulled the plug to simulate a power outage. All of the PCs stayed online and immediately we could hear the internal cooling fan in the SmartPro get much louder than normal, no doubt due to the increased load and from having to power from the battery and the resulting heat increase.

Tripp Lite backs up their product with a $250,000 Ultimate Lifetime Insurance policy, which may give peace of mind to their customers. Additionally, both the external battery packs and the internal batteries are hot swappable, which means no downtime when its time to replace the batteries. Overall, TMC Labs was quite pleased with the number of management interfaces, the excellent runtime, the sleek black design, and the powerful PowerAlert software and would not hesitate to recommend the SmartPro 2U to suit your power protection needs. IT

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

# Making The Business Case For VoIP & IP Telephony


What do you think when you hear the terms VoIP and IP telephony? Some may use these terms interchangeably, but they are in fact different technologies. VoIP describes the actual method of transmitting voice over an IP network; IP telephony (IPT) describes telephony devices that use IP as the native transport for voice and call signaling. IP telephony needs VoIP (define - news - alert) to send calls over the network; VoIP does not need IP telephony. An IP network that supports voice (telephony), video, and data with proper quality of service features is also referred to as a converged network. Now that we are all on the same page, we can move onto the business justifications and deployment scenarios for both.

There is no doubt that VoIP is here to stay, so chances are you are probably contemplating a migration — if not now, then potentially in the near future. The important part to consider is that your business has many options for a deployment strategy, and your business drivers should dictate which you exercise.

This article will help you with your decision: When should you implement VoIP and IP telephony, and what steps should you consider when making the upgrade?

The first step in determining if IPT and VoIP are right for your company is to understand what potential benefits a converged network provides and factor them into your ROI calculations.

At first, hard cost savings were the primary business drivers behind VoIP. These savings include reduced toll charges from long distance, conference bridges, and PSTN service providers as well as capital and operational savings from utilizing shared resources for multiple sites and reducing administration tasks. For example, a Virginia-based client with a development center in Asia recently recognized a $40K/month savings on long distance, which helped pay for the new system. And while cost savings are still key drivers in converged network deployment, other benefits are fast becoming equally as compelling and should not be ignored, such as employee productivity, employee retention, and office real estate savings.

VoIP and IP telephony increase employee productivity by literally changing the way your company does business. Companies that deploy a converged network gain the benefits of virtualization; an employee's physical location becomes irrelevant and tasks can be distributed to anyone, anywhere on a converged network — with very little cost or effort. Complimentary technologies such as unified messaging can deliver e-mail, voice mail, and faxes to a single source, creating a truly cohesive communication experience.

VoIP also offers resiliency and business continuity, which TDM telephony systems simply cannot; this is because VoIP is based on IP, so it leverages the decentralized nature of IP networks. Call Processing Controllers can be geographically dispersed and clustered for





Go To Table of Contents | Go To Ad Index

By Lawrence Imeish

failover. A properly designed and deployed IP telephony network could easily roll calls over to another office during an emergency. Employees can even work from their homes if necessary and utilize an IP softphone and a VPN connection to give them almost all of the functionality they enjoy at their desks.

An important differentiator between IP phones and their TDM predecessors is the fact that the IP phone is not "just a phone" — it is an informational device offering a host of new and integrated applications that are accelerating return on investment and facilitating collaboration. Law firms can now reduce lost billings by utilizing an application that meets requirements like call tracking, billing, and cost allocation. Universities can now allow class registration and post schedules on IP phones. Government organizations are currently using IP phones to receive emergency broadcasted messages right to their screens, speeding up response times. Schools are using them to take student attendance more quickly, and hotels are using them to check in/out hotel guests more efficiently. The ability of these new applications to offload work from people and PCs means that companies gain process efficiencies, which translate into measurable cost savings.

The next step is to understand where in the cycle of your current telephony infrastructure you are, and what possible upgrade opportunities are approaching. Questions to consider are:

Q. How long is the lease on my current telephony system (PBX, etc.)?

A. If you just purchased or leased a PBX in the last year, then a full-blown IPT deployment may not be for you.

Q Will we be relocating or opening a new office in the near future?

A. A converged voice and data network is a perfect fit for office moves, not only does it cut down on the required cabling, but it makes the upgrade to IPT easier because users are already expecting "new things" at the new office.

Q. Are there any features that IP telephony provides that will benefit the business and are not currently available on the current TDM telephony platform?

**There is no doubt that VoIP is here to stay.**

A. Government-mandated business continuity or company-wide initiatives for telecommuting are very powerful IP telephony drivers.

## Building The Foundation And Minimizing ROI Costs

Whether or not you are ready to implement a converged network right now, odds are you will in the coming years. Consider this possibility with every networking purchase, and you will reduce the cost of tomorrow's converged voice and data network today. For example, if you're in the process of buying new routers for your remote branches, make sure those routers are easily upgradeable to also serve as a voice gateway. If you need a new Ethernet switch, make sure it supports Power over Ethernet (PoE). Usually the cost difference between a PoE Switch versus a non-PoE switch is minor, and these upgrades are not generally used in the final ROI calculations as your organization needed them anyway. If your company insists on using these upgrades in the calculation, then only the delta between a base router or switch and an integrated router or POE-enabled switch should be used. Including the cost of existing hardware support in your calculations makes a strong case for upgrading. If you've depreciated the cost of your router and it is essentially "free," remember to add in the cost of support when calculating ROI. An old router that adds $7,500 in support charges each year isn't really free.

The final cost to add to your ROI model is the cost of the call processing, IP telephony handsets and voice gateways, as well as the services for designing and deploying the new network. Usually at least 50 percent of cost is made up by the handsets, and the rest by the support infrastructure.

## Assess Your IT Environment

When developing a business case for IP-based communications, a readiness assessment is very important. It provides many important details regarding the current state of your network and its ability to support VoIP. If done correctly it gives you a clear understanding of what network elements are ready for converged communications, which need upgrading, and which need to be replaced. A good assessment will make your budgeting and forecasting more accurate and ultimately result in a better deployment.

The best converged IP networks came about through thoughtful planning and a gradual, sensible migration. Companies with multiple offices where LAN infrastructure currently does not support IP telephony can start with VoIP between offices on their WAN, while still using their existing PBXs and handsets. This allows them to quickly reap the rewards of toll bypass and reduce long distance charges—without requiring a complete revamp of their network. Then, as the LAN is upgraded to meet the requirements of a converged network, they can systematically phase in IP telephony at remote offices and headquarters and build on their existing VoIP WAN. The most important part of any IP telephony migration or VoIP deployment is integration with existing TDM devices.

Whether it's the CEO's favorite conference phone, or a PBX (define - news - alert) in a closet, a migration plan must take into account the challenges of a temporary interconnection between the IP and TDM worlds. This is where choosing the right solutions provider becomes critical. Many providers can design and deploy a pure IP telephony network; very few providers have the technical skills, project management experience and successful track record to implement phased deployments. Just as important as the design and deployment, an ROI calculation must also take into account ongoing support. The most desirable scenario is one where the people that designed and deployed your IP telephony network also support it. This eliminates the opportunity for finger pointing or deflecting responsibility. Your ROI calculations will not do you any good if it is 9 a.m. on Monday, and your network is down and you have no one to call. You must evaluate how reliable a partner's support solution will fit into your company's worst-case scenarios.

## Have A People Plan

Whether you want to do a gradual implementation or a forklift and replace depends upon your circumstance – either scenario is plausible. Regardless of which you choose, your plan needs to consider the human factor, since a converged network is as much about business processes and the people using the system, as it is about technology. The users will need to be trained before they are migrated; this cost should be considered in your calculations. A good deployment plan takes training into account from the beginning. The better trained the users are the more likely they are to use the advanced features of the IP phone and the more benefits realized by your organization. Finally, a deployment will affect your support groups; no longer will the data and voice teams work independent of one another. The combining of these two groups centralizes all of the networking and voice skills and creates an environment of valuable knowledge transfers, making each member twice as valuable to the organization. IT

*Lawrence Imeish is solutions architect at Dimension Data, (news - alert) a services and solution provider that helps clients plan, build, and support their IT infrastructures. For more information, please visit the company online at http://www.dimensiondata.com.*

# How Cable Operators Market Telephony

Today cable operators have a unique opportunity to leverage their last mile access to subscribers' homes via coaxial networks and deliver a full suite of digital converged services. Video on demand, digital television recording, high-speed Internet and now, digital telephony using VoIP technologies are all available to operators looking to generate new revenue opportunities and create more value for their customers, also making it more difficult for the subscriber to cancel their service. The opportunities cable operators have today are real, but need to be integrated with the rest of the cable operators' offering to fully leverage the power of the triple play. Many cable operators today position their phone service as a traditional phone service, sometimes calling it digital phone; some operators such as Cablevision, Time Warner, and others rarely market the service as Voice over Internet Protocol (VoIP)-based technology.

## Competitive Threat Or Opportunity?

Cable operators face as many threats as they have opportunities. With satellite operators and DSL operators presenting viable competitive offers at a discount to cable's video and high speed data services, VoIP (define - news - alert) bypass operators also present a new threat leveraging cable's broadband infrastructure. These parasitic telephony startups present a viable challenge as they target early adopters (typically through Web-based marketing) and compete on price drawing from cable subscribers who are most likely to spend on the new services. These providers generate publicity and establish a reputation and a pricing threshold for VoIP-based cable telephony.

Lastly, incumbent telephone companies (Baby Bells, in particular), which are most directly threatened by cable operators and are responding to the cable threat by reducing the price of their DSL broadband service and their traditional all-inclusive voice plans (e.g., SBC's recent reduction of its unlimited calling plan to $40, not including taxes and fees), introducing their own VoIP service and even testing their own video services to be delivered over DSL.

## How To Deploy Voice

After determining whether to build their own phone service or outsource the development effort to a service provider such as Net2Phone, operators must determine whether the telephony service will be positioned as a primary replacement service versus an alternative line service similar to the VoIP-based telephony startup. The quality of the overall service and how it is positioned determines how much the cable operator can charge for the service. Primary line replacement services tend to be priced from $35 to $40 for unlimited call plans, while the alternative line services tend to be priced from $20 to $29

By Richard Gilbert

for its unlimited calling plans.

Leveraging the PacketCable specification, cable operators are supporting a DOCSIS 1.1 (or better) broadband network and have the ability to offer higher levels of quality of service (QoS) to further differentiate their voice service from the VoIP startups. The specification provides end-to-end QoS by enabling the cable operator to identify voice packets and reserve pre-allocated bandwidth.

Deploying a DOCSIS 1.1 network allows the operator to avoid any contention with other high-speed Internet services and eliminate some of the typical problems with Internet telephony that may lead to degradation of quality including packet loss, latency and jitter. A primary line service can be positioned as a viable replacement to the incumbent phone company and be bundled with cable's video and data service.

Other factors impacting the positioning of the service include the installation approach of the new voice subscriber used by the cable operator, whether to offer number portability, and the package pricing of the service vis-à-vis the other services offered in the bundle.

## Installation Approach

While it has become easier to install

cable phone service, allowing your customers to install the service themselves not only presents higher degree of risk of churn or cancellation, but also eliminates the opportunity for the cable technician to interact with the subscriber, test the line, finalize the number portability (if applicable), and most importantly, having the opportunity to take over the in-house wiring, enabling subscribers to utilize all the phone jacks in the home. By offering an installation service (free of charge) the operator conveys a sense of quality control that supports the overall message of quality and value, enabling the cable operator to convert the subscriber from the incumbent.

Many operators feel the cost of a truck roll ($75 to $125) is justifiable, especially for new triple-play subscribers (allocating the cost over three services). However, even for a single service such as voice, the operator has the opportunity to "touch" the subscriber, explain the features and benefits, and possibly even identify and promote cross-sell/up-sell opportunities, making the truck roll an important differentiator. If a port activation is involved, the scheduled install date should be the day of the port (firm order commitment/FOC date) offered by the losing LEC, in order to limit the installation to one visit.

## Number Portability

Local number portability allows subscribers to change their phone providers while keeping their phone number. However, offering number portability presents many operational challenges for the cable operator from the submission of the local service request (LSR) to the losing LEC to coordinating the timing of the installation with the availability of the port and the collection of a letter of authorization (LOA). (LOAs are mandated on a state by state basis, principally to alleviate the slamming that occurred between long distance providers in the 80's & 90's. Some states may require third-party authorization (TPA) in lieu of a LOA.)

## Pricing

Today, price appears to be the greatest driver for switching service providers and adopting VoIP-based cable telephony service. According to recent Sanford Bernstein research, Verizon's unlimited long distance Freedom Plan costs $45 per month and more than $60 when taxes and fees are factored into the total monthly invoice for traditional voice services (including voice mail). VoIP services do not have such fees, allowing cable operators to offer an all-inclusive price that is still 50 percent below the incumbent (Table 1).

It should be made clear, that these taxes and fees do not represent the potential savings offered by cable operators today as Verizon's Freedom plan is still five to ten dollars more expensive than Time Warner Cable's Digital Phone or Cablevision's Optimum Voice, but still represent additional savings enjoyed by the subscriber.

**Table 1:** Representative costs from a sample customer bill from Verizon's Freedom Plan.

| Additional Fees Charged by LECs | (in Dollars) |
|---|---|
| FCC Subscriber Line Charge | $6.27 |
| Federal Universal Service Fund Surcharge | $0.64 |
| 911 System/Emergency Response Fee | $0.90 |
| Federal Tax | $2.09 |
| State Tax | $4.39 |
| Federal Universal Service Fee – Long Distance | $1.31 |
| **Total** | **$15.60** |

**Cable operators face as many threats as they have opportunities.**

For most operators pricing the service and communicating the savings is the call to action driving the subscriber to the voice service and appears to be the most challenging when offering telephony. How much of a discount to the incumbent should be offered; how should cable operators value other services such as voice mail, Web-based subscriber account center, and finally, how the service should be priced as part of the bundle are all considerations when determines when determining a price and how to best communicate it.

Ultimately, telephony's greatest measures of success for cable operators are the ability to reduce churn and drive new revenues. In addition to these benefits, one of the most surprising statistics we have seen has been the incremental growth of 10 to 20 percent of new video subscribers. The fact that operators are able to grow their video base introducing a basic service such as phone is significant and is encouraging many other operators to further invest in phone service to expand their base. Much of the success is due to the clear and simple positioning developed by these operators emphasizing price, quality and convenience. IT

*Richard Gilbert is vice president, Strategic Planning at Net2Phone Cable Telephony LLC. (news - alert) For more information, please visit http://www.net2phone.com.*

# VoIP Opening Night:

## Are Service Providers Truly Prepared?

It's opening night for a new play. The crowd settles in its seats. The orchestra tunes in the pit. Finally in costume and makeup, each actor knows his lines. However, there's a catch: the actors have rehearsed separately, not together. Not one actor has yet heard the others speak their lines.

One can bet on miscues, confusion, dropped lines, and a weak performance at best. The audience will snicker, then remember the ticket price and walk out ready to spread the bad word. The critics will not be kind.

Testing the readiness of a group of actors separately rather than together may seem the height of foolhardy behavior. To truly view, troubleshoot, and refine what the audience will experience, requires viewing and listening to all the actors, lighting, props, and the rest operating together.

Yet a similar scenario awaits any Internet telephony provider who fails to analyze end-to-end performance of VoIP networks. Testing each piece of network equipment in a lab, while necessary, will reveal little about how that equipment will interoperate with the rest of the network in delivering a quality VoIP performance when the curtains are raised to a paying audience.

To ensure a flawless opening night, VoIP providers must supplement their lab testing and conduct pre-deployment VoIP readiness assessment on the actual network. Paying subscribers have proved themselves willing to accept some quality tradeoffs in exchange for valuable benefits such as mobility. However, for lifeline voice and premium IP video and multimedia services, providers should expect little subscriber tolerance for quality that fails to meet or exceed expectations set up by years of user experience.

In short, voice and video make measuring quality of experience (QoE) more critical then ever.

### Distributing Testing For Distributed Networks

Assessment of network readiness during pre-deployment should be conducted under a variety of scenarios, evaluating each network link under stress — testing not only of peak capacity, but also utilization curves over time. Common VoIP (define - news - alert) scenarios also will include interconnections of on-net VoIP with the public switched telephone network (PSTN). Consequently, testing should cover not only IP, but also hybrid IP/PSTN links and protocol conversions.

Readiness assessment also requires distributed, link-by-link testing, if it is to reflect the end user's actual experience of VoIP and IP multimedia applications. For example, if the operator wishes to validate VoIP readiness in a four-city network, then tests must be applied to each city and every link. Testing each link produces an end-to-end picture of QoE, protocol conformance, and other factors from City A to Cities B, C and D; from City B to Cities A, C and D; and so on.

Through distributed testing the operator can then isolate specific problems. For example, the cause of dropped packets in the B to C link might be traced to a concurrent rush of multicast IP video for training sent across the entire network. Indeed, the mix of voice, video, and data at any given time will affect performance of real time applications making it imperative that validation testing takes this larger picture into account.

Distributed test systems that place test platforms across the network can yield substantial cost reductions and time savings, compared to moving test gear and repeating tests in serial fashion at each location. The distributed approach also yields integrated reporting and analysis that covers the entire network, rather than requiring rationalization of multiple reports.

### Real Time QoE: A Multidimensional Challenge

Network readiness assessment for real time voice and video applications requires a wide range of testing capabilities. Among requirements that test system suppliers should meet are the abilities to:

- Emulate real-world traffic from multiple points on the network;
- Test the full range of IP communications and collaboration applications likely to travel the network; and





Go To Table of Contents | Go To Ad Index

By Bahaa Moukadam


- Test the full range of network statistics and media quality, including packet loss, jitter, delay, voice and video MOS, and echo.

Key call statistics will include such metrics as call setup time, percent of call completions and call termination reason. Key QoE analysis may be derived from metrics such as Mean Opinion Score (MOS) and Perceptual Evaluation of Speech Quality (PESQ).

The QoE issues examined below represent some of the most common problems encountered in readiness assessment. These issues are best tested and evaluated during pre-deployment using a distributed network assessment solution.

## Data Packet Loss

Dropped packets that produce no hiccups for non-real-time applications like e-mail and file transfers can create unacceptable degradation of voice and video traffic quality.

That degradation may appear as garbled voices on a phone call to graininess and dropouts in a video signal. Enough packets lost, and entire spoken phrases or frames of video will not reach their destination. Severe packet loss can cause a break in the entire voice or video session.

Testing of packet loss must take into account not only the voice and video

Voice and video make measuring quality of experience more critical then ever.

bearer traffic, but also signaling information (such as Session Initiation Protocol, or SIP, and H.323 signaling) required to set up and tear down connections.

TCP/IP routing technologies such as Differentiated Services (DiffServ) and IP Precedence packet markings are now being used to instruct routers and gateways to give real-time applications priority treatment in a congested network.

## Delay/Latency

Latency, or delay, causes two problems — echo and talker overlap. Echo is caused by the signal reflections of the speaker's voice. Echo becomes a significant problem when delay is greater than 50 milliseconds. Talker overlap becomes significant if one-way delay is greater than 250 milliseconds.

Exceeding these acceptable delay/latency levels can make for a voice call in which parties are forced to pause, waiting for the other party to hear what they've said. Video impairments can include frozen frames, "tiled" images or, in videoconferencing, sound out-of-sync with lip movements. These problems can be solved by QoS mechanisms that monitor and buffer packets as they travel across the network, ensuring that they are transmitted with acceptable amounts of delay and in the proper order.

Because delay can be caused by constantly changing factors like traffic volume, testing should be conducted under varying traffic scenarios to accurately determine maximum latency that may occur, and to properly configure network capacity and QoS solutions to accommodate that level.

## Voice Echo

In a hybrid network, analog phones can be the source of echo, but an IP telephony configuration is also susceptible, since some calls will inevitably interface with TDM (define - news - alert) network elements at some point when they go "off net." Factors such as the volume and delay of the echo and the impact of hardware, such as handsets and routers, can contribute to the echo level.

Echo cancellation and suppression processors, generally used in media gateways, audio coder/decoders and other equipment, are the most common fixes for unacceptable echo. Consequently, providers must test both echo itself and the performance of echo cancellers.

## Voice And Video Jitter

The human ear can detect too much variation in delay (jitter) as inconsistency in quality or simply something unnatural about the voice. Jitter is caused by actions like queuing and routing that affect the path of packets through the network. Network congestion generally produces higher levels of jitter, but quality of service controls like queuing and bandwidth allocation can control the problem.

Video tolerances for jitter are even lower, given the demand for smooth motion in film, sports or other TV applications. While great advances in video compression continue to squeeze more and more video information into less bandwidth, compression does not address issues that generate delay or jitter, such as congested router ports or malfunctioning buffers at any given point the end-to-end path. Most QoS solutions also include a jitter buffer, which adds small amounts of delay to packets received so they all appear to have equal and acceptable amounts of latency. This also ensures that packets are transmitted in the correct order.

Testing of the performance of these QoS mechanisms, as well as testing of application quality from the end-user's point of view (i.e., from the origination and destination end-points), is essential to determining how much jitter a network will experience and how it should be handled.

## Clipping

Clipping within VoIP calls occurs when either the beginning or end of words, or whole words, seem to be cut off during a conversation. This can occur when voice activity detectors and other solutions that work in tandem with echo cancellers are thrown out of sync.

Echo cancellers deal with background noise and "double talk" in addition to echo. For instance, an absence of background noise during a call can confuse users into thinking a call has been dropped. Yet improper levels of background noise can result in annoying voice clipping.

## Winning The Customer On Opening Night

By addressing each of the QoS issues detailed above, service providers can substantially increase their chances to win the hearts and minds of paying subscribers. Failing to resolve these issues effectively prior to service turn up will result in subscriber unrest and revolt.

Network operators who undertake distributed pre-deployment testing on VoIP services stand the best chance to pull off VoIP opening day to rousing reviews.

As an added bonus, operators employing these tools to ensure the performance of demanding, mission-critical VoIP services can enjoy maximum confidence that their networks will support high quality delivery of all other "triple play" real time and non-real-time applications for some time to come. IT

*Bahaa Moukadam is vice president, IP Telephony at Spirent Communications, (news - alert) a provider of integrated performance analysis and service assurance systems that enable the development and deployment of next-generation networking technology. For more information, please visit http://www.spirentcom.com.*


CallCenter Anywhere®
Open the world to your possibilities.™
Most vendors can't extend your contact center technology resources to service remote contact center locations and, at the same time, enable your local business units to manage their own technology-driven business processes. As a result, multi-site organizations are challenged to gain economies of scale by sharing IP-based infrastructure and software licenses across sites.
Whether deployed in-house or hosted by a world-class service provider such as MCI, Siebel or TELUS, Telephony@Work's CallCenterAnywhere multimedia IP contact center solution is designed to empower resource sharing across locations – while enabling local managers to maximize productivity by optimizing their own business processes in real-time. No sacrifices.
TELEPHONY @work®
ADAPTIVE IP CONTACT CENTER TECHNOLOGY
Contact us for a free information kit | www.telephonyatwork.com | 858.410.1600 | 888.854.4224 toll-free | infokit@telephonyatwork.com
© 2005 Telephony@Work, Inc. All rights reserved.

# Beyond Triple/ Quadruple Plays

## *Interworking SIP and PacketCable NCS over an IMS Architecture Can Deliver Feature-Rich Multimedia Applications Quickly and Profitably*

Convergence may be today's buzzword, but to deliver its full potential, it must go far beyond delivering silos of voice, data, and video applications over the same bit-pipe. Instead, new revenue-generating applications and services that blend triple-play media across both fixed and wireless networks must offer more value to end users than simply lower prices on POTS equivalents.

In recent years, the development and increasing deployment of the session initiation protocol (SIP) is helping to lay the track needed to bring this potential to life. And, for cable service providers, SIP interworking with the PacketCable network call signaling (NCS) protocol within the IP multimedia subsystem (IMS) architecture will open up major opportunities for multimedia applications and mobility services.

These new applications and services will blend media and offer users presence-awareness and location-based features across a unified network domain, a common interface across endpoint devices, and mobile transparency whether at home, in the office, or on the go. In turn, these applications and services will provide service providers with new revenue streams and greater competitiveness. They will also help attract and retain customers.

Looking back, CableLabs — an interoperability testing body backed by a consortium of multiple system operators (MSOs) — developed its PacketCable specs to define a complete architecture for providing primary line telephony services over a data over cable service interface specification (DOCSIS)-qualified infrastructure. The aim was to equal to or better the voice services provided by the public switched telephone network (PSTN). That goal has now been realized by the likes of Cablevision, Comcast, Time Warner Cable, and other MSOs as they begin to deliver high-quality VoIP telephony services on a large scale.

So what's next for cable VoIP (define - news - alert) and IP applications in general?

In short, the answer is multimedia and mobility. PacketCable's longer-term goal has always been to provide subscribers with high-quality, secure, and authenticated multimedia services through PacketCable 1.5, also known as the PacketCable Multimedia (PCMM) specifications. These specifications have occurred coincidently with the maturation of SIP, a distributed signaling protocol between intelligent endpoints that was defined by the Internet Engineering Task Force (IETF) as early as 1999. Although it has been a buzzword of sorts in the past few years, SIP's ubiquity, if not its legitimacy, was assured when Microsoft included it in its 2002 release of Windows XP.

Today, SIP appears not only in PCs but also in phones, PDAs, set-top boxes, and even soft clients on PCs and PDAs that enable telephony, gaming, and other multimedia applications and services between and among SIP endpoints. As a distributed, application-layer signaling protocol, SIP is quickly finding its way into all sorts of communication devices, more and more of which will be connected to DOCSIS networks. This suggests the need for SIP to coexist with the NCS protocol that provides call control and routing within cable networks and through gateways outside to the PSTN.

This coexistence is inevitable. After all, despite SIP's continuing evolution and its many "flavors" today, nearly all endpoint devices currently in development have SIP (define - news - alert) in their roadmaps, except those POTS-

By Mike Clement

only phones designed for lowest-cost production and pricing. In other words, SIP is coming and MSO networks need to be ready for it. On the one hand, this coexistence may require additional engineering at the network core, especially in the call-management server (CMS), to enable better SIP–NCS interworking and prevent any undermining of QoS, reliability, security, and authentication that DOCSIS and PacketCable have defined.

On the other hand, enabling if not optimizing SIP interworking and compatibility across DOCSIS networks coupled with PCMM and PacketCable 2.0 implementations will help spawn a new generation of feature-rich applications that MSOs can use to generate incremental revenues, profits, and competitive differentiation. For that matter, SIP can also help MSOs bridge the mobility gap in order to better compete with telcos, which now possess a competitive advantage through wireless offerings.

## Network Versus Endpoint Control

In the early days of PacketCable, operators defined a common architectural approach for their networks. They faced the same debate telco architects had argued for decades: Should intelligence (e.g., session control and call states) reside in the network or be distributed in "smart" endpoint devices? At the time, SIP had not yet been defined, but its concepts favored smart endpoints. The problem was that, in the view of those favoring an NCS approach, a distributed model would give operators less control over network traffic and routing. At the same time, they were concerned that calls could be made and services stolen without their knowledge.

The NCS approach won, of course, with the PacketCable specification ensuring QoS, security, and authentication over hybrid fiber/coaxial (HFC) networks. And similar to what SS7 signaling did to enhance POTS by enabling custom local-area signaling service (CLASS) features like caller ID, call return, and call blocking, NCS likewise provided not all CLASS features but those that have continually proven themselves most popular. It is important to note that, aside from these technical concerns, there were economic ones as well. NCS, for example, enabled MSOs to debut VoIP services much faster and more cost-effectively over MTAs/eMTAs

**Voice telephony alone hardly expresses MSOs' ultimate potential for new services.**

compared to the wait and cost of developing and deploying smart endpoints. Analog POTS phones may not have multimedia bells and whistles, but they are everywhere, and they are cheap and cost-effective endpoints for PacketCable networks.

## NCS Versus SIP: The Best Of Both Worlds?

While NCS and SIP each has its strengths, together they are highly complementary, made more so by PCMM and PacketCable 2.0 specifications (Table 1).

As protocols go, SIP is among the most straightforward. It is text-based, similar to the World Wide Web's hypertext transfer protocol (HTTP), and the signaling messages generated by its call-processing language (CPL) are akin to the hypertext markup language (HTML) used to create web pages. Structurally, session set-ups, connections, and terminations by endpoints are as straightforward as well, using syntax reflective of interpersonal communications among humans. ("Hi, can you chat? Yes? Great. Shall we talk, use pictures, or movies, or all three? All three? Terrific, let's go. OK, all done. Goodbye.")

Currently Vonage (news - alert) and AT&T's (quote - news - alert) Call Advantage VoIP service offerings are good examples of a SIP-based approach to architecture. Both services use SIP to provide telephony services that ride over any public IP broadband network. The downside to these non-facilities-based SIP services, however, is that they are inherently "best-effort," which means QoS, bandwidth, reliability, and security cannot be assured. The QoS mechanisms that manage jitter, delay, and packet loss are simply not available to the overlay SIP service provider.

**Table 1:** A Comparison Of NCS And SIP

| Issue | NCS | SIP |
|---|---|---|
| QoS | Network centric via DQoS | Distributed, best-effort, unless enabled by PacketCable |
| Security | Secure under PC 1.x | Susceptible to hacking, spoofing, and DOS attacks |
| Endpoint technology requirements | Requires MTAs for phones (with CM for PCs or eMTAs) | No MTAs required; endpoints can be SIP soft clients, phones, or MTAs/IADs |
| Regulatory | PC 1.x covers all requirements for telephony | SIP has no provisions for regulatory compliance |
| Location-presence | Can associate network device awareness with a specific subscriber location via database lookups | SIP endpoints may "plug-and-play" anywhere in a network, known only by their IP addresses which are not necessarily location specific but could be mapped to a database |
| Multimedia applications | PCMM defines a robust multimedia framework over NCS cable plant, but lacks SIP's flexibility | Highly flexible and relatively simple syntax enables fast application development and easy mixing of media for potentially richer, more valuable, and compelling applications |

## Unleashing An Exciting Future

Although NCS and PacketCable 1.0 helped operators enter the VoIP business quickly and cost-effectively, voice telephony alone hardly expresses MSOs' ultimate potential for new services. That is especially true given cable's two-way digital plant and the amount of available programming and content poised for interactive, multimedia/modal applications and services.

The PCMM specification in PacketCable 1.5 and coming specifications of PacketCable 2.0 promise enormous strides in this direction. By adding SIP interworking, the potential for taking the subscriber experience to new levels becomes much greater because SIP enhances the ability of developers to create new applications and services that leverage such IP technologies as instant messaging, presence, IPTV, and video telephony.

Combining these and other technologies — including URLs, Java applets, XML calls, and other objects and Internet protocols — can result in an almost infinite variety of applications and services, bounded only by the imagination. It is possible to envision a future in which services are created dynamically, adapting to inferences drawn from users' behavior patterns, with the "killer" app being a unique, temporal combination of applications and services that meets a single user's need when, where, and however that need appears. These possibilities may include wireless applications too, bridging the mobility gap with fixed-mobile convergence via PacketCable 2.0 or an IMS architecture as specified by the



Go To Table of Contents | Go To Ad Index

The 2006 INTERNET TELEPHONY® Buyers' Guide

# Enhance Your Listing in the ULTIMATE BUYERS' GUIDE SINCE 1998!


## REACH QUALIFIED BUYERS 365 DAYS A YEAR!

### YOUR ENHANCED LISTING INCLUDES:

**Highlighted in Yellow:**

- Your Company Name (Bold)
- Address
- Phone Number & URL
- E-mail Address
- Enhanced Online Version
- Brief Company Description
- 100 Words Max
- Company Logo

(please submit vector-based .eps)

V
Viper Networks
Your Internet Phone Company™

**Viper Networks (A)**
7660 Fay Avenue, Suite H-339
La Jolla, CA 92037
888-205-6017
www.vipernetworks.com
sales@vipernetworks.com
Products/Services: 93,94,96
Other: USB To RJ-11 Adapter
Viper Networks is a publicly traded (OTC) VoIP supplier of hardware, software, and termination services worldwide. Viper Networks delivers the promise of cost effective crystal clear Internet Telephony via handsets and PDAs.

**Basic FREE Listing**

**VocalData, Inc (C)**
972-354-2100
www.vocaldata.com
Products/Services:
59,63,71,82,36

**VocalTec Communications (A,D)**
201-228-7000
www.vocaltec.com
Products/Services:
84,91,94,104,44

**Voda One (E)**
402-334-4204
www.vodaone.com
Products/Services:
96,95,108,130,80

**Vodavi Communications Systems (A)**
480-443-6441
www.vodavi.com
Products/Services:
94,95,31,36

**Vodawave (A,B)**
213-385-7005 x202
www.vodawave.com
Products/Services:
94,115,29,45,46

**Voice2Voice Communication (A,E)**
800-828-8048
voice2voice.biz
Products/Services:
61,65,94,36,46

**Voiceboard Corporation (A**
805-985-6200 x1245
www.voiceboard.com
Products/Services:
2,6,97,105,78

**VoiceClips (A,E)**
905-564-8575 x205
Products/Services: 65,67,70
Other: VoIP Recording and Quality Measurement

**Contact Your INTERNET TELEPHONY Sales Manager at 203.852.6800 Ext. 139**

**Don't Miss this Once-A-Year Opportunity!**
Enhanced Listing Deadline: October 31
Your Cost: $750

***NEW ADVERTISER SPECIAL OFFER:***
Purchase an Enhanced Listing and receive 10% off any Advertisement in December's Buyers' Guide Issue*
*New Advertisers Only

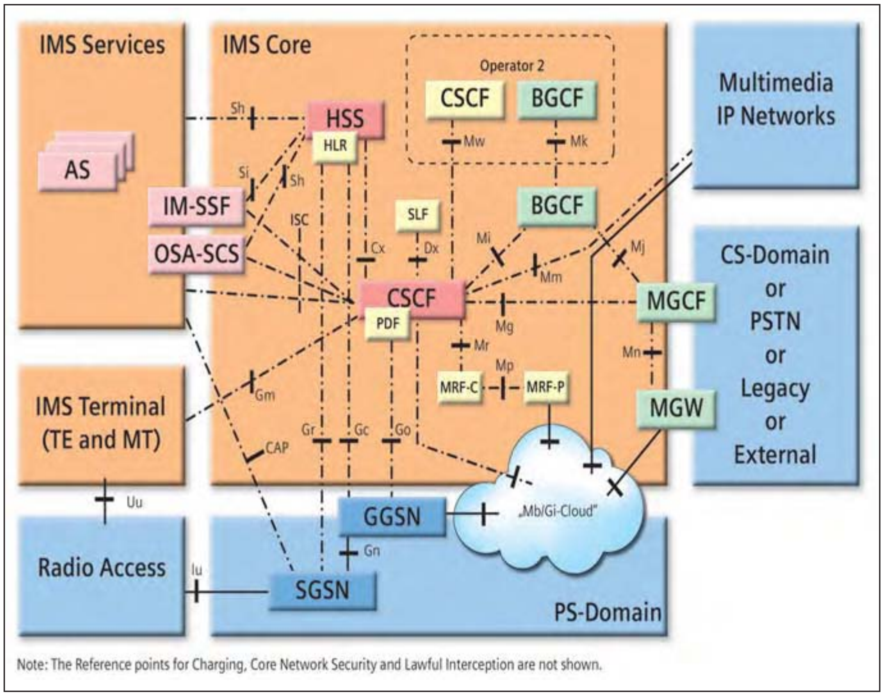
3GPP working group and being adopted by service providers worldwide (Figure 1).

The interworking of NCS and SIP is inevitable and available now. Fixed-mobile convergence — along with the convergence of public and private networks — can then enable these capabilities to interwork seamlessly over unified domains supporting SIP-based devices that help to unify the end user's experience. The result? Anytime, anywhere information access, entertainment, and communications, together the Holy Grail of the digerati for years now. IT

*Mike Clement is director, NGN Networks, Siemens Communications, Inc. (news - alert) For more information please visit the company online at networks.siemens.com.*

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

**Figure 1:** 3GPP IMS R5 Architecture Network Entities and Reference Points



# VoIP Is Here — What Are You Waiting For?

Moving toward a converged, reliable digital system opens up a world of opportunities for the enterprise and creates a truly global office for users. While challenges still exist in the adoption of VoIP, the benefits of this technology over traditional voice networks are many — from increased productivity and operational flexibility, to greater cost reductions and investment protection. The first steps in accepting VoIP systems as tomorrow's enterprise communications infrastructure are understanding exactly what VoIP is, what your options are, how it can benefit the enterprise and its end-users, how to choose the right vendor, and where it's going from here. There is no question that VoIP is the next great revolution in voice communications. In fact, industry analysts believe that by 2008, corporate IP telephony will reach $5.5 billion worldwide. So what is everyone waiting for? Let's get on with it…

## The Groundwork Has Been Laid

While it is certainly in its early stages of growth, much of the foundation from which VoIP (define - news - alert) will take off, in terms of its infrastructure, and its growing recognition as a business necessity, is in place. In fact, today's traditional circuit-switched telephony already employs some of the same network elements as pure VoIP systems. What's more, traditional Public Switched Telephone Network (PSTN) carriers are acutely aware of the tremendous financial and technical efficiency of transmitting data as digital packets. And, in the enterprise, as businesses are faced with increasing competition and a simultaneous need to reduce costs, the efficiencies and risk management capacity that VoIP can offer are truly compelling.

With that said, there is great potential in the marketplace for an open-standard VoIP solution — a solution that can transform an enterprise's communications infrastructure and that can scale to meet its telecom needs going forward.

## VoIP 101 — The Options

First, it's important to realize that Voice over IP comprises much more than just "voice." Today, the technology supports a comprehensive integration of voice, e-mail, fax, instant messaging, video and more, significantly increasing its value over Private Branch Exchange (PBX) infrastructures. These old-fashioned PBX configurations rely on circuit switching technology, meaning that every time a telephone call is made, a virtual circuit must be opened between two endpoints. VoIP, on the other hand, transmits the voice as digital packets, just as e-mail and Web browsing is transmitted. While, in the past, this method of transmission was a sticking point with potential users, today's more robust and proven technologies can effectively route and prioritize packets — delivering them in near real-time.

So what options exist for an enterprise considering deploying a VoIP solution? Here are three approaches:

***A Hybrid Approach:*** The Hybrid PBX system, in which IP-enabled PBXs combine IP technology with traditional Time Division Multiplexing (TDM) technology, essentially connects an existing traditional infrastructure with IP-based telephony. This option is effective for the short-term, enabling users to leverage existing investments in PBX equipment.

***A Pure Approach:*** Instead of using

By Ron Lott

legacy equipment, such as PBX and trunklines, a Pure IP-PBX system routes all calls over the LAN/WAN, converting all telephony to data transmitted over the corporate IP backbone or Internet, only connecting to the PSTN when IP routing is not available.

*A Hosted Approach:* An IP Centrex or similar, hosted IP telephony system does not require companies to invest in any sort of PBX or other equipment to obtain the benefits of VoIP. The equipment providing the call control and service logic functions is owned and operated by the service provider and maintained on the provider's premises. IP Centrex services are interoperable with both traditional analog equipment and IP-based equipment, making them immediately available to the enterprise without any change to existing infrastructures. This option provides a transitional step toward an in-house VoIP system.

## Greater Operational Efficiencies

VoIP technology, by its very nature, is more flexible and extensible than traditional voice transmission technologies due to its distributed and disaggregated architecture. This design, in which application layers are separated into different components that can be integrated or substituted as needed in the overall system, significantly improves management of the system (enabling remote and centralized management) and allows it to be more dynamic, adaptable and customizable overall. For instance, a VoIP system finally makes it easy to accommodate adds, moves and changes in telephone extensions (the bane of many a business's existence), which in turn greatly reduces a corporation's costs in terms of skill sets, training and personnel.

Adding to VoIP's adaptability is a core foundational technology called SIP (Session Initiation Protocol). SIP is the accepted standard for all major communications equipment manufacturers, and many software companies. Based on the notion that over time things such as client devices, services and underlying infrastructures will change, SIP separates the signaling from the hardware and media. As such, SIP allows voice calls, video conversations, etc. to occur between infrastructures that use different communications platforms, telephony clients, and even networking infrastructure. This adaptability to change is a key selling point for businesses that

**There is great potential in the marketplace for an open-standard VoIP solution.**

want to protect their investments.

### Increased Productivity

Because VoIP combines voice with numerous value-added features such as e-mail, conferencing, fax, and mobile communications, VoIP significantly enhances productivity by centralizing and simplifying all communications. Additionally, the distributed architecture of VoIP technology requires less real estate and overhead, and allows a business' personnel to work anywhere, anytime with the same user interfaces and features.

### Greater Cost Reductions: Lower TCO And Higher ROI

There is no question that VoIP delivers greater cost effectiveness and better utilization of IT dollars than traditional voice networks. By consolidating multiple communications technologies — voice, data, and video — into one system and by consolidating multiple applications — such as e-mail or IM — VoIP immediately reduces overhead costs, eliminating infrastructure and maintenance redundancies.

Additionally, with the toll bypass, communication costs are significantly reduced by the simple fact that calls over the Internet or corporate backbone do not incur a surcharge beyond what the company pays for it's corporate data network. Imagine the costs savings for a business with branch offices, as calls from one branch to another can be made as if both offices were located in the same location.

### The Perceived Challenges

From its very beginnings, VoIP has faced the issue of quality of service (QoS), often being labeled as not yet "100 percent carrier grade," as compared to the PSTN (define - news - alert). This concern is being eliminated based on the growing reliability of IP telephony systems, and the fact that many of today's VoIP systems provide the capability of routing calls over traditional TDM networks, should the IP telephony network fail.

Closely tied to issues such as interoperability with data networks and QoS, another challenge for VoIP is security — from toll fraud to eavesdropping to DDoS/Viruses/Worms. This challenge can be addressed by enterprise IT managers placing particular importance on the security of the underlying infrastructure they choose as the foundation for their VoIP networks, as well as their company's adherence to enterprise security policies, processes and procedures.

Yet another challenge for the migration to IP telephony is the question of expense. After reviewing the advantages, it's clear that the expense of migrating to VoIP is low compared to the savings it can bring. Collapsing voice services into existing data network helps reduce capital outlays and operational expenses associated with the administration and maintenance of traditional circuit-switched voice equipment and wiring. Plus, there is toll bypass as yet another reduction in business expenses, not to mention the increased productivity and workflow efficiencies that ultimately help improve a business' bottom line.

What about 911? Emergency 911 calls are yet another hurdle VoIP adoption faces. Not all VoIP providers provide 911 access, and most don't offer the phone number and address notification that TDM networks provide. While full 911 functionality remains an opportunity for market differentiation among VoIP providers, many industry experts agree that progress is being made on this front and will be within reach in the coming years.

### Who Ya Gonna Call?

If the advantages of VoIP are too compelling to ignore, it's important that you know what to look for when selecting a potential VoIP vendor(s). Here are some guidelines that will steer you in the right direction:

**Look for a vendor:**

- with extensive experience in the telecom industry, including experience building systems for both traditional and VoIP telephony alike.
- that supports open, integrated, IP-based environments (hardware- agnostic, binary-compatible, open-source) as opposed to one that favors proprietary, vendor-centric, end-to-end solutions. This is critical for investment protection and future upgrading.
- with a range of best-of-breed partners that can provide access to key functionality, applications and new services (such as wireless VoIP).
- that can enrich an enterprise's legacy investment by offering solutions that connect existing technology to the power of VoIP.
- that builds its systems and solutions with security in mind, from the inside out. Are the systems based on a highly scalable and reliable OS? Are they built using a secure application programming language and protocol?

### The Future Of VoIP

So where is VoIP headed? Straight to the top of companies' technology priorities. As adoption increases, so too will the functionality that VoIP can provide — from fixed/mobile convergence, in which cellular devices are free to roam from GSM to VoIP within a home or enterprise, to IP conferencing, to other enhanced services, such as gaming and other entertainment applications. There is no time like the present to take advantage of this burgeoning technology. Consider the advantages, and then consider your options and your potential vendors. You've got nothing to lose and everything to gain. IT

*Ron Lott is in VoIP Market & Industry Development for Sun Microsystems. For more information, please visit the company online at http://www.sun.com.*

By Joseph Curcio

# Securing Voice In An IP World

On May 9, 2005, *The New York Times* referred to an alarming incident that occurred last year in which an intruder breached a major network and seized programming instructions for many of the computers that control the flow of the Internet, including those serving the U.S. military, NASA, and research laboratories. Said the *Times*: "...the case illustrates the ease with which Internet-connected computers — even those of sophisticated corporate and government networks — can be penetrated, and also the difficulty in tracing those responsible."

For those of us who stay up at night thinking of how to protect organizations from security breaches, this case provides a reality check about what it will take to harden our defenses against increasingly destructive "zero day" exploits — or previously unknown breaches — in which intruders find new ways to break through an organization's defenses and do serious damage. A successful attack could cripple an enterprise, interrupt business continuity, and result in lost revenue.

The convergence of voice and data networks complicates the issue. While it fosters a more intelligent communications environment — where workers, processes, and customers are connected to the right people at the right time — convergence also requires skillful management of a new generation of cyber threats. Until now, most attacks have targeted data networks, but as voice applications become even more strategic on IP networks, they too can be exposed to many of the same vulnerabilities that plague data networks, including denial of service (DoS), application layer attacks, spoofing, trust exploitation, and so on. In fact, in another example of today's vulnerabilities, on July 14 *The Wall Street Journal* reported that critical flaws in one vendor's IP telephony software could allow hackers to gain control and shut down voice systems, redirect phone calls, eavesdrop, or gain access to other computers running that vendor's telephony.

But there's good news — and no need to panic. For one thing, all IP telephony platforms in the market do not share the same architecture. Vulnerabilities in one are likely to be absent in another. Further, multiple lines of defense can protect businesses that launch into the new era of intelligent communications. Most communications vendors provide solid security solutions now and are developing many others. But in order to succeed against the hackers, thieves and phishers that threaten us, businesses must first recognize that security in a converged world does not have a simple, one-step, one-layer, one-vendor solution. The industry itself must also come together to educate customers about potential dangers and to develop new security solutions to address emerging threats.





Go To Table of Contents | Go To Ad Index

The first step is to know what can be done. Here are some key principles for defending converged networks, ensuring that voice applications remain secure and protecting an enterprise's global communications.

## Protect Communications Flow At Every Level Of A Multivendor Network

With cyber attacks becoming more sophisticated, organizations that rely on securing the network infrastructure alone will find themselves defenseless if an intruder penetrates that first level. Enterprises must also provide strong security at the application and unified access layers. Every vulnerable point and application within the organization needs to be able to defend itself from attack.

Take denial of service for example. Providing protection only at the network infrastructure layer (i.e., the router) can pose a major problem if an intruder gets into the network — and most attacks happen within the network infrastructure. Remember the old adage: You're only as strong as your weakest link. Ignoring DoS protection within the server at gateways and endpoints can make a business less secure.

The best strategy is to provide layers of protection all the way from the network infrastructure through the application level to the user device. A practical fact of life is that most networks are multivendor — through acquisitions, mergers, and reuse of legacy systems and the best technology combinations. So, multilayer protection must work well even in a network created by multiple vendors.

While it is imperative to repair and harden the problems of network infrastructure, just relying on the infrastructure alone is playing with fate. New threats will evolve, new attackers will emerge. Make it harder for attackers by increasing the number and types of hurdles they have to pass through up and down the communications stack.

## Use Open Rather Than Proprietary Solutions

The major benefit of open standards is that vendors and users alike are able to build on what's already been accomplished to solve security problems, so the level of protection is constantly

**An open standards approach is vital to securing business communications.**

improving. The protection is also more likely to work in multivendor environments and continue to protect the network as new elements are added, as long as they, too, adhere to industry standards.

In a proprietary implementation, businesses rely on a vendor's claims of the security being provided, which could put a company at risk if those claims have not been tested and certified through rigorous industry peer review. In an open environment, vendors are challenged to meet specific government certification criteria, and must be tested to prove they do what they say they do. Open standards let customers choose the best, most cost-effective security solutions that make the most sense for their particular business. Those who are locked into one vendor's proprietary security system year after year could find themselves in a precarious situation through lack of interoperability and an inability to keep up with the best available solutions.

Because a total security solution requires protection at the unified access and business application levels, as well as the network infrastructure level, an open standards approach is vital to securing business communications. Every vulnerable point and application within the organization needs to be able to defend itself from attacks, and an open standards-based strategy for multivendor networks works best.

## Trust Vendors That Take A Holistic Approach To Security And Collaborate With The Industry

Security-conscious vendors take a multifaceted approach to delivering secure products and solutions, not just by addressing all entry points along the stack, but also all milestones in the lifecycle of an installation. Be sure to ask vendors about their approach and evaluate how they deliver secure products solutions.

Media encryption is one example: It is not just offered on high-end elements, those requiring third-party adjuncts, or for just point-to-point calls. Why secure an executive telephone only, if every telephone on a floor is also a point of system vulnerability? There should be a clear understanding of the vendor's policy for addressing new vulnerabilities, such as a rapid response team that initiates assessment, communicates whether the customer is at risk and, if necessary, takes steps to mitigate that risk.

Some vendors also offer key services that can enhance the security of converged networks. A knowledgeable vendor can help in the initial architectural planning of a secure converged network, paying special attention to the availability of mission-critical IP telephony applications. In implementing a solution, make sure the chosen vendor has the expertise to not only execute, but also service security processes. Once a network is in, experts must help monitor it securely, so choose a vendor that offers a level of support commensurate with business needs. A vendor should be able to remotely maintain a network securely, and provide a total managed service that includes monitoring, root cause analysis, and remediation for security functions across the converged infrastructure.

However complete any one vendor's approach might be, no single vendor is likely to be able to protect customers from all future threats. Meeting this challenge requires the industry to pull together and share information to help educate customers and, as new threats and solutions are developed, provide defense mechanisms that can be rapidly tailored to the new challenges. Why reinvent the wheel if people have already done productive work and have done it well?

It is important for today's leading vendors to work with groups like the VoIP (define - news - alert) Security Alliance (VOIPSA), a premier alliance of the VoIP and security communities, focused on increasing vendor and customer awareness of threats to IP communications. Another industry group, Trusted Computing Group (TCG), develops and promotes open specifications for use in products that check the integrity of a computing platform, and protect it against software-based threats. TCG provides an architecture that helps enterprises validate clients before admitting them into a network, raising defenses to another level.

As an industry, we are not yet at the point where every business network and the applications on it are completely safe from malicious attacks. The network penetrations already reported demonstrate that these threats are still very real. But steps can be taken to reinforce and protect business communications against bedeviling attacks. Holistic security must transcend all layers of the enterprise, and encompass network infrastructure, communications and business applications, as well as end points. Open, standards-based, multivendor solutions and services, combined with industry best practices, are the path to greater integrity in an increasingly hostile communications environment. Once these essential principles are incorporated throughout a communications framework, a business will have a secure foundation from which to reap the extraordinary benefits offered by the new age of intelligent communications. **IT**

*Joseph Curcio is vice president of security strategy and development at Avaya. (quote - news - alert) For more information, please visit the company's Web site at http://www.avaya.com.*

# Subscriber-Centric Policy Management:

## Laying The Groundwork For IMS

### *To Converge IP Access, Providers Must Converge Rules to Manage Subscribers, Applications and Network Resources*

Service providers worldwide are preparing to implement fixed/wireless convergence architectures based on the IP Multimedia subsystem (IMS) standard originally developed by the 3GPP standard group. IMS aims to create an IP services layer that operates independently of underlying networks, thus yielding seamless subscriber access to IP multimedia applications from any device over any network.

In the IMS vision, easier usage begets more usage. The broadband subscriber on the road will be able to open their cell phone or laptop PC browser and immediately link to IP services via a 3G network, hotel WiFi network or corporate or home wireless LAN. Afforded this maximum flexibility and mobility, the subscriber will call on IP services for information, entertainment, and productivity more often.

The service provider implementing IMS will be transforming its infrastructure to an all-IP network. This convergence of networks will produce substantial savings over the costs of operating multiple different networks for different services. Additionally, the shift to all-IP services positions the service provider to tailor application experiences on a subscriber by subscriber — or subscriber-centric — basis. For example, a bandwidth-intensive video-telephony data session allocated more resources should yield commensurately more revenue than a standard VoIP (define - news - alert) call.

This subscriber-centric vision can be accomplished only through implementation of subscriber-centric policy management that empowers service providers to tune the allocation of network resources to the subscriber's willingness to pay. Put another way, per-subscriber, multi-network access requires a mechanism to right-size both resource allocation and pricing of each service based on the resources each subscriber consumes.

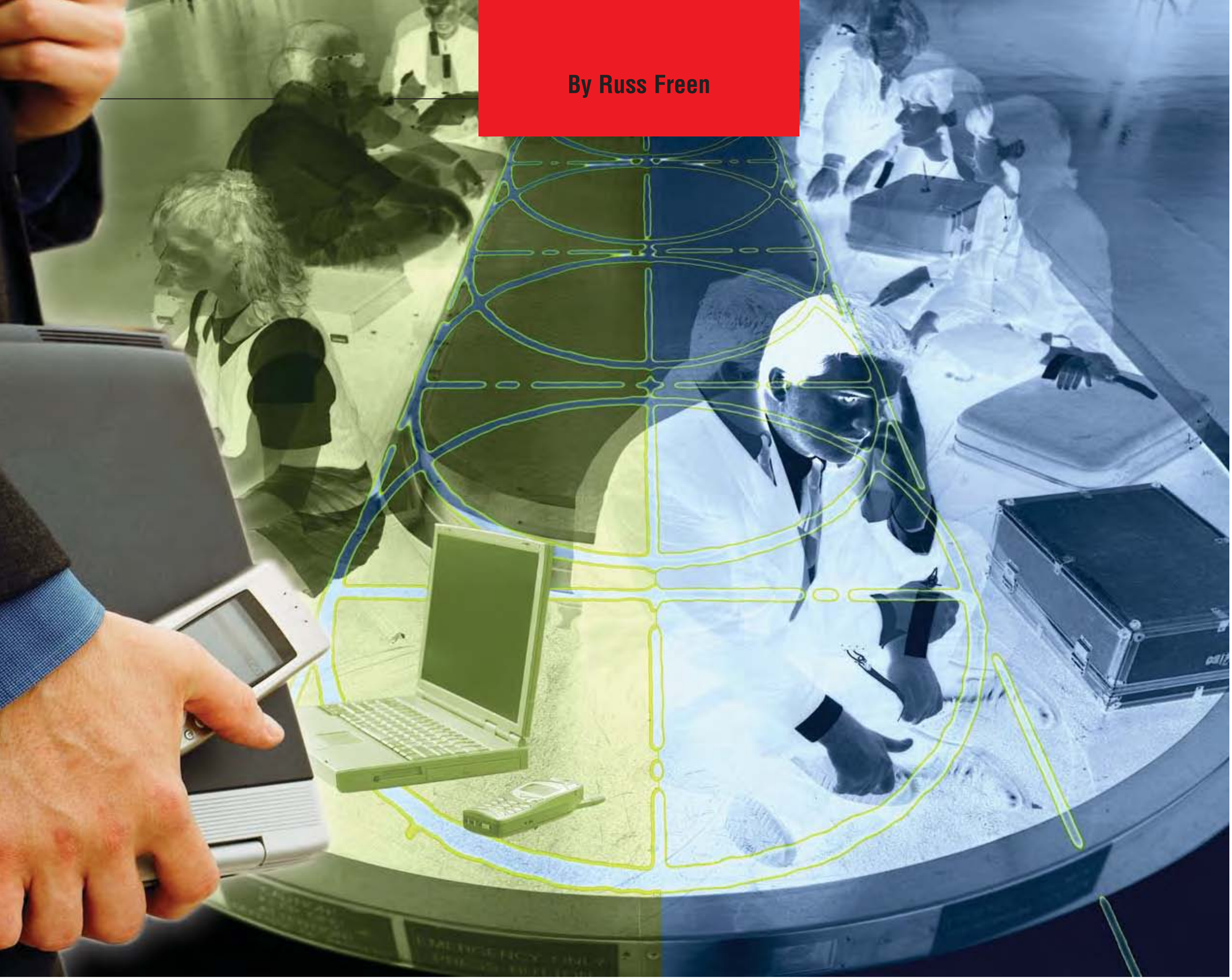
Only through correlating each subscriber to the services they use can service providers monetize the true value of their networks. In the bargain, subscriber-centric policy management will prove crucial to creating innovative business models that make tiers and bundles of basic and premium IP services irresistible to subscribers, as well as easily managed for service providers.

### Laying The Groundwork

From the subscriber's point of view, IMS and fixed/mobile convergence will provide a single point of entry to IP services, regardless of the device or the fixed or mobile access network used.

Subscriber-centric policy management

By Russ Freen



paves the way for this network independence by providing a repository of subscriber profiles, application requirements and rules for bringing subscribers and applications together. The service provider controls the definition of these rules, or policies, and the policy management system enforces those rules on a per-subscriber basis.

The IMS architecture directly addresses policy through what it describes as the Policy Decision Function (PDF) and Home Subscriber Server (HSS). These components of the architecture include centralized subscriber and application profiles and policies. The HSS leverages the profiles to enforce policy in the network through a logical chain of processes. As the subscriber registers with the network, the HSS consults the policy database to learn the roaming permissions, account status time-of-day permissions and based on these provides the subscriber with the ability to access the IMS framework. The subscriber then accesses an application, which is authorized by the HSS at the point of request. During the authorization process the HSS consults the policy database once again to learn the application-specific permissions and other policies to be applied to this subscriber. The HSS then authorizes access, directing the call processing elements to allow a session to commence.

The DSL, cable or 3G service provider can implement subscriber-centric policy management for a single network today. Doing so will yield immediate benefits, including the powers to correlate subscribers with usage, to enforce fair usage of bandwidth and to leverage granular usage intelligence to develop new services and service bundles.

Implementing subscriber-centric policy in the near term also positions the provider to extend these benefits of subscriber control across both fixed and wireless networks, laying the groundwork for offering device- and network-independent access to applications in the future. The same subscriber profile, application profile and policies applied to a cable network today, for example, could be applied to WiFi, WiMAX or mobile access in the future.

## Challenges

The IMS vision contemplates a radical departure from the relatively static operations and business processes of historical communications networks. IMS

**Enabling subscribers to tailor applications to their personal needs will prove crucial to generating loyalty.**

requires subscriber-centric policy management because making IP applications available over any network requires a focus on the subscriber, rather than on any given network.

Traditional telephone networks have been engineered for largely predictable peak calling periods. Such engineering practices cannot help but produce under-utilized networks, because much of the peak capacity will remain unused throughout the rest of the year. Additionally, capacity allocation in traditional phone and data access networks has been largely a fixed and network-centric affair: a DSL subscriber signs on and gets a fixed amount of bandwidth for the duration of his online activity. In effect, resource allocation policies are impersonal and network-centric: they are determined by network capacity for predictable events and effectively engineered into the network as a one-time-only task.

By contrast, traffic patterns and capacity demand in the IP multimedia realm cannot reliably be known in advance, either in aggregate or per subscriber. Nor can the provider rely on provisioning the exact same bandwidth to each subscriber or application every time, since the subscriber may enjoy various QoS permissions. For example a subscriber may be using a 300-Kb/s mobile data connection one moment, and the next moment use a 3-Mb/s cable modem connection. Resource sharing among thousands of unpredictable subscribers must be managed not just once, but dynamically and continually.

Hence, service providers must respond to a primary challenge presented by dynamically changing subscriber and application requirements. Automating resource allocation on a per-subscriber basis through policy definition addresses this challenge. Only then can the provider ensure service availability and QoS both in the aggregate and for each subscriber.

The multi-network nature of IMS presents another key challenge: scalability of operations. Implementing subscriber-centric policy management in a multi-network environment will involve managing millions of subscriber profiles, possibly hundreds of service and application profiles and supporting tens of thousands of simultaneous user authentication, service access and mediation transactions.

## The Benefits

Subscriber-centric policy management can provide a powerful edge to providers engaged in IMS competition. As triple play services proliferate, and as boundaries between fixed and mobile networks blur, the battle increasingly will put ownership of the subscriber relationship at stake. This trend already is well underway as wireless, cable and telephone companies all face the dual challenges of wooing each other's subscribers while defending against attrition among their own.

Most service providers now believe that enabling subscribers to tailor applications to their personal needs and desires will prove crucial to generating more loyalty and greater willingness to pay for value. This provides the surest path to retaining customers, growing market share and increasing average revenue per user (ARPU).

Subscriber-centric policy management benefits service providers in this quest by supplying them with a unified view of the subscriber and by supplying one set of converged service credentials and single sign-on. The converged authentication process also ensures service continuity across network-to-network handoffs via centralized subscriber tracking.

The subscriber-centric model further aids the provider by simplifying correlation of services to users. These correlations, in turn, enable dynamic, real-time control of personalized service access.

On the front end of this system, providers can leverage this correlation to simplify service definition and provisioning and to accelerate creation of complex services and service bundles. Providers can define service tiers based on anything that is easily measured and simply controlled, such as monthly consumption, bandwidth and QoS. In sophisticated systems, application-specific tiers are possible, such as tiers with and without VoIP features.

At the same time, per-subscriber policy enforcement can be applied to fair use of bandwidth. Subscribers who threaten the quality experience of other subscribers, and threaten the provider's revenue model, with peer-to-peer or other bandwidth hungry applications can be identified and redirected to a portal where they can be offered fair connection characteristics.

Some control also can be shared with the subscriber via a self-management Web portal. Providers can enable each subscriber to choose a unique lineup of applications, which also will reduce the provider's operational costs by effectively recruiting the customer to provision services.

These benefits all owe their genesis to the power to set policies and enforce them dynamically.

A fundamental shift is afoot in communications. In the any-network future, IP applications are becoming less tied to infrastructure and more closely tied to the subscriber. Subscriber-centric policy management is essential to every service provider determined to thrive through that shift. IT

*Russ Freen is the co-founder and chief technology officer for Bridgewater Systems, (news - alert) a developer of subscriber-centric policy management software for IP-based services. For more information, please visit http://www.bridgewatersystems.com.*

# VoIP MARKETPLACE

# VOIP MARKETPLACE

# CEO Spotlight

**Jeff Kirchner**
**CEO/President**
**Epygi Technologies**

**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 Jeff Kirchner, CEO/President of Epygi Technologies. (news - alert)**

**GG: What is Epygi's mission?**

**JK:** To deliver cost effective, innovative, and reliable communications solutions for the small business market. We focus on a segment of the market, which is underserved by the large manufacturers, yet is the fastest growing in most economies around the world and in exports.

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

**JK:** We make a combined IP-PBX (define - news - alert), router (VPN, firewall, IDS, etc.), and SIP-based VoIP gateway in one box. Our systems bridge the world of VoIP (define - news - alert) and traditional telephony by supporting both analog lines and phones, as well as IP phones and IP trunking in every unit and in the base price. All of the PBX features you would expect are included, as well as a "VoIP friendly" integrated router, firewall, and VPN, with a very reasonable price tag. We are delivering easy to use and reliable advanced communication to the small business user for a very cost effective price.

**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?**

**JK:** I believe the channel to the customer remains a challenge. We need to educate the small business user of the benefits of converged solutions, whether it be cost savings or productivity enhancements. We also need to overcome any lingering fears of quality or reliability. All of these need to be communicated to and through our channel partners. Many of these partners are entering telephony for the first time (e.g., data resellers) or are expanding beyond traditional telephony (voice resellers) into the world of data and VoIP. The great news is that everyone, including the channel partners, recognizes the size of the opportunity. Now all it takes is old fashioned work and focus, focus, focus.

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

**JK:** We have developed and deployed all of the basic telephony and VoIP technologies. Increasingly, we spend much of our time in support of real world deployments — tuning protocols for interoperability and the like. We have conducted interoperability testing with over (20+) VoIP carriers worldwide and no two are exactly the same. It is the nature of the game as SIP is a standard, but the kinks need to be worked out in real life deployments. Thankfully, there are no big or significant issues, just old fashioned work and again, focus, focus, focus.

> The channel to the customer remains a challenge.

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

**JK:** Oddly, I think our job is to make ourselves disappear behind the benefits and solutions for the customers. In the end, it will be anytime, anywhere communication with intelligence behind it. Does anyone care that VoIP was involved in intelligently routing that "must have" call to them during a meeting, while sending less important calls to their voicemail? Does anyone care that VoIP was used to reduce the cost of overseas calls by 50 percent? IP telephony and VoIP, as a term, should fade into the background as the benefits and new applications take center stage… applications, which just happen to ride on VoIP or IP telephony. IT

# CEO Spotlight

**Charles E. Hoffman**
**President and Chief Executive Officer**
**Covad**

**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 Charles E. Hoffman, President and Chief Executive Officer at Covad.**

**GG: What is Covad's mission?**

**CH:** We aim to be the most innovative provider of broadband communications with our unparalleled assets: a nationwide network, superior customer experience, efficient operating support systems, and services that enhance the way people live and work.

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

**CH:** With America's largest nationwide DSL broadband footprint, Covad (news - alert) is uniquely able to provide broadband services to consumers and small-and medium-size businesses, either directly or through our array of wholesale partners like AOL and EarthLink. Covad offers an extensive portfolio of Internet access services for businesses, including data-only T1 services, four business-class symmetrical DSL (SDSL) services and dedicated-loop services.

Since becoming the first company to offer DSL in the U.S., Covad has continued to innovate, introducing new and better communications services to the market. We are a market leader with Covad VoIP, a business-class broadband service that truly integrates telephone service and high-speed Internet access. Covad also offers Voice-Optimized Access (VOA) that allows us and partners to provide VoIP (define - news - alert) service on a fully managed line to ensuring call quality. And later this year, in partnership with EarthLink, Covad will offer Line-Powered Voice Access utilizing next-generation ADSL2+ services that will allow Covad to offer multi-megabit broadband, voice and ultimately video services.

**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?**

**CH:** Education is the main obstacle. With so many different VoIP packages on the market, it is difficult for customers to identify the service that fits their specific user habits. Once the lines between true business-class and residential VoIP are further clarified to the customer, the industry will see an increase in adoption and less disappointment by those who misunderstand the technology.

Quality and reliability, of course, are key. We have made and will continue to make the necessary investments to ensure an excellent customer experience.

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

**CH:** Wireless last mile business broadband service and line-power voice access (LPVA) are two disruptive technologies in which we are among the market leaders. Our fixed wireless broadband technical trials have demonstrated the feasibility of using this technology to further expand our existing network reach and offer new services, like VoIP over WiMAX (define - news - alert).

Covad LPVA is a new hybrid phone service that combines the simplicity and stability of the traditional wired telephone with the flexibility and pricing of VoIP. LPVA is a true ILEC-replacement residential voice service; it is created with the ability to work on your existing inside telephone wiring and stays operational in the event of a power outage. We have a commercial launch of this service scheduled in the fourth quarter of this year with EarthLink.

**GG: How will the acquisition of NextWeb affect Covad's future?**

**CH:** Our acquisition of NextWeb accelerates Covad's entry into the wireless broadband market by up to two years versus building it out ourselves, and will provide the cornerstone of our wireless efforts. This is one of our key strategic imperatives, and is another step in taking control of our destiny. The NextWeb team has a proven track record of profitable growth, and we expect the economics of this business to improve with Covad's extensive network assets and expertise and distribution channels. With wireless last-mile connectivity, Covad will be able to reduce its direct recurring costs per customer for an equivalent wireline T1 by up to 60 percent, deliver an improved customer experience as a result of being able to control the service end-to-end, reduce dependence on copper and the regulatory landscape, and add high-bandwidth services targeted at the small- and medium-sized business market.

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

**CH:** IP telephony is the future — it's only really now beginning. We have seen significant growth throughout 2005 and anticipate that trend to continue. In only a few short years, we're going to be seeing a vast array of new IP applications, ranging from video, to VoIP over fixed wireless, and other applications people are just now cooking up. Businesses especially will benefit from these technological advancements, making corporations even more efficient and productive than today. IT

# ADVERTISING INDEX


Feeling Deserted By Your Media Gateway Vendor?
Does your media gateway vendor make you feel like you're alone on a deserted island, with disparate protocols and no lifeline to customer service and support? Excel Switching, an award-winning provider of carrier-class telecom equipment for nearly two decades, heard your SOS for multi-protocol support, scalability and value and developed the IMG 1010 integrated media gateway.
Excel's IMG 1010 Integrated Media Gateway Solution:
• Scalable VoIP Gateway
• Integrated SS7
• IP-IP Transcoding
• Routing/Translation
• Media Server
Benefits:
• Reduced CapEx and OpEx
• Faster Time to Market
• Improved Network Performance
• 15+ Years Telco Experience
• Worldwide Deployments and Support
• Investment Protection
EXCEL
Excel Switching Corporation
(508) 862-3000
For a complimentary copy of Excel's White Paper, "Enabling SS7 on an H.323-Based VoIP Network," visit
www.ExcelSwitching.com/Gateway