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The VoIP Authority Since 1998™

The Birth Of Just In Time Communications (page 6)

VoIP
Creating Value In Consumer VoIP

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- Elon Ganor Speaks on the Future of VoIP
- IMS and the Challenges of Convergence
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Group Publisher and Editor-In-Chief, **Rich Tehrani** (rtehrani@tmcnet.com)

EDITORIAL
Editorial Director, **Greg Galitzine** (ggalitzine@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**
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EXECUTIVE OFFICERS
Nadji Tehrani, *Chairman and CEO*
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Editorial Offices: 203-852-6800
Customer Service: For all customer service matters, call 203-852-6800.

ADVERTISING SALES
Sales Office Phone: 203-852-6800
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)

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Circulation Director, **Shirley Russo,** *ext. 157* (srusso@tmcnet.com)
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EXHIBIT SALES
Sales Office Phone: 203-852-6800
VP of Conferences and Online Media
Dave Rodriguez, *ext. 146,* (drodriguez@tmcnet.com)

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The VoIP Authority

By Greg Galitzine

VoIP: Running With The Big Dogs

For someone who does not consider himself a road warrior in the truest sense of the word, the last five weeks have made me feel like one of those people you read about in the in-flight magazines they distribute on airplanes. And while I am certainly not 'there' yet, I think I managed a slight glimpse of what Harry Chapin wrote about in "Cat's in the Cradle." Frightening.

Still, the past month has seen a number of major industry events, and it seems to me that the whole world is focused in on VoIP (define - news - alert) these days. Perhaps one of the most telling signs that our industry is maturing, is the fact that we are attracting come hither stares from some of the bigger names in consumer technology, who are clearly looking to score with VoIP. Hopefully their interest is more than just a passing fancy.

Microsoft (quote - news - alert) and AOL (quote - news - alert) both had big things to say in recent days. Jon Miller, Chairman and CEO of America Online, announced (well, sort of) that regarding VoIP, "…into that landscape, AOL has decided to wander." The company has plans to work with Level 3 for outsourced VoIP services and Sonus Networks on the equipment side.

Miller outlined AOL's plans to offer VoIP as more than simply a replacement for, or cheaper alternative to, traditional phone service. "We think (VoIP) can be much broader," Miller said. "There's a totally different opportunity by integrating presence. You can screen the call, take the call, or send an instant message — your buddy list becomes a dashboard."

That's exactly the kind of thinking our industry needs. The one constant thread in this magazine, since the day we launched our first issue, has been the idea that applications would drive this industry to the fore. Sure, low prices will get 'em in the door, but it's the services and solutions heretofore unavailable using POTS that will keep 'em and generate revenue.

Not to be left out in the cold, Microsoft Chairman and Chief Software Architect Bill Gates outlined his company's vision for integrated communications. According to Microsoft, the goal of integrated communications is to help information workers easily and effectively communicate with colleagues, customers and partners in real time. Microsoft will deliver on integrated communications by building presence awareness into all its software applications, integrating various modes of communication (e-mail, phone, instant messaging (IM), short message service (SMS), videoconferencing and Web conferencing) to allow seamless transition from one mode to another, and delivering intelligent software that can manage communications with the context of a person's availability and preferences.

Microsoft Office Communicator 2005 (formerly code-named Istanbul) will be the recommended client for Microsoft Office Live Communications Server 2005. Communicator 2005 is currently in development and is scheduled for release in the first half of 2005. Eager beavers can participate in beta testing by downloading the client from http://beta.microsoft.com.

-Greg Galitzine, ggalitzine@tmcnet.com

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Contents

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90

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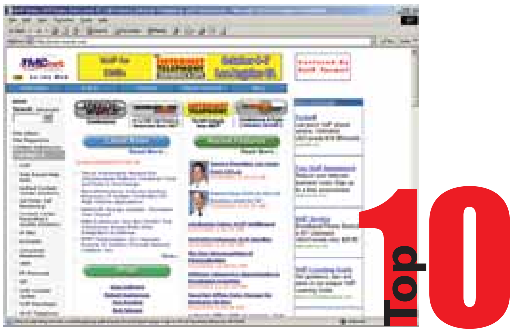

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3.	Canada	8.	Netherlands
4.	United Kingdom	9.	Japan
5.	India	10.	France

QUOTE OF THE MONTH:

"Nationally, the U.S. consumer pays almost 10 percent of his phone expenses on privacy features. To be a leader in consumer VoIP, carriers must cater to consumers' demand for privacy, and will be rewarded by their willingness to pay to protect it. In fact, VoIP adds incremental value by adding dynamic next-generation services that give consumers greater privacy and call control based on who is calling them, when they are calling, why they are calling, or all of the above."

– Stan Little

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 www.tmcnet.com for all the latest news and analysis. With over 3.9 million unique page visits per month, translating into nearly half a million unique visitors, TMCnet.com is where you need to be if you want to know what's happening in VoIP.

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

Walmart.com And Shop At Home TV To Sell 8x8 Services
VoIP has made it to the mass retail market thanks to a new partnership between Packet8 Walmart.com and Shop At Home TV.
http://tmcnet.com/100.1

Thoughts From Siemens On VoIP
One of the largest VoIP vendors shares some thoughts on the technology.
http://tmcnet.com/101.1

Cisco Passes VoIP Deployment Tests For U.S. Dept. Of Defense
Cisco announced it has passed additional U.S. DoD VoIP tests confirming the system meets interoperability, reliability, and resiliency requirements of DoD's multi-vendor voice network.
http://tmcnet.com/102.1

Vonage: Still Being Blocked?
Fresh off a regulatory victory to keep its business from being blocked on telecom lines, Vonage believes it's happening again.
http://tmcnet.com/103.1

Rick Boucher: Another Friend Of VoIP In D.C.
Virginia's Ninth District Congressman Rick Boucher is one of the most active and pro-high-tech congressmen on Capitol Hill.
http://tmcnet.com/104.1

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By Rich Tehrani

The Birth Of A New Industry?

With all the progress we've made in the last two decades with respect to communications, including telecom deregulation, the fax machine, cellular phones, e-mail, chat and WiFi (define - news -alert), communications is essentially more inefficient than it was at any time in the last ten years.

One may argue that the advent of the cellular phone was a boon for productivity and efficiency and e-mail further allows us to communicate more effectively. The reality is that the more devices and modes of communications we invent, the more inefficient we become. An inordinate amount of time is spent trying to find each other in this world where everyone is supposed to be connected 24x7.

In order to contact a person today it is necessary to call their office phone, cell phone, home phone, VoIP phone, send them an e-mail, an IM, call their secretary and so on and so forth until you connect. Assuming you can't find them, you have left a trail of possibly incomplete e-mails, IM messages, and voicemails in different mail boxes. How does the contact know which box to check first? Worse, how does a person know which voicemails are more important and which e-mails are crucial and do they really check these mailboxes in a timely manner?

An inordinate amount of time is spent checking mailboxes and messages. Some analysts estimate that over 60 percent of all voice communications consists of listening to or leaving voicemail! Murphy's Law tells us that when you check these boxes, other more important messages are in turn being banished to your cellular or office voicemail systems.

The problem gets worse when customer facing workers are not able to take phone calls in the same prioritized system a company values these contacts. In other words, a top customer should always get through to a top salesperson or service representative while it may be acceptable to sometime place lesser customers on hold for a few minutes during peak calling times. Top customers sending an e-mail should be responded to within minutes. In most companies, these customers aren't even identified, let alone dealt with in a prioritized manner. In many companies in fact, e-mails from customers still don't get answered!

When a boss calls an employee, the call should always be taken unless of course the employee is on the phone with an important customer (some might argue any customer.) Again there need to be rules that can be individualized to coincide with company priorities. Currently, if a boss calls an employee who is on the phone with anyone, regardless of whether it is work related, the call from the boss goes into voicemail. Of course this assumes no caller-ID and related technologies. If the employee isn't diligently checking voicemail they may not even know the boss needs to communicate.

The same issue arises when a spouse calls. At our last ITEXPO in Miami, most of the audience indicated they would take a call from a spouse every time they call... yes, even in meetings. I do most of the time as well. How many times have you been interrupted in an important meeting to answer "What do you want for dinner?" or "Can you stop by the dry cleaners," etc.?

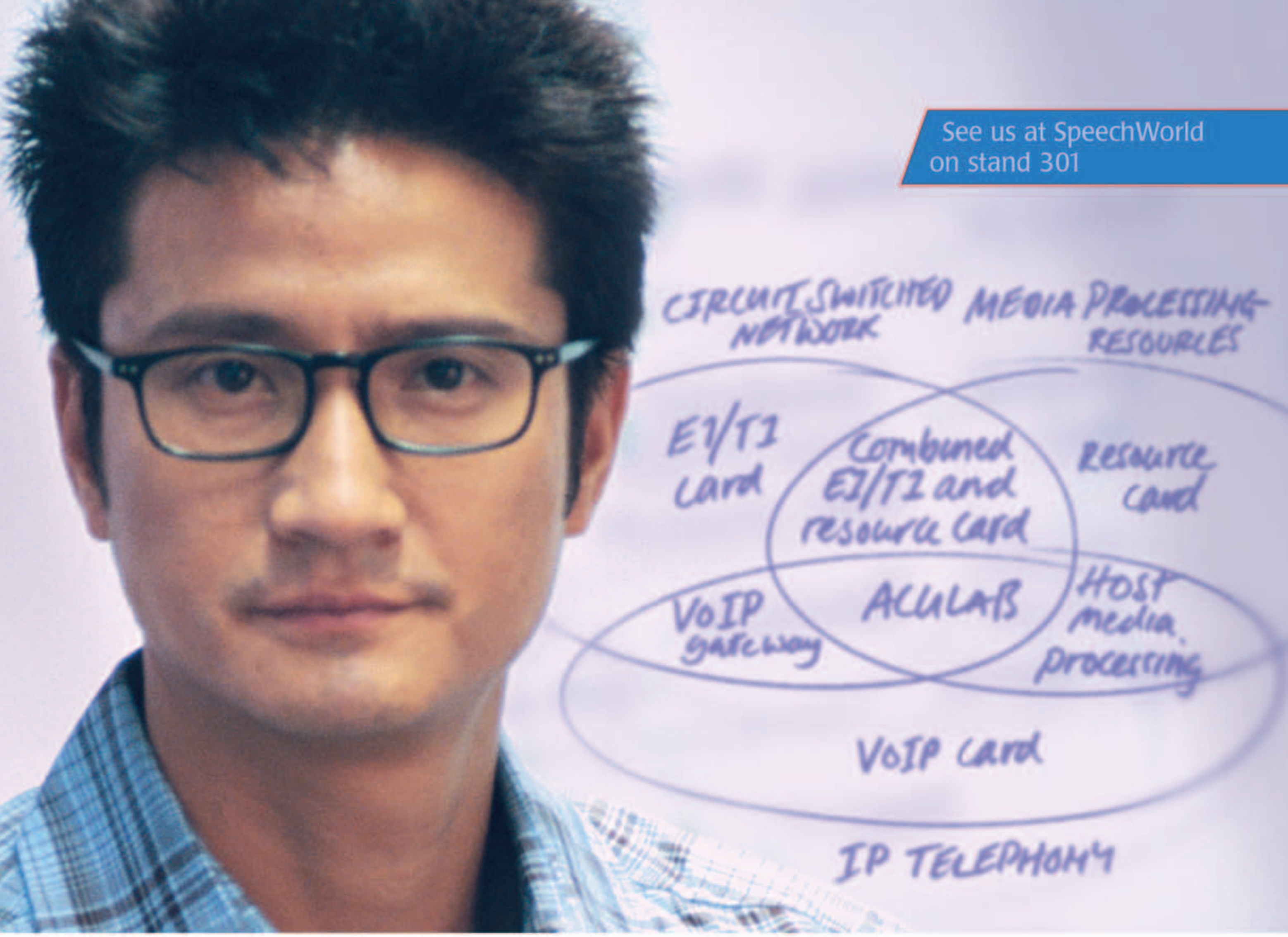
Many times we know who is calling but it is more important to know why.

In my opinion, communications is broken or at least severely damaged. We are becoming less efficient in dealing with the human elements of communication. Technology is only making our lives more challenging. We all keep in touch with many more people than we ever could because of IM, cell phones, and e-mail. Our lives are becoming unmanageable. Virtually everyone I know works on e-mails on nights and weekends. Others come to work an hour early to handle voicemail. There is just no other way to keep up. I am not advocating the cessation of work but I am advocating technologies that help automate the tasks that can be automated.

In my opinion, communications is broken or at least severely damaged. Technology is only making our lives more challenging.

There are so many things that we need to have at our disposal. Companies need to be able to set up ad-hoc conference bridges, with the ability to use application sharing and white boarding technologies. All important employees need to be remotely connected to their offices when they are on the road. They need to be seamlessly connected so when their office phone rings, they take the call.

Prioritization by caller is only part of the equation as priority level needs to naturally take into account the reason for the call. Interrupting a customer call to take a call from your boss only to find out that she wants to know if you



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want chicken or fish at the company luncheon is counterproductive. Similarly a top customer calling to cancel or initiate a million dollar contract is an essential contact and should be routed immediately to the highest level within a company. To put this in other words, a million-dollar customer call should be answered by the CEO if no one else is immediately available!

We are near the point where presence is becoming widely deployed. But really this technology needs to evolve. The next level of presence is something I have referred to in my live presentations as "Presence Genius."

In this scenario, callers should not have a phone ring unless their priority (this is derived from their personal priority level coupled with the reason for the call) is above a certain threshold. In all other cases, the call will result in a request for a conversation via a desired communications medium such as phone or e-mail or chat. Automation can even auto-schedule a phone appointment between parties as long as the priority level is established and verified.

Both parties have the ability to postpone the virtual meeting at will if needed. Think of this as a snooze button for your conversation. The other party will be notified of the delay and automation will determine the appropriate time to schedule the next call based on openings on your calendars.

An example of the above is that I may not want to take a call from a family member to invite me to dinner if I have a terribly busy work day. It should go to voicemail. I will

TMC Event Outlook

The success of our recent Internet Telephony Conference & EXPO notwithstanding, the TMC events team is plugging away, making sure that our upcoming events are worthy of the TMC name. Here is a snapshot of some of the conferences we have coming up in the next few months:

Speech-World Comes To Dallas

I can't be more excited about our upcoming Speech-World and IP-Contact Center event in Dallas, TX, May 24–26, 2005, at the Westin Park Central. These two technologies are revolutionizing how we do business. Speech for example is helping companies reduce cost while increasing customer service levels. This technology has gone from the lab to actual successful deployment in the last few years and now is the time to check it out for yourself. VoIP too has entered the call center and is causing a tremendous change in the way we do business. The world will never be the same. What is most amazing about IP is how it allows companies to take advantage of what they never could easily do before. For example the advent of the virtual contact center allows agents to be located anywhere, from their homes to another continent. The key in all these cases is seamless management. You don't lose any control yet you can take advantage of lower labor rates and increased redundancy by turning your physical center into a virtual one.

I look forward to personally welcoming you all to this historic event. We are very excited to be launching this show in Texas. The exhibit hall is free and the conferences come with a guarantee unheard of in our industry. Please check out www.speech-world.com for details.

VoIP Developer

VoIP is changing the way the world communicates, and the world is taking note. If you are a developer, and if you have any intention of seizing what is undoubtedly the most promising telecom opportunity of the past several decades, then I encourage you to attend the VoIP Developer Conference, August 2–3, 2005 in San Francisco, California. The event will be taking place at the South San Francisco Conference Center, and our team is hard at work preparing the conference as you read this.

The second VoIP Developer Conference is the only chance you'll have in the United States this year to come learn how to quickly develop new VoIP applications that are in high demand.

For all you developers out there and those looking for partners, make sure to mark your calendars for VoIP Developer. This event is expected to have a sold out exhibit hall once again and this conference is the only one in the world focusing exclusively on partners and developers. We expect this show to be a resounding success.

Service Provider Summit

Through the years, Internet Telephony Conference & EXPO has earned its reputation for being the preeminent venue for service providers to learn and to network with colleagues. As we now move into the next phase of VoIP, our upcoming Los Angeles event (Los Angeles Convention Center, October 24–27, 2005) will be no exception. Our inaugural Service Provider Summit (sponsored by the IPCC) is dedicated to discussing the issues most important to service providers, and we plan to make this an amazing educational experience unmatched in the industry. Pure education. No Fluff. We have an amazing collection of great ideas that we are turning into sessions. In fact, at our most recent event in Miami, our Service provider education sessions were standing room only! The Session Border Controller Shootout was jammed as was the UNE-P to VoIP Summit. And, a key fact to note is that both of these sessions took place at the same time.

Service providers are the fastest growing segment of our conferences. This will be a must-attend event for every service provider touching VoIP. Wireless, WiFi (define - news - alert), Cable, Triple-Play, CLECs, ITSPs (define - news - alert), and ILECs (define - news - alert). The program is designed to help service providers of every stripe solve their pain points in rolling out VoIP in their networks. IT

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return it at lunch. If I am being called for a family emergency, then of course I need to be interrupted. Do you see the irony here? In order to find out whether or not you should answer a call, you need to answer it. Sure, some of us have assistants that perform this function but assigning a personal assistant to the world population is not something likely unless President Bush changes his stance on cloning.

Communications by its very nature is inefficient and in a world where productivity increases are more and more important, we need to squeeze inefficiency out of how we interact with one another.

The good news is that companies are trying to solve the problems above. The concern I have is that they are all going about it in the wrong way. At a recent Internet Telephony Conference & EXPO, many of the keynote speakers talked about how they would solve the above problems. If you listened closely to their presentations, they all called what they are doing by a different name. Here are examples of what vendors call this sort of technology: VoIP (define - news - alert), productivity boosting applications, converged applications, communications software, mobility solutions, real-time communications, etc. There are probably 100 different descriptive phrases floating around that are meant to describe the same thing!

Are these companies kidding? Who on earth is going to buy these solutions in bulk if we describe them in a hundreds of ways? Can you imagine a company getting presentations from five PBX vendors and they all call their applications by a different name? That is how it works today! These technologies are essential yet we aren't seeing fast enough adoption because CFOs could care less about this stuff. I don't blame them. VoIP was adopted quickly once the industry agreed on this term. Call centers grew quickly once we agreed on a term. CRM(define - news - alert) is another term that literally turned into a multi-billion dollar market overnight! We need a term and that term isn't real-time communications as some analysts like to tout. This is totally wrong.

When you are on vacation in Hawaii and your cell phone rings and your boss calls to ask you if you want chicken or fish at the company luncheon, real-time communications is exactly what you don't want.

So what do I advocate as a term? We are trying to wring efficiency from communications and make us all more productive. The same ideas were applied to manufacturing and resulted in the term Just in Time Manufacturing. Borrowing a term that works and makes sense, we should call it Just in Time Communications.

We are all trying to solve the same problem and if we want to be taken seriously by CEOs and CFOs, we better agree on a term and stick with it. This term has implications for consumers, contact centers, enterprises, government and the military. It works for every vertical market out there from financial to insurance to medical. It just plain works and is desperately needed to take us to the next level. The next time you are in an important meeting and you take a call that isn't so important, think Just in Time Communications.

We are trying to wring efficiency from communications and make us all more productive. We should call it Just in Time Communications.

Whenever a new term is developed to describe something I always like to see if it passes the cocktail party test. WiFi was coined to rhyme with HiFi. I think I could tell people at a cocktail party that I am involved in Just in Time Communications and they would know what I do. Of course once we acronymize it (yes, I invented yet anther term) the term turns into JITC... You may not want to mention this term over martinis at the next happy hour — unless you enjoy blank stares. IT

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Popular Telephony, Teledex In Strategic Alliance

Teledex (news - alert) and Popular Telephony (news - alert) have announced that they have entered into an agreement whereby the two companies will work together to develop applications and products designed to enhance the functionality of both conventional and IP-enabled hospitality PBXs, as well as telephones and other endpoints.

Utilizing Peerio, Popular Telephony's peer-to-peer technology, and Teledex's VoIP designs, the two companies aim to produce content-supporting applications, services, and back-end support applications than can augment existing hospitality PBX functionality without requiring additional PBX upgrades or investments. By deploying such applications on Popular Telephony's "serverless" network architecture, utilizing the shared computing power and storage capacity of a "virtual" network, hotel operating companies can recognize substantial cost savings, as well as provide significant guest-facing technological innovations that can be constantly upgraded at little or no cost.

Popular Telephony's Peerio networking solution creates a virtual network that resides on the clients on the network; a serverless architecture that distributes the computing power and storage capacity necessary to deliver telecommunications functions.

Teledex has shipped over 8 million guestroom telephones to over 125 countries.

In other news, Popular Telephony has announced the immediate general availability of the Peerio Open Source NAT/Firewall (FW) Traversal library.

The Peerio Open Source NAT/Firewall Traversal library is designed to enable VoIP users and carriers to easily bypass firewalls when communicating over IP networks. The NAT/Firewall Traversal is also being incorporated across all Peerio-based enterprise solutions and end-user applications, including the newest version of Peerio GNUP, creating a seamless bridge between various standards-based protocols.
http://www.populartelephony.com
http://www.teledex.com

Go To Table of Contents | Go To Ad Index

Aastra Launches P2P IP-based Enterprise Phone System

Aastra Technologies (quote - news - alert)launched the VentureIP, an enterprise-class, P2P (peer-to-peer) IP-based phone system. Powered by Nimcat Networks' nimX software, the system is designed to enable small to-medium business (SMBs) to install, operate and manage a full-featured phone system by simply plugging the VentureIP 480i telephone into their local-area network (LAN). The system scales on a phone by phone basis and can be connected to the local PSTN via the VentureIP Gateway.

When a user connects a VentureIP 480i telephone into their data network, the device automatically configures itself, allowing calls to be made and received without any additional setup or centralized server equipment. Additionally, through its connection to the LAN and the Internet, users have Web-based access to the VentureIP system to manage their set's options, as well as access to future software enhancements.

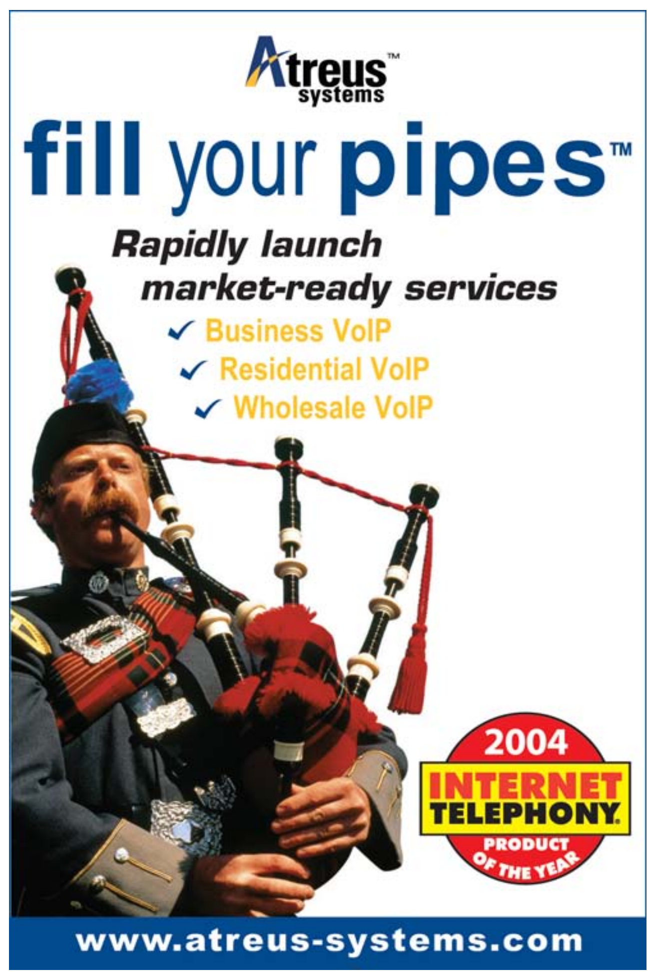
Each VentureIP Gateway supports up to four telephone lines, while multiple gateways can be supported on the same network. Just like the VentureIP 480i telephone, the VentureIP Gateway is a self-aware device and automatically configures itself within the system.

Key Features of the VentureIP Telephone System include:

- Low-cost IP-based P2P telephony system;
- Complete PBX telephony features including: voice mail, intercom/page and auto-attendant;
- Eight-line graphic display with six dynamic context-sensitive softkeys;
- Premium speakerphone performance;
- Automatic start-up with no configuration required;
- Systems scales set by set – no need to purchase a minimum number of units;
- Completely automated for adds, moves, and changes.

The system is available immediately from current Aastra distributors. Suggested retail pricing is $379.00 and $289.00 for the VentureIP 480i and the VentureIP gateway respectively.

http://www.aastra.com



Zyxel Delivers VoIP Station Gateway

ZyXEL Communications, Inc., (news - alert) has announced its Prestige 2302R Voice over Internet Protocol (VoIP) Station Gateway designed to enable small office/home office (SOHO), small-to-medium businesses (SMBs), and home users to share broadband resources while fulfilling VoIP application demands through their existing DSL/Cable modem simultaneously.

The ZyXEL P-2302R is an integrated high-speed broadband sharing gateway designed for customers of telecommunications providers and IP telephony service providers (ITSPs). With two 10/100 Ethernet ports and two POTS phone interfaces, the P-2302R enables users to simultaneously make VoIP conversations while surfing the Internet. As each phone jack works independently, the P-2302R can be used with two separate VoIP accounts without purchasing additional adapters. In addition, users do not need a special VoIP phone as the P-2302R is compliant with thousands of analog handsets available on the market today.

The ZyXEL P-2302R supports the open SIP (Session Initiation Protocol RFC 3261) multimedia protocol to ensure interoperability with other SIP-based equipment. Its auto-provisioning functionality, a mechanism that allows service providers to manage and configure the P-2302R remotely, is designed to significantly decrease installation costs. In addition, the ZyXEL P-2302R's VoIP platform features multiple SIP and voice channels for convenient design and deployment of VoIP services.

Also available from ZyXEL is the Prestige 2602HW all-in-one ADSL VoIP Integrated Access Device (IAD) to offer IP-based voice communication with superb sound quality allowing users to take advantage of many new and existing IP telephony features. In addition to firewall and VPN security, the Prestige IAD provides high-speed wireless connectivity with IEEE 802.11g wireless standard compliance.

The ZyXEL P-2302R is currently available through ZyXEL's national network of authorized resellers.

http://www.us.zyxel.com

Go To Table of Contents | Go To Ad Index

Sony, GlowPoint Join To Deliver A Customized Solution

Sony Electronics, Inc.(quote - news - alert), and GlowPoint, Inc.(quote - news - alert), recently announced that they have entered into a memorandum of understanding outlining a strategic alliance for the two companies to create and launch a complete, Sony customized, user-friendly video communication solution focused on broadening the use of IP-based video in and out of traditional office environments.

The Sony service, powered by GlowPoint, will bring together Sony's state-of-the-art line of video conferencing systems with GlowPoint's patent-pending advanced IP-based video applications and network services. Together, Sony and GlowPoint aim to increase usage of traditional video conferencing systems and accelerate video communications' migration to the desktop. The two companies are also developing joint initiatives to support the growing use of IP-based video for more diverse and innovative applications, including the broadcasting production segment where Sony is a recognized leader in providing customers with advanced audio and video equipment and systems.

Through a "white label" arrangement, the two companies plan on delivering a customized Sony experience that will allow subscribers to see and talk to virtually anyone, anywhere around the world regardless of network, technology and device.

Scheduled to be available by mid 2005, the GlowPoint powered service will incorporate the company's "All You Can See" Unlimited Video Calling plans and its video communications features, including Direct Dial Video Numbers, which simplify video calling by replacing confusing IP addresses with standard 10-digit telephone numbers; "000" Live Video Operator Service, which instantly connects callers to a live video operator for assistance and information; and "My Video Meeting" which provides each subscriber a personalized video number for spontaneous, multi-party video calls.

According to Michael McCausland, vice president and general manager of Sony Electronics' IP Communications division, "We share the GlowPoint mission to make video an integral communications tool, and by working with a recognized industry leader, Sony now has the ability to more quickly introduce customer driven video solutions tailored to our brand, product and distribution channels."

"The alliance with Sony allows GlowPoint to address a greater variety of innovative IP-video uses more quickly, while at the same time to reach an increased number of potential customers more effectively,'" said David Trachtenberg, president and CEO of GlowPoint. "Who better than Sony to adapt technology seamlessly into customers' everyday lives?"

http://www.sony.com/videoconference
http://www.glowpoint.com

snom360 Business IP Phone Launched At ITExpo

ABP Technology (news - alert) recently announced the launch of the newest member of snom technology's family of SIP business phones. The snom 360 offers a number of powerful business telephone features, including dedicated keys for quick access to all key audio and call control functions; context-sensitive menus for efficient management of relevant capabilities as calls progress; 12 programmable keys with LED; tiltable backlit graphical display; sophisticated call control features; full call detail, configuration options, and online help accessed via Web browser; and more.

"This is the best snom phone yet" says Robert Messer, ABP Technology's President "In our eyes, it is the fulfillment of many years of work building the ideal vendor independent and open standards-based business phone."

http://www.abptech.com
http://www.snom.de

Telco Intros Access241

Telco Systems (news - alert) recently introduced the Access241, a VoIP Analog Telephone Adapter (ATA) designed to integrate traditional telephone service and VoIP phone service over a single telephone for enhanced calling capabilities and to facilitate home and small office networking.

The Access241 adds automated call routing to Telco's ATA product line. For instance, during an active VoIP call, if another call comes in on their traditional service, a call waiting beep indicates an incoming call, and a standard flash-hook enables the user to switch between calls. In addition, a push of a button on the phone dial pad enables the user to switch between phone services when making a call.

The Access241 also provides enhanced telephony services, such as 911 and user defined call routing. When dialing 911, the emergency call is automatically routed to the traditional phone service to ensure the emergency operator is able to determine the exact location of the emergency. Users can also designate which service to route a call over based on a specific phone number or area code.

http://www.telco.com

Quick Hits

Pangea Adds Voice Messaging

Pangea Communications (news - alert) has announced the addition of Voice Messaging to its Concert Fax Solution. CFS enables service providers to offer Internet fax and messaging services including Fax to e-mail, e-mail to Fax, Web to Fax, PC to Fax, and Fax Broadcasting services to their clients. SishaFax of Johannesburg, South Africa will be the first Licensed Operator of CFS to offer the new service to its clients.

http://www.pangea-comm.com
http://www.sishafax.co.za

AccessLine Adds VoIP E9-1-1 To SmartVoice Service

AccessLine Communications' (news - alert) new SmartVoice Service for business will offer enterprise customers E9-1-1 protection over VoIP connections. Based on V9-1-1 Mobility Service technology from Intrado Inc., the solution provides a similar level of E9-1-1 protection to that available over traditional wireline connections.

http://www.accessline.com

Mediatrix Unveils Liaison

Mediatrix Telecom, Inc.(news - alert), announced the launch of its new Mediatrix Liaison residential VoIP product line, a range of VoIP access devices offering seamless convergence of broadband modem and VoIP gateway functions. The Mediatrix Liaison is designed for home or home-office use, connecting up to two analog phones and/or faxes, as well as a PC or network router to a broadband connection.

http://www.mediatrix.com

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Go To Table of Contents | Go To Ad Index

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TWT Launches VoIP Metro Ethernet Solutions

Time Warner Telecom (quote - news -alert) recently announced its VoIP-based business solutions strategy and new service offerings at the Internet Telephony Conference and EXPO. The company's business-class VoIP solution suite — TW Telecom ONE SOLUTION — is available to business customers in 21 initial markets.

"The burning question in the minds of customers is not, what technology, but what solution, will help me win in a very competitive marketplace? How the industry takes technologies, like VoIP, and crafts them into real solutions that deliver real benefits for customers, is what separates the contenders from the pretenders. As more and more businesses converge their voice and data networks, aligning with the right service provider is becoming even more critical," said Michael A. Rouleau, senior vice president of strategy and business development for Time Warner Telecom.

The first phase of VoIP-based products offered under TW Telecom ONE SOLUTION, provides solutions for PBX customers. These products include: TW Telecom ONE CONNECT, a VoIP trunking solution; TW Telecom ONE REACH, an IP FX virtual numbering service; and TW Telecom ONE FORUM, a conferencing solution. In concert with this VoIP-based business strategy launch, Time Warner Telecom is offering free VoIP VPN service to all its site-to-site customers in the 44 markets it serves.

http://www.twtelecom.com

tel^x Launches Voice Peering Fabric

telx (news - alert) announced the launch of The Voice Peering Fabric. The deployment of the fabric is designed to facilitate a radical economic shift in voice communications as calls can now traverse the fabric via on-net networks and omit hitting the traditional phone companies' networks, thereby eliminating the need for calls billed per minute.

"The VPF is the first and by far, most predominant Ethernet Fabric designed to enable complete transparency between buyers and sellers of Voice over IP traffic," states Shrihari Pandit, the CEO and Founder of Stealth Communications. "The VPF allows VoIP service providers, enterprise customers, and university network managers to instantly know all the available on-net networks, their most competitive rates, and then… they are able to get connected to these networks and services within hours. This not only reduces network operating costs, it adds the choice of competitive, wholesale rates, and control to utilize any on-net service provider, at any given moment."

http://www.thevpf.com

Amedia, Lightspeed Technologies Join Forces

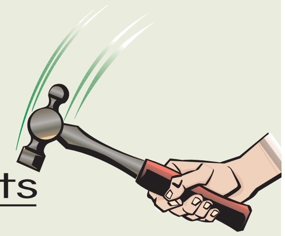
Amedia Networks, Inc.(news - alert) , and LightSpeed Technologies, Inc. (news - alert) , announced that they have entered into an agreement to include Amedia's QoStream products as a part of LightSpeed's overall value proposition.

Under the agreement, LightSpeed will provide marketing, sales, systems integration and professional services to meet the expanding market of homes served by Fiber-to-the-Premises (FTTP) systems, a market that doubled in 2004.

Amedia will make available to LightSpeed the QoStream AS5000 Aggregator Switch, the QoStream PG1000 Premises Gateway (both the inside and hardened outside versions), as well as the QoStream Director Network Management System. These Ethernet Switched Optical Network (ESON) based products allow each subscriber to receive up to 100 Mbps over a 90 km reach from a central office or head end.

http://www.LightSpeedt.com

http://www.amedia.com



Quick Hits

Northland Cable Launches Service With Net2Phone

Northland Cable Television (news - alert) announced that it has launched its "Northland VoiceLine, Powered by Net2Phone" digital phone service, offering consumers premium full-featured voice service. With plans starting at $18.99, Northland VoiceLine is designed to offer consumers a new feature rich low cost local and long distance choice in voice service. Its all-inclusive monthly package for $37.99 delivers unlimited calls within the United States and Canada.

http://www.northlandcabletv.com

http://www.net2phone.com

Band-X, NexTone Team For VoIP Services

Band-X (news -alert) and NexTone Communications (news - alert) announced that Band-X has deployed NexTone's Multiprotocol Session Controller as a critical part of their strategy to expand its VoIP services portfolio and to support the growing number of carriers to which it provides VoIP interconnections. New VoIP customers can connect via H.323 or SIP to access the full Band-X supply base of over 100 operators.

http://www.nextone.com

http://www.band-x.com

Netspoke To Use NMS' Open Access

Netspoke (news - alert) recently announced a hosted VoIP conferencing service, which it will build using the Open Access development platform from NMS Communica-tions. In addition to offering features such as real-time communication and collaboration capabilities, the Netspoke IP conferencing service will provide enterprise users other key benefits, including capital expenditure reduction, lowered organizational costs and increased workforce efficiency.

http:///www.netspoke.com

http://www.nmscommunications.com

Go To Table of Contents | Go To Ad Index

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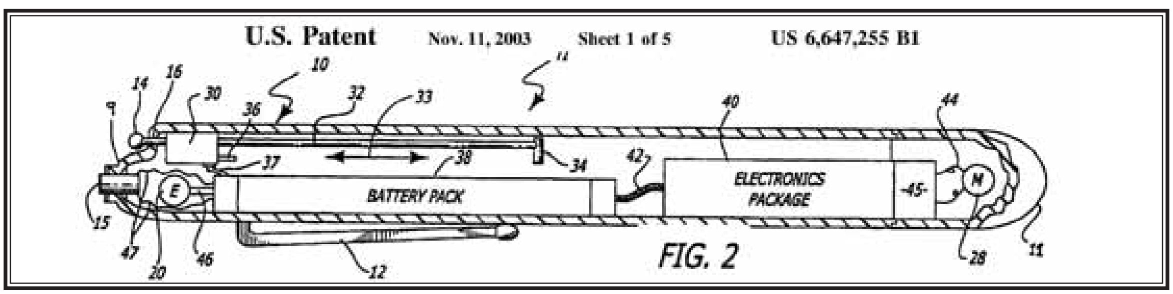
Broadcom Eases WiFi Installation

Broadcom (quote -news - alert) announced that its WiFi Phone reference platform now includes SecureEasySetup software designed to enable consumers to effortlessly install their wireless IP phones with the push of a button.

Driven by the availability of cost-effective VoIP solutions, the number of subscribers who will use WiFi-based voice services in the United States is estimated to reach 75 million by 2008 according to In-Stat. Installing a WiFi Phone in today's secure wireless network can be a complicated and time-consuming process, requiring a user to manually configure several settings on a phone keypad. The Broadcom SecureEasySetup software is designed to simplify the installation of wireless IP phones and other WiFi devices enabling users to configure their WiFi phones by simply pushing a button on the wireless router or access point, and a button on the handset. The phone automatically connects to the wireless network and WiFi Protected Access security activates. WPA is the strongest security standard for protecting home wireless networks from unauthorized use.

http://www.broadcom.com

"Ebay" Auction: Mobile Phone Patent Portfolio



The bidding begins, for a limited time, on May 11, 2005

In a revolutionary, **ONE TIME**, offer this valuable patent portfolio will be sold at auction on "ebay.com". The bidding begins, for a limited time, on May 11, 2005. Enter "U.S. Patent Mobile Phone" in the search box at "ebay.com" on that date.

The same team that prosecuted the famous Ronald A. Katz patents prosecuted this portfolio. Those patents generated **over 700 million dollars in income**.

Because of the limited bidding period, all interested parties should complete their analysis well before the bidding begins. For file histories, etc., simply e-mail your request to: **telpatgroup@charter.net**

The portfolio includes U.S. Patents: 6,049,710; 6,149,353; 6,292,675; 6,298,250; 6,308,053; 6,400,967; 6,405,029; 6,473,610; 6,580,927; 6,584,327; 6,647,255; 6,751,482; 6,754,481; 6,845,234 and 6,862,463. One additional patent is in issue and there are three pending applications.

This is a great opportunity for cellular, dot com, numberless, data locating and other telephonic, as well as financial, marketing and service operations wishing to protect and expand their businesses.

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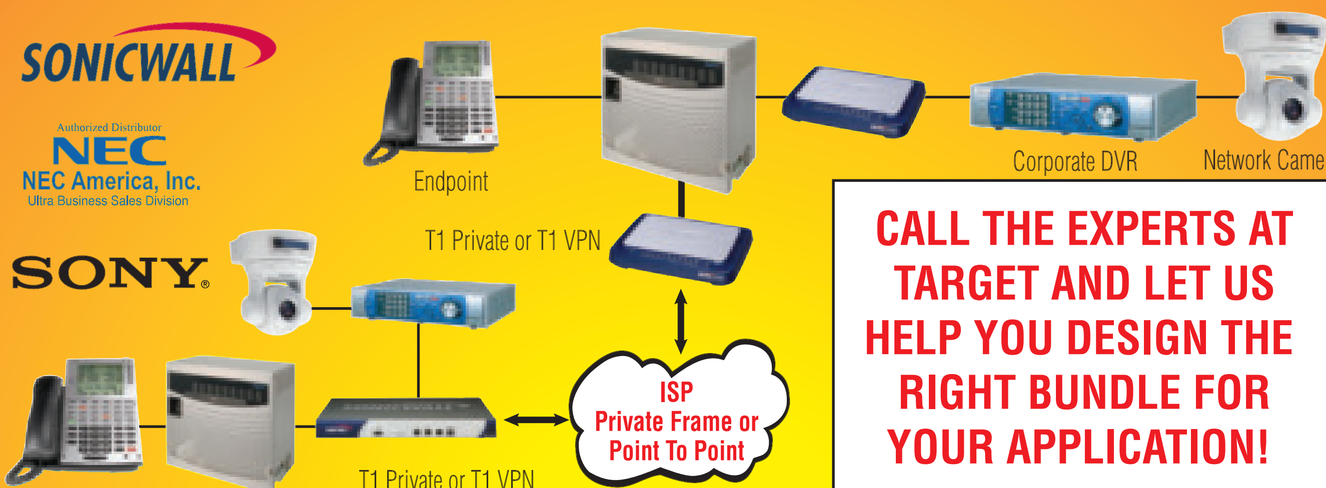

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Linksys Ships Over One Million VoIP Ports In Less Than Six Months

Linksys, (news - alert) a division of Cisco Systems, Inc., announced it has shipped over one million VoIP ports in six months. Linksys products for VoIP include a phone adapter and wired and wireless routers with phone ports. These products, bundled with a VoIP service are designed to enable customers to make phone calls using their broadband connection.

Since August, Linksys has announced relationships with leading broadband VoIP service providers including AT&T CallVantage, PeopleCall in Spain, PhoneSystems.net in France, Verizon VoiceWing, and Vonage in both North America and the United Kingdom. Linksys hardware, combined with these service offerings, makes a complete, solution for consumers.

Synergy Research Group, a leading market research firm, reports that today there are close to 8 million home users worldwide using VoIP to make phone calls over their broadband connections. By 2009, Synergy anticipates this number to grow to 58.9 million home users using VoIP to talk to friends and family around the world using broadband and VoIP technology.

http://www.linksys.com

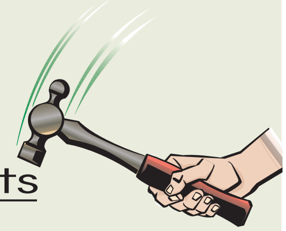
SpectraLink Unveils New WiFi Handset For Healthcare

SpectraLink Corp. (quote -news - alert) recently announced its new NetLink h340 Wireless Telephone. Designed for the needs of the healthcare industry, the WiFi handsets provide immediate communication for nurses, doctors, and hospital staff with the privacy, security, voice quality, and other call features they depend on.

The NetLink h340 Wireless Telephone is only 5.5 inches long and weighs just four ounces. Belying its small size, the handset is constructed with some of the most durable components in the industry.

The NetLink h340 leverages the features of a hospital's existing telephone system including multiple line appearances, caller ID, call transfer, conferencing and voice mail integration. Privacy and security are assured through industry-standard wireless security protocols, and excellent voice quality is maintained over a converged wireless network by the widely adopted SpectraLink Voice Priority (SVP) quality of service (QoS) mechanism.

http://www.spectralink.com



Quick Hits

DataLogic Introduces Integrated WiFi Data/Voice Router

DataLogic International (quote - news -alert)announced that it will add to its product suite an integrated data/voice Wi-Fi router next month. The 802.11g router will be offered through subsidiary IPN Communications. The Wi-Fi router will support up to four IP voice ports, allowing the router to not only wirelessly network devices but to also route traditional telephone calls to an IP network. The Wi-Fi router can be configured to route calls through IPN's long-distance carrier or any carrier of choice.

http://www.dlgi.com
http://www.ipncom.com

Telephony Patent Portfolio Up For Auction On eBay

Mr. Ronald Katz created a patent portfolio that has produced over $700 million in licensing royalties. Now, another portfolio prosecuted by the same team that prosecuted the Katz portfolio is going to auction on eBay. The auction will take place on May 11, 2005 — simply go to the eBay (quote - news - alert) Web site and enter the search term: U.S. Patent Mobile Phone. Included are 15 patents that apply to various telephonic fields, including prepaid cellular, voice activation, data transfer, financial transactions, cellular programming, business location, disposables, and numberless phones.

http://www.ebay.com

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PACTOLUS

TranSwitch, Octasic Devices Deliver Voice Gateway Card

TranSwitch Corporation (quote - news - alert) and Octasic Semiconductor, Inc., (news - alert) have announced an integrated solution for voice over packet (VoP) gateway implementations designed to allow OEMs to increase density and lower costs. Through the complementary fit and functionality of the TranSwitch TEPro, Octasic OCT8304, and OCT6100 devices, the two semiconductor vendors are able to implement a gateway's voice plane functionality on a single card, using only three chips, resulting in a faster time to market for OEMs.

The companies also announced that a leading manufacturer of PON optical line terminals (OLT), used to enable fiber-to-the-premises (FTTP), has completed its design and field trials of a retrofittable DS3-rate high-density VoATM gateway using the TranSwitch/Octasic solution.

The Octasic and TranSwitch devices are interconnected with their common H.110 interfaces. Originally designed for CompactPCI applications incorporating a TDM backplane, H.110 is also usable as a device-to-device interconnect, enabling the full-duplex transfer of over 2,000 voice channels across one bus.

"Octasic's devices combined with TranSwitch's TEPro greatly reduce time to market for equipment manufacturers due to the ease of integration," said Frederic Bourget, Octasic's director product management. "Together we provide an end-to-end solution from packet interfaces to Voice Processing to TDM."

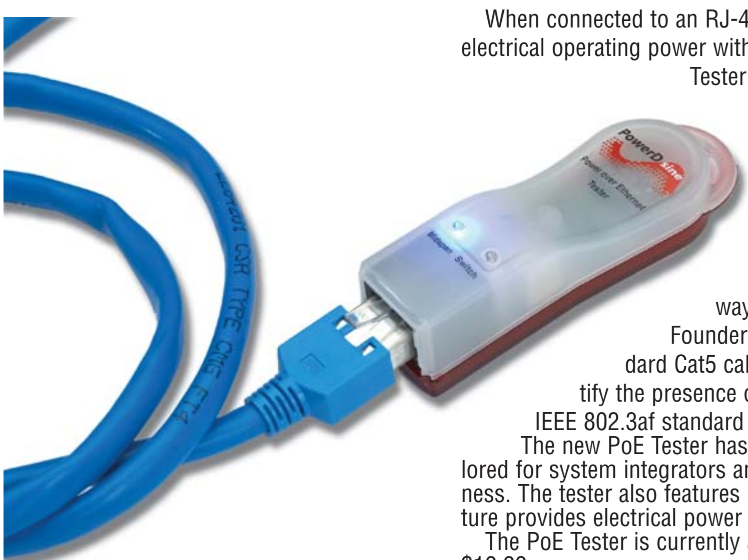
Octasic's OCT8304 Packetization Engine offers complete, standard compliant, AAL2 implementation at an unprecedented 1,023 channel capacity, the highest capacity in voice packetization and aggregation on the market. The device provides standalone high-density G.711 packetization for VoIP and VoATM while enabling easy integration with devices where other requirements are needed.

Octasic's OCT6100 Series Echo Canceller performs carrier-class transparent echo cancellation and Voice Quality Enhancements (VQE) with extremely low power and small board space requirements (16mm square footprint, the smallest in the industry). Some features include:

http://www.octasic.com
http://www.transwitch.com

PowerDsine Releases Power over Ethernet Tester

PowerDsine Ltd. (quote - news - alert) announced the commercial release of the PowerDsine PoE Tester, which is designed for IT administrators, system integrators, and installers, who want to check if electrical operating power is being carried over the Ethernet network.



When connected to an RJ-45 outlet, PowerDsine's PoE Tester checks for the presence of electrical operating power within existing Ethernet cabling infrastructure. In addition, the Tester identifies the Power Sourcing Equipment (PSE), which can be either an Endspan (a PoE-enabled switch) or a PoE Midspan. Specifically, the PoE Tester can detect IEEE 802.3af standard-compliant Midspans and Endspans. The tester can also be used to detect operating electrical power originated by Cisco pre-standard in-line power switches.

"The PowerDsine PoE Tester is a quick and convenient way to identify PoE in an Ethernet LAN," said PowerDsine Co-Founder and CEO Igal Rotem. "An IT administrator can plug a standard Cat5 cable into our tester and an RJ-45 outlet, and immediately identify the presence of power, the source of the power, and whether the power is IEEE 802.3af standard compliant."

The new PoE Tester has a compact design, the size of key, which is specifically tailored for system integrators and installers and is simple to use with plug-and-play readiness. The tester also features LED display, which indicates whether the Ethernet infrastructure provides electrical power as well as the source of the power.

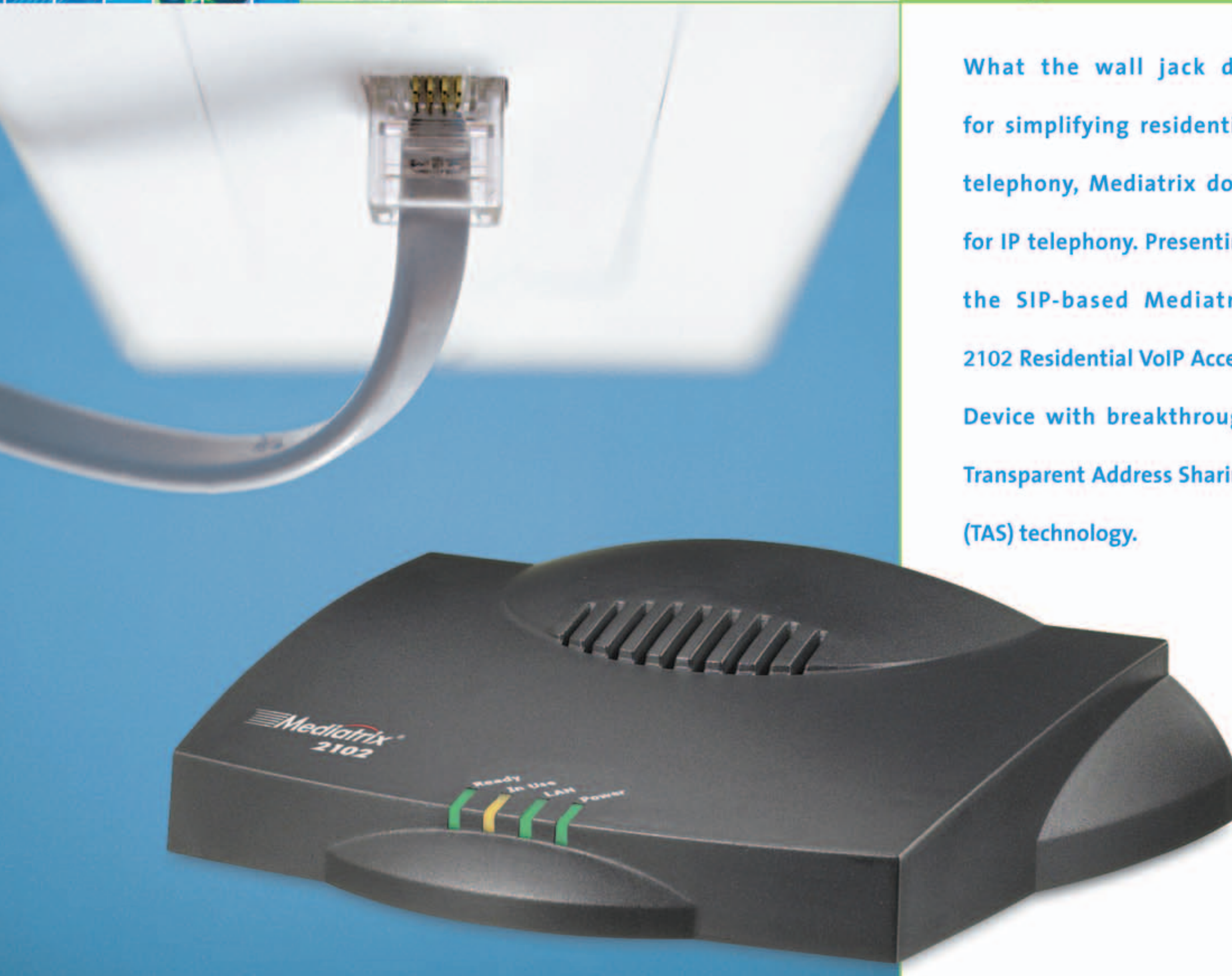
The PoE Tester is currently available from PowerDsine's distributors at a list price of $19.99.

http://www.powerdsine.com

Subscribe FREE online at www.itmag.com

Go To Table of Contents | Go To Ad Index

Residential VoIP... simplified



What the wall jack did for simplifying residential telephony, Mediatrix does for IP telephony. Presenting the SIP-based Mediatrix 2102 Residential VoIP Access Device with breakthrough Transparent Address Sharing (TAS) technology.

Keep it simple with Mediatrix

The Mediatrix 2102 does away with conventional wisdom to enable standard home phones, faxes and PCs to simultaneously access the Internet - using a single broadband connection and without the need for a router or a NAT.

For service providers, the Mediatrix 2102 simply means peace of mind and increased cost savings. It requires little or no end-user intervention: it auto-provisions itself, it's remotely manageable through HTTP and SNMP, and it features built-in QoS to ensure superior quality voice transmission every time.

To your customers, it's as easy as plugging a phone jack into a wall outlet. And to you, that means new immediate revenue generating opportunities.

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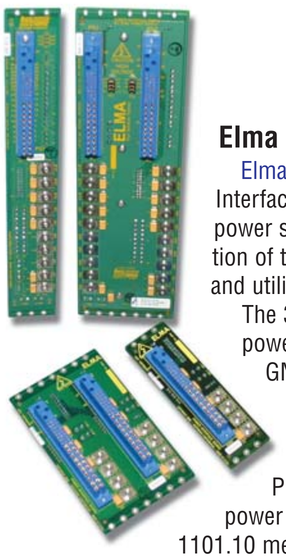


Elma To Offer PIB Family In Several Configurations

Elma Electronic, Inc., (news - alert) announced that it now offers Power Interface Boards (PIBs) in 3U and 6U versions with either one or two 47-pin power supply connectors. The PIBs are separate boards for the power section of the backplane used to facilitate pluggable power supplies, headers, and utility connectors.

The 3U and 6U power boards contain one or two 47-pin hot pluggable power supply connectors, and power taps for +5V, 3.3V, -12V, +12V, and GND. Other features include a header for voltage sense and current sharing, an IPMB interface, an ATX power connector, an auxiliary/disk drive connector, and a power switch header. The IPMB interface is compliant to system management specification PICMG 2.9 Rev. 1.0. The PIB is also designed to comply with the power interface specification PICMG 2.11 Rev. 1.0 and with the IEEE 1101.10 mechanical specification.

http://www.elma.com

Themis Announces 64-bit PowerPC VMEbus Computers

Themis Computer (news - alert) has announced a new family of PPC64 VMEbus computers. The PPC64 is Themis' first in a new family of 6U VMEbus computer boards based on the IBM PowerPC 970FX processor. Two PPC64 configurations are available; a single processor single slot computer and a dual-processor, symmetric multiprocessing, two-slot solution. The PowerPC 970FX processor provides maximum performance for existing 32-bit applications and new 64-bit applications.

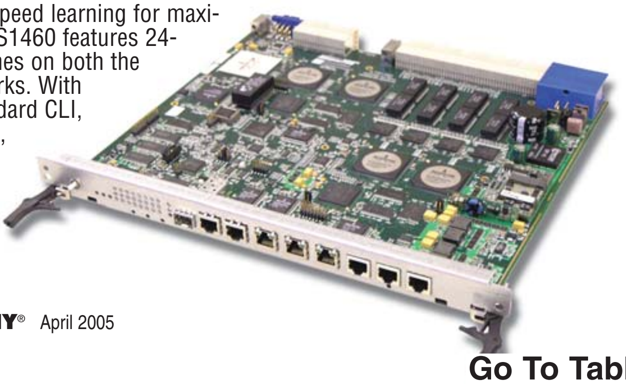
With the addition of the PPC64 to its rapidly growing product family, Themis offers an ever-wider breadth of rugged, high performance VME embedded computers. The 1.8 GHz PPC64, in a single-slot configuration, can be configured with up to 4 GBytes of DDR400 memory and includes a Gigabit Ethernet port and support for Dual Ultra320 SCSI drives. The single-processor PPC64 also includes a high performance Universe II VME64x interface, two 10/100/1000 Ethernet ports, two USB ports, AC97 audio, and two serial ports. A second processor may be added, in a second VME slot, communicating with the baseboard via a Hyper-transport link.

http://www.themis.com

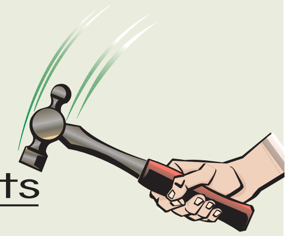
Diversified Expands AdvancedTCA Product Line

Diversified Technology, Inc., (DTI) (news - alert) recently introduced its new AdvancedTCA Hub Switch. The ATS1460 Blade is an AdvancedTCA hub board for a Dual-Star fabric based ATCA shelf. The board is compliant with the PICMG 3.0/3.1 specifications.

The ATS1460 Hub Board is an AdvancedTCA 3.0 and 3.1 Option 4 switch with support for high availability. It provides separate control plane switching, data plane switching, and storage plane switching for ATCA shelves. It supports gigabit Ethernet on the base control network. On the fabric network it supports both gigabit Ethernet for data and 2-gigabit Fiber Channel for storage. All three networks are non-blocking and feature wire-speed learning for maximum performance. The ATS1460 features 24-port gigabit Ethernet switches on both the base and fabric data networks. With support for a industry standard CLI, telnet/SSH, SNMP, RADIUS, and a web interface, the ATS1460 provides robust management.



http://www.dtims.com



Quick Hits

GL Communications Releases DCOSS v5.10

GL Communications, Inc. (news - alert) has announced the release of the latest version of DCOSS: version 5.10. DCOSS is a key piece of testing/simulation equipment used by telecommunications network engineers for testing switches, gateways, and networks using a variety of signaling protocols. These available protocols include T1/E1 CAS, PRI ISDN, SS7, and SS5.

http://www.gl.com

TI, Telchemy Team Up For QoS

Texas Instruments, Inc. (quote - news - alert), and Telchemy, Inc., announced that TI has licensed Telchemy's VQmon/EP-DS for integration into TI's VoIP solutions making it faster and easier for equipment manufacturers to incorporate VoIP performance management technology into theior solutions. The fusion of TI's VoIP chipsets and Telchemy's software is designed to enable service providers and enterprises to monitor, diagnose, and troubleshoot complex problems in real-time, ensuring high service quality and availability for next generation services.

http://www.ti.com
http://www.telchemy.com

Spirent Rolls Out Distributed Abacus Platform

Spirent Communications (news - alert) has announced availability of the Distributed Abacus IP Telephony Rollout Platform designed to assess and validate signaling performance and call quality in pre-production networks. Carriers using Distributed Abacus can introduce large-scale VoIP services by taking IP telephony testing beyond the lab. By emulating real-world traffic from multiple points on the network, network operators, may avoid the risk of testing VoIP quality and performance with live customers.

http://www.spirentcom.com

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Nortel, SIPquest Bring SIP-Based Multimedia Services To Wireless

Nortel (quote - news - alert) and SIPquest (news - alert) announced they are working together to enable service providers to deliver advanced SIP-based multimedia services to residential and corporate customers over wireless handheld devices.

The companies will initially focus on delivering advanced capabilities such as Presence and IM for services delivered to handhelds such as the RIM's Blackberry, Sony-Ericcson P900, and other smartphones. Using SIPquest's IMS Cellular Client, a soft client residing on the handheld device, and Nortel's Multimedia Communication Server (MCS) 5200, service providers will be able to deliver value-added services and productivity-enhancing tools to cellular users.

"Service Providers will be able to take advantage of network convergence and deliver, under their own brand, new and innovative multimedia services to their customers", said Alain Mouttham, SIPquest CEO. "This is an initial step in enhancing the user experience. As the market evolves, we anticipate bringing cellular and WiFi client solutions together to provide dual-mode functionality and a seamless end-user experience."

"Both companies share a vision of transforming networks, enhancing the human experience, igniting global commerce and improving security," said Jim Dondero, vice president of Carrier Packet Networks marketing, Nortel. "Today's announcement is an important first step in delivering profoundly powerful communications to end-users, providing a level of control and personalization from a handheld device not yet experienced. We will work with SIPquest to evolve these capabilities even further."

http://www.sipquest.com
http://www.nortel.com

Stalker Software Launches CommuniGate Pro v4.3

Stalker Software (news - alert) announced CommuniGate Pro v4.3 Real-Time Communications server, featuring an expanded SIP communications offering. Both single-server and dynamic-cluster 99.999 percent uptime installations are designed to provide secure SIP messaging and telephony for users anywhere they go — whether in the enterprise, at home, or on the road — dynamically adjusting for any firewalls, VPNs, and NAT devices in between.

Based on current Internet and telephony standards, CommuniGate Pro enables small and large enterprises, telecoms, and cable providers to use SIP for VoIP, multimedia, instant messaging, remote application sharing, and other real-time presence applications. CommuniGate Pro provides registry services for millions of VoIP users, SIP proxy services for thousands of call signals per second, and extensive "near-end" and "far-end" NAT traversal services only provided previously through complex architectures or expensive hardware solutions.

In the new 4.3 version, VoIP integration with e-mail has been extended to allow users to initiate SIP calls directly from their address book, using either the CommuniGate Pro MAPI connector for Outlook or Web mail. In the near future, CommuniGate Pro will extend this capability to include cutting-edge messaging control such as VoIP call management at the mailbox and user-defined SIP filtering and spam prevention techniques.

In addition, CommuniGate Pro v4.3 will now feature three new graphical user interfaces (GUI) available in 12 languages. Users have the option of using either a Microsoft Outlook-like or Microsoft Entourage-like GUI, or a new Internet mail-like GUI for Web mail access. Users can easily switch to a new skin or language simply by selecting the option from within any of the interfaces. The original, intuitive, light-weight and fast interface will still be offered for those that wish to build custom skins of their own on the framework, or for users that want a fast light interface for browsers like Lynx over a shell session remotely.

http://www.stalker.com

Data Connection SIP Supports Over 1 Billion Calls

Data Connection Limited (DCL) (news - alert) announced that Tellme Networks, Inc., has incorporated Data Connection's DC-SIP software solution into their network, enabling over 1 billion directory assistance calls using VoIP this year. Tellme is using Data Connection's SIP software to connect directory assistance calls to Tellme's data centers nationwide.

SIP is an essential part of the family of Voice over IP (VoIP) technologies that is powering innovative companies, such as Tellme, to offer more services and flexibility to their customers but at significantly less cost and complexity than that of traditional voice networks.

"Directory assistance will drive millions of SIP sessions per day to the Tellme data centers, which essentially makes our network one of the largest SIP endpoints in the world," said Troy Chevalier, Tellme's VP of Platform Engineering.

"Tellme handles a third of all the directory assistance calls made in the United States. The use of our DC-SIP solution in the Tellme application is a true testimony to the scale and high reliability of our products," said Phil McConnell, Data Connection's CEO.

http://www.dataconnection.com/sip

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Ingate To Offer VoIP Survival CPE For Hosted VoIP

Ingate Systems' (news - alert) VoIP Survival is an application module for hosted VoIP communications services designed to secure full redundancy in a SIP-based Hosted PBX or IP Centrex environment all the way out to the customer premise. The application, which can be used with any hosted VoIP service, has been validated by BroadSoft, Inc., to serve as a customer premise backup to enhance the reliability and availability of the BroadWorks VoIP application platform.

In the hosted environment, Ingate VoIP Survival ensures continued communications service even if the hosted server goes offline due to connection failure or malfunction. This increased availability means that unnecessary downtime is minimized.

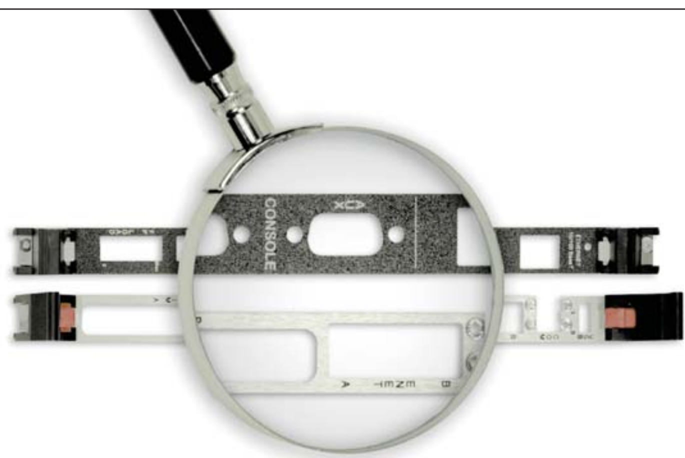
A hosted communication platform offers many advantages to enterprises adopting VoIP. However, there are occasions when it is desirable to have a redundant capability at the customer premise. Should a network outage occur, Ingate VoIP Survival offers continuous, uninterrupted availability of mission-critical VoIP service.

In a Hosted PBX environment, the server situated in the service provider's core network is in constant contact with — and dynamically sending data to — the Ingate Firewall or Ingate SIParator situated on the company LAN. The intelligence built into the hosted solution is automatically transferred to the local side, including settings and registrations. The Ingate CPE keeps track of updates and stores a mirror image of user settings.

In the event that contact with the hosted server is lost, the Ingate CPE will automatically detect the failure and make use of stored settings to ensure that communication methods are maintained. By using the mirror image of user settings, local calls within the domain are handled by the CPE. Any calls made to the outside will automatically be routed towards the SIP/PSTN gateway, given that there is SIP/PSTN gateway on the LAN. The CPE will automatically detect when the communication link is restored.

Ingate VoIP Survival is available now for all Ingate Firewall and SIParator products.

http://www.ingate.com
http://www.broadsoft.com



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Yankee Group Survey Reveals Contact Center Reluctance

Yankee Group (news - alert) announced that contact centers have been cautious in their adoption of VoIP technology. In cases where penetration has occurred, the switch has been driven solely by application needs and, in isolated cases, reduced infrastructure costs. These results prove the lower total cost of ownership (TCO) metric is not driving migration of circuit-switched agent stations to voice over IP (VoIP).

The culmination of research conducted with more than 1,000 contact center managers, the report entitled *VoIP in Contact Centers Is Inevitable But Not Imminent* highlights five current "use cases" where contact center managers are seriously weighing VoIP deployments. SMB contact centers emerged as the clear early front-runners in adopting packet switched infrastructures.

"Contact centers are reluctant to consider VoIP unless there is a clear use case and ROI", according to Art Schoeller, senior analyst at Yankee Group. "Although the SMB sector is where the significant volume has been through 2004, larger centers are seeing advantages for distributed operations because the technology facilitates increased agent utilization over traditional circuit switching."

VoIP technology creates more efficient data routes to seize full advantage of dispersed agents. Contact centers with packet switched technology could generate cost savings of seven percent from additional utilization.

http://www.yankeegroup.com

Netrake Launches VoIP And IP Multimedia Call Center Solution

Netrake (news - alert) recently announced the general availability of the Netrake Call Center Solution, an session control offer designed to accelerate the call center industry's transition to next-generation IP communications. Netrake's Call Center Solution is designed to afford call center companies lower cost connectivity; enhanced security; efficient call routing; access to advanced VoIP features; and more seamless voice, video, and data integration for real-time customer interactions.

West Corporation, the third-largest U.S. call center services provider, and Genesys Conferencing, the application service provider subsidiary of Genesys S.A., are among the first providers selecting the call center functionality of Netrake's nCite Session Controller platform.

Netrake's nCite Session Controller capabilities are well-suited to the unique needs of the call center industry:

Connectivity Savings: The Netrake Call Center Solution empowers call centers to gain immediate and substantial toll-charge savings. These savings result from replacing costly public switched telephone network (PSTN) connections with secure, reliable IP-to-IP connectivity among all call center customers.

Enhanced Security: The security functions of Netrake's solution address the mission-critical nature of enterprise communications and data typically carried by call center and conferencing services. nCite security features include Network Address Translation (NAT)/Firewall Traversal, as well as denial of service (DOS) attack prevention through deep packet inspection at both the SIP signaling and VoIP media layers.

Inter-working VoIP Signaling: nCite performs real-time conversion between SIP-based and H.323-based VoIP sessions. This function is designed to deliver transparent inter-working among any combination of SIP-based and H.323-based networks operated by call center peer networks.

Scalable Quality Control: nCite partitions each customer's Quality of Service data collection and captures inter-departmental bill-back and service level agreement (SLA) enforcement metrics on a per-session basis. Netrake's nCite DE Session Controller enables this benefit at tier 1 carrier scale — reportedly supporting up to 42,000 secure, concurrent G.729 speech coded calls in a single platform. Customers also can employ nCite SE Session Controllers across smaller distributed points of presence. Additionally, nCite Session Controllers can partition each customer into a virtual domain, enabling discrete service, policy and trouble management per customer.

"Netrake has taken the lead in solving crucial cost, security, quality and inter-working barriers to the call center industry's transition into next-generation IP communications," said Bruce Hill, president and CEO of Netrake.

http://www.netrake.com

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Telrex Calls On Ingram Micro To Dial Up VoIP Resellers

Telrex (news - alert) has announced a referral agreement with Ingram Micro, the world's largest technology distributor, to offer CallRex IP call recording software to the distributor's resellers in the United States.

"We're delighted to be Ingram Micro's only packet-based call recording solution," said Bob Cordes, director of marketing at Telrex. "Ingram Micro's resellers now can offer an affordable packet-based IP call-recording solution to their small and medium business customers. Since CallRex is entirely software-based, it is 40 to 50 percent less expensive than solutions requiring third-party telephony hardware."

"This agreement further illustrates Ingram Micro's aggressive push into voice over IP," said Justin Crotty, vice president channel marketing, Ingram Micro North America. "Our solution providers are continually looking to add to their revenue stream through adopting new markets and new solutions - like CallRex IP call-recording software. Telrex's VoIP product offers just such an opportunity with the additional benefits of high margins, easy installation and post-sale support."

http://www.telrex.com/callrex.htm
http://www.ingrammicro.com

EADS Signs Partner To Resell VoIP, Contact Center Solutions

EADS (news - alert) has announced that Connect-tek is the latest company to join the EADS Certified Partner Program. This recent addition to EADS' channel further expands its presence in the marketplace, with Connect-tek customers in the Pennsylvania area. As a value-added reseller for EADS, Connect-tek will offer Nexspan, the company's sixth generation VoIP solution, Unified Communication Platform, and Centergy Contact Center solutions to address the needs of small to medium-sized enterprises delivering traditional PBX, converged systems or full VoIP capability.

EADS and Connect-tek will target Connect-tek's existing customer base as well as focus on new opportunities.

http://www.eadstelecom-na.com
http://www.goconnect.com

Comdial And Tech Data U.S. Establish Distribution Agreement

Comdial (quote - news - alert) and Tech Data (quote - news - alert) have announced an agreement expanding Comdial's distribution and extending Tech Data's telephony product offering, allowing both companies to capitalize on growth opportunities in the IP telephony market. Comdial products offered through the new agreement with Tech Data meet a variety of business requirements. One such product is Comdial's CONVERSip MP1000 Media Platform, designed exclusively for small and midsize enterprise VoIP deployments.

Through its Telephony Specialized Business Unit (SBU), Tech Data will begin distributing the MP1000 and other Comdial products to U.S. resellers this month. The SBU is focused on supporting the needs of value-added resellers targeting the telephony market.

http://www.comdial.com
http://www.techdata.com

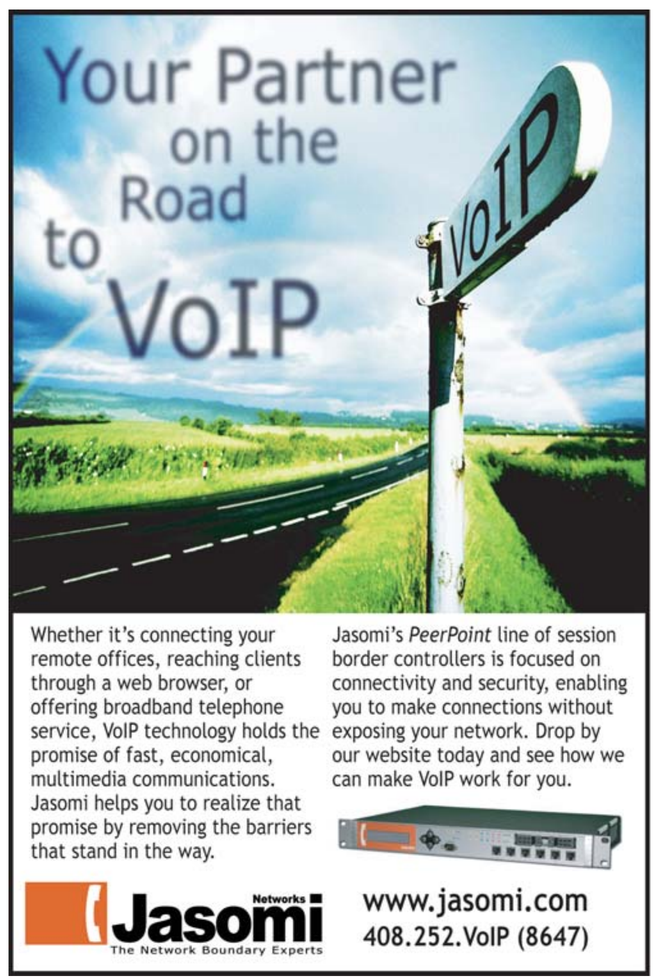
Alliance Systems, tekVizion Sign Joint Services Agreement

Alliance Systems (quote - news - alert) and tekVizion PVS, Inc.(news - alert), announced they have signed a joint services agreement intended to help smooth service providers' transition to reliable converged networks. By working together, Alliance and tekVizion offer completely integrated, tested, and supported solutions including servers, software applications and middleware from leading independent software vendors (ISVs).

The two parties have agreed to work together to create market opportunities for each other's products and services and to further expand their portfolio of shared vendor relationships with market-leading ISVs.

tekVizion offers specialized professional services including consulting, custom software and portal development, integration, support, and test services that include the tekVizion Labs(TM) testing and certification facility. Alliance Systems is an original design manufacturer (ODM) serving ISVs and original equipment manufacturers (OEMs) by providing value-added services including the design, build, ship, and support of Intel server technology.

http://www.alliancesystems.com
http://www.tekvizion.com



By Marc Robins

Notes From The Field

It's high season for industry trade shows — with VoiceCon, Internet Telephony Conference and EXPO, and VON, to name a few — hitting one after the other. I've been taking full advantage of these events to suss out the pulse of the industry and explore the nooks and crannies of the exhibit halls looking for the innovative new products, services, and applications that serve to define the current state-of-the-art in IP communications. As I've strolled down the exhibit floor aisles, walked the hallways of hotel meeting places and convention centers, and (thankfully) rested during strings of pressroom meetings, I've also had the opportunity to engage in entertaining debates about trends and gather a variety of info nuggets along the way.

Judging from the overall attendance at the shows, interest in VoIP (define - news - alert) and IP communications is at an all-time high. While the turnout by service providers (and vendors offering solutions for providers) was significant at both these events, it was also clear that the growing adoption of solutions by the enterprise market is dominating the scene. The small business market — underserved in the past — has also started to gain some real steam. Rumor has it that Cisco is now selling around 2,000 Call Manager Express systems every month, and other vendors not currently selling to that segment have taken notice — Mitel (news - alert), for example, will be coming out with a small biz system shortly.

Digium(news - alert), the people responsible for bringing the free open source phenom product Asterisk to market, were also deluged with throngs of fans checking out the 15 or so partners displaying their wares at the "Asterisk Marketplace" at VON. Coming soon from Digium is an Asterisk Business package — a version that includes training and implementation support a la Red Hat — that might answer the question about how Digium plans to make money from its product. Apparently the company isn't having a hard time developing a loyal and quite substantial base: At a recent user conference hosted by the company, more than 10,000 supporters showed up! A bunch of fans even showed up at Mark Spencer's house recently, (the brilliant, young developer of Asterisk), with a hot tub in tow to bestow upon him!

Judging from the overall attendance at the shows, interest in VoIP and IP communications is at an all-time high.

Based on what I saw during a demonstration of some new applications from service provider and app developer Teleo, the age of true Web "click-to-call" may be upon us. Teleo, run by president/CEO Peter Sisson, has developed a slick SIP-based toolbar app that mates with Internet Explorer that finds and highlights any phone number on any Web site, making it 'click-to-talkable.' Go to Google to search for pizzerias in your hometown, and with Teleo, you can seamlessly dial the numbers that appear on the screen. Shop on eBay, and you can call the buyer to ask questions right from the phone number listed. Teleo hosts the service beyond the scenes, providing a free 30-day trial, after which the company charges $29.70 for six months of service (comes to around $4.95/month). You get a personal phone number, unlimited inbound calls, and free peer to peer calling. You can also make calls to most TDM phones worldwide for two cents/minute (except cells and some countries) and can make calls from a circuit switched phone into the network for the same price. The company is only a few months old, and so far has users in 90 countries and is logging about 150K minutes/month. I've always been a big believer in this type of application, but for one reason or another it never took off. With Teleo, I think click-to-call may finally have found a company that has the chops to make it stick.

Also seen during my wanderings include some neat new endpoints from Mitel, including their Navigator "phone strip" that creates a whole new form factor for the office phone set, and a new WiFi base station/NIC base for their IP phones that allow for wireless add/moves/changes — ideal for retail and other establishments that desire the flexibility that wireless provides. **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 24 years. For more information, call RCG at 718-548-7245 or e-mail robinsconsult@optonline.net.

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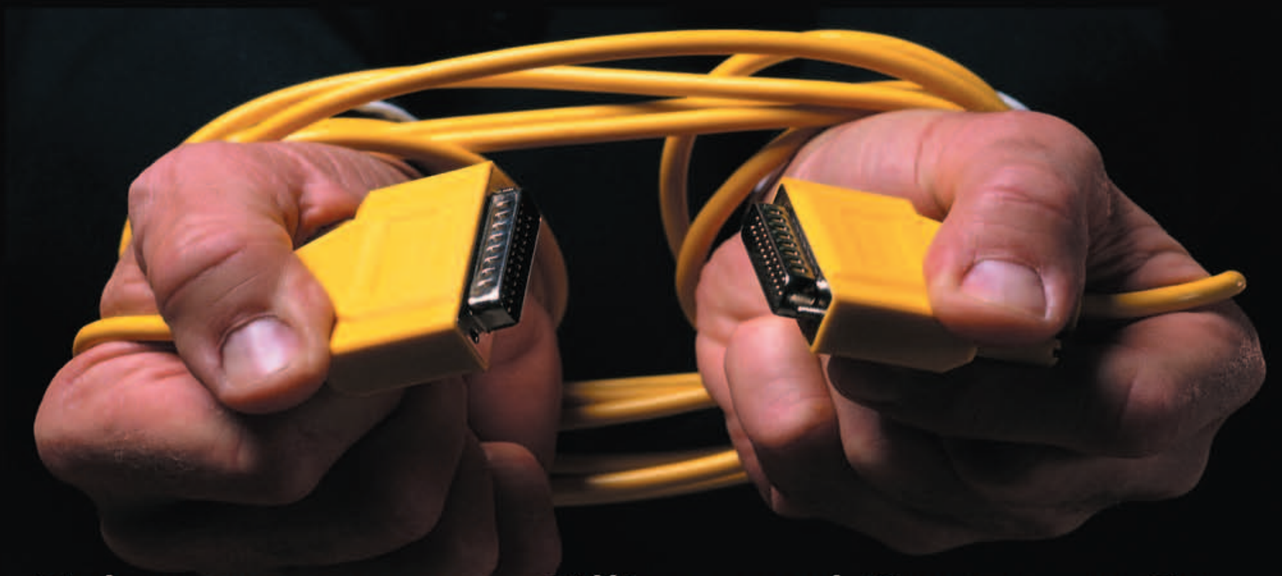

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By Tony Rybczynski

The Path To Business Continuity

"Do more with less." "Make your employees more productive." "Grow your revenue base." Underlying these CIO concerns, and in some cases the subject of regulatory requirements, is business continuity and disaster recovery after natural and man-made disasters, such as the 2003 north-east blackout, the SARS outbreak, and escalating security attacks. Business continuity ensures mission-critical services and applications will continue to operate in the event of disaster, failures or outages, and as necessary, reestablishes full functioning as swiftly and smoothly as possible.

Convergence enables the virtual enterprise with the convergence of protocols and networks, communications and applications, and the dissolution of distance across the enterprise. The virtual enterprise in turn enables profoundly more comprehensive business continuity approaches, that go beyond the data center to include telephony (and unified communications systems in general), branch operation and even employees.

Business Continuity For Telephony

Traditional telephony has been deployed on a site by site basis. Disaster recovery requires that a total replacement switch (i.e., a key system or PBX) be deployed, a process that can take days for a large installation. The wide-scale acceptance of IP telephony has for the first time created the opportunity to economically address business continuity requirements for telephony systems, by disaggregating the function of call control from switching. The simplest approach is basically a cold standby approach whereby IP phones register with a standby pre-configured communications server if the primary system fails. Another utilizes load sharing across distributed servers (e.g., at a central site and at a data center), through database replication, providing total recovery from failures within minutes. The ultimate approach supports real-time mirroring with sub-second recovery times. These solutions are all being delivered today.

Keeping Branches Running

Centralization of IT resources has been the path to lowest total cost of ownership (TCO), providing higher utilization of servers and reducing server costs, reducing operations costs and providing increased agility to accommodate business needs. IP telephony opens the door to leverage the opportunities of centralization, previously only economical in the data center. However, loss of network connectivity between a branch office and a central site will result in loss of dial-tone for branch users (an unacceptable situation). Branch business continuity leverages survivable gateways to ensure continued featured telephony operation, including local public network access. This provides the right balance between IP telephony costs and the costs of enhancing network reliability, while meeting business continuity needs.

Employee Business Continuity

IP telephony has also disaggregated the users from the site architecture where they were previously wired to the PBX. When disaster hits, it is imperative to allow critical resources to continue to perform their duties. For example, after the north-east blackout of 2003, the Ontario government asked employers to close down their facilities while the nuclear power plants were brought back up on line. During this one-week period, I and many of my co-workers were able to continue to be productive, using our laptops configured with IPSec VPN (define - news - alert), IP telephony and multimedia clients. Leveraging IP telephony in contact centers also allows agents to be quickly relocated in case of disasters. IP telephony systems running on WLANs and wireless mesh networks can also be important tools for emergency response teams. The virtualized enterprise can respond much more quickly when disaster strikes, while the proliferation of end user devices is providing important business continuity options.

Business Continuity And The Converged Network

The business continuity imperatives as they relate to the converged network are recovery from failures in human time (i.e., tens of msecs), gradual degradation under failure conditions, continued operation under overload, and load sharing across all network resources. Additionally, the business continuity imperatives from a security perspective are to proactively block security threats before they penetrate the network and to rapidly isolate any infiltrations that take place. This requires acquisition of information on potential threats, assessment of these threats, and rapid action to block and/or contain these threats. Multilayer security implies security in the very DNA of the network, including user authentication and admission controls at every wired and wireless port and centralized security intelligence to ensure consistent application of enterprise security policies.

Business Continuity is one of the most critical issues enterprises face. Convergence enables business continuity to go beyond specific focused back-up plans for data centers, to become part of the fabric of communications across the enterprise, cost effectively empowering individuals and the business overall to continue to operate virtually. For the CXO, having a business continuity strategy across these domains can minimize business loss, safeguard the enterprise's reputation with its customers, partners, and employees, and allow IT to focus its investments moving the business forward. **IT**

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

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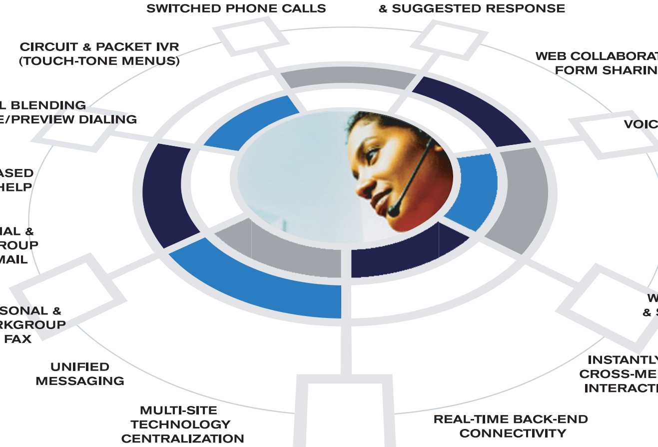

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By William B. Wilhelm, Jr., Esq.

The New Chairman Of Change

FCC Chairman Powell is leaving and much remains to be done.

Chairman Powell is gone. Responsible for guiding the Commission through the Pulver and Vonage preemption orders — the ex Chairman embarked the FCC on a voyage that, as of yet, is incomplete. While many challenges await the new FCC Chairman, the issues surrounding broadband, VoIP, and IP enabled services must remain at the top of the Commission's 2005 agenda.

First among the new Chairman's priorities should be to find that VoIP (define - news - alert) applications are information services. While the FCC (news - alert) resolved this issue with respect to the computer-to-computer services at issue in the Pulver docket, the FCC had not explored the classification issues in a comprehensive fashion. Despite the FCC's inaction, the 8th Circuit recently affirmed the earlier decision of a lower federal court that found Vonage's service is an information service not subject to regulation by the Minnesota Public Service Commission. At this point, delay is unwarranted. In order to permit the development of the full potential for this remarkable technology, as well as other IP-enabled services, the FCC should resolve it's pending IP-Enabled Networks Proceeding to affirm what has already been decided by the courts — that VoIP applications that perform net protocol conversions are non-regulated information services.

Next, the FCC should act promptly to preserve a national regulatory regime for VoIP services. The FCC in November correctly found that one form of VoIP is an inherently interstate service-not subject to state regulation in part because VoIP service is portable in that the subscriber and the VoIP telephone number are not tied to any specific location. This decision will help assure that many new VoIP providers are not unnecessarily and counterproductively subject to legacy telephone regulation. Nonetheless, state commissions are attempting to overturn these decisions. California, Minnesota, Ohio, and New York recently filed appeals of the FCC Order. The FCC, in cooperation with the VoIP industry, should vigorously defend its November decision and additionally reach out to state regulators and work with them to realize the benefits that will be provided to consumers and businesses if VoIP is permitted to grow subject to a national, deregulated framework instead of a patchwork of disparate state regulation. If necessary, the new Chairman should encourage Congress to enact legislation such as the VoIP Regulatory Freedom Act previously introduced by Senator Sununu and Representative Pickering.

Although it's not going to be easy — the new Chairman must also take bold steps to fix the existing intercarrier compensation framework. Everyone agrees that the current regime is broken. Various key industry players have submitted proposals to the FCC for comprehensive reform. These proposals vary from retaining but rationalizing pricing to abolishing intercarrier compensation payments altogether. The FCC just initiated yet another proceeding seeking comment on reform. One thing is clear, however: it makes no sense to apply a broken system to new VoIP providers.

The new Chairman must further the Commission's focus on homeland security and work with the industry to facilitate industry solutions for E-911 and disability access. VoIP providers have strong marketplace incentives to make their products as attractive and valuable as possible for consumers and businesses. Government policy should be directed towards letting industry work out solutions to key issues including 911 access, rather than assuming that approaches that have worked in the context of the traditional telephone network should be applied to VoIP. VoIP providers are working hard to craft solutions that preserve the new features and service options while also meeting important public policy goals. Leading VoIP providers, state PSAPs and NENA are working to provide 911 access that provides functionality comparable to the traditional telephone network. VoIP providers are also crafting solutions to improve access by disabled persons and in fact, the inherent advantages of VoIP, such as the greater ability to convert IP encoded voice messages to text, facilitate disabilities access. Rather than imposing specific regulatory requirements on VoIP providers, the FCC should ensure that VoIP providers are permitted reasonable access to the underlying telecommunications infrastructure in order to improve upon the transmission and delivery of 911 calls.

Finally, the FCC must take clear steps to ensure a right to consumer access to IP applications over all broadband platforms. Chairman Powell got it right when he stated that in order to realize the full potential of IP-enabled communications, regulators "must ensure that a willing provider can reach a willing consumer over the broadband connection." Nothing could harm consumer choice or the growth and development of new broadband communications applications more than if network providers were able to block access to the customer's choice of application or computing device. It is a testament to the FCC's strong leadership that no significant problems have developed thus far. Clearly, however, the new Chairman needs to continue to articulate its policy in this area as well as remain vigilant and proactive in the event abusive or anti-competitive practices become apparent. IT

William B. Wilhelm is a Partner in the firm of Swidler Berlin Shereff Friedman, LLP. For more information, please visit http://www.swidlaw.com. The preceding represents the views of the author only and does not necessarily represent the views of Swidler Berlin Shereff Friedman, LLP or its clients.

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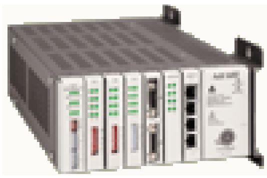

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By Hunter Newby

Making Sense Of VoIP Peering

VoIP Peering is becoming commonly known as the process of interconnecting two or more VoIP networks or devices from the physical layer (see the chart below) all the way up to the application layer, including many layers in between. Specifically, VoIP Peering can occur at layers 1, 2, 3, 5 and 7. It can get so confusing it's almost like giving New York City subway directions to a tourist.

The confusion gets amplified when we discuss the financial aspects of billing the calls. Some say VoIP (define - news - alert) Peering is a way to enable "free calling" — no matter whom or where you call or for how long, you pay one flat rate. Others say billing will be the same way as it has been for decades, per minute with higher rates for certain remote call destinations. As it is with almost everything else, the truth lies somewhere in the middle. Understanding the necessary infrastructure of VoIP Peering may shed light on how to drive the future economics of it.

VoIP Peering combines the best of two worlds: the telephone system and Internet Protocol with a little help from its friends Ethernet, ENUM (define - news - alert), and SIP (define - news - alert). Depending on how you view the world (from the application layer down, or the ground up), each component is essential. If any one particular layer is removed, the full suite of VoIP Peering capabilities wouldn't be possible.

The different layers, 1, 2, 3, 5 and 7, relate to physical layer cross connections, Ethernet, IP (the Internet and private IP), ENUM, and SIP respectively. Depending on the situation, various combinations of these components can produce what is known as VoIP Peering. The two major components of VoIP peering that actually enable VoIP devices to peer are ENUM and SIP. ENUM works together with SIP to resolve number locations and can operate over the Internet, or a private IP network. SIP can be used by itself to interconnect gateways, or switches over the Internet, or a private IP network.

Each layer and its function have different breeds of new service providers with different rates and connection methods. ENUM is basically for carriers servicing end users (LEC's and VoBB (define - news - alert) service providers) and entities that use blocks of numbers (enterprises, government, universities). SIP is mainly a carrier to carrier VoIP switch enabler, but can be used for IP PBX's to communicate with carriers or other IP PBX's directly.

VoIP Peering combines the best of two worlds: the telephone system and Internet Protocol with a little help from its friends Ethernet, ENUM, and SIP.

The economic functions of ENUM and SIP vary as well. ENUM service providers available today do not charge any per call or per minute fee. ENUM and the Layer 5 specific service providers are essentially enabling free calls for all of their users. (By the way, that sucking sound you hear is the bottom dropping out of the PSTN.)

The CLEC (define - news - alert) and VoBB service providers using ENUM are either allowing their customers to peer or communicate directly, with no metered charge because they have figured out a (profitable) way to charge a flat rate for access to their network, or they are still billing the customer per minute and pocketing the savings. Either way the calls are free — it just depends for whom. Their customers are end users or enterprises that don't want to (or can't deal with) managing their own VoIP infrastructure.

The enterprise, government, and university segments that use ENUM today do so either via the Internet or a private voice Internet. They pay for their own access, run their own SIP gateways and query a central ENUM registry for every call they make. They take on more risk but get greater rewards. Every call that can be peered with another on-net device is free. The more peered calls, and resulting ENUM match-ups, the better it is for those who are already using it.

ENUM VoIP Peering Is Really A New Business Model For Voice

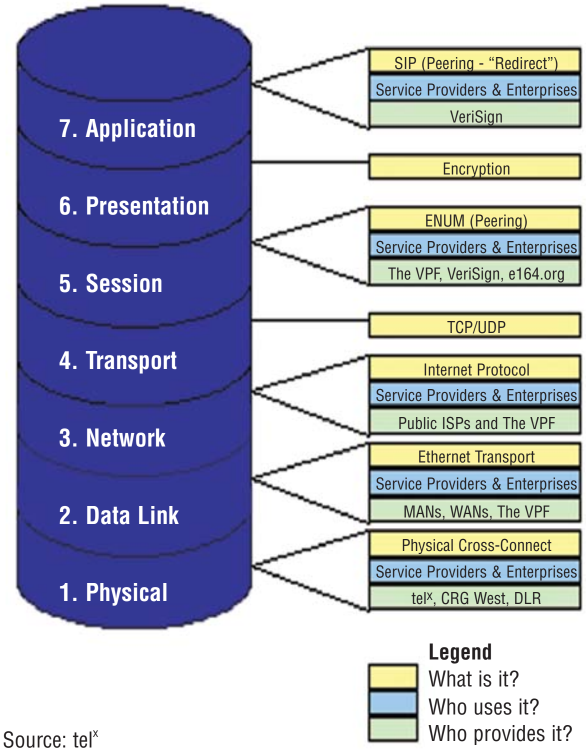
SIP is not really a standalone service yet but rather a way for VoIP switches to find each other. SIP look-ups can become (and in some form, already are) a standalone service. SIP is very popular with the international wholesale minutes providers. Those VoIP switch operators using SIP have destination routes they want to sell. They make their money on selling minutes and getting paid per minute. These are typically international destinations because there is still some margin left in that end of the business, but SIP connections are also used for setting up PRI DID circuits for local numbers over IP. That's how you can get a number with a New York area code in Japan. The buyers of these services via SIP include other carriers, hosted/outsourced IP PBX providers, VoIP ASP's, calling card platforms, enterprises, government entities, and universities.

SIP VoIP Peering Is Really A New Way To Connect Two Voice Switches

IP PBXs using SIP as a means to communicate directly

Go To Table of Contents | Go To Ad Index

The OSI Model For VoIP Peering



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facilitate ENUM, or end user device direct connections. Essentially they are building their own private VoIP networks. Initially the enterprises, government, and university groups will interconnect their own nodes. Then they will look to directly interconnect with each other. That is when the combination of SIP and ENUM deliver the maximum value of VoIP Peering.

As VoIP Peering gains in popularity, it is exposed to the risk of being too widely defined and simultaneously lacking specificity. This is primarily because VoIP Peering can mean different things to different people. In some instances, these broad strokes are socialized by those who wish to keep the "old way" alive in some fashion so as to gracefully engineer themselves out of a disadvantageous pricing model and billing system. For that reason, during this initial growth period, it is important to insure that each style of VoIP Peering, the user type, method and associated economics, gets clearly defined and equal coverage. Whether you represent a carrier or an enterprise, accurately weighing risk tolerance against the bottom line to find a balance is the only way to determine your path. If you are equipped with a good knowledge base, you will be able to ask informed questions and get accurate answers. IT

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

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By Ben Guderian

Is Your Network Ready For Wi-Fi Telephony?

Having a Wi-Fi (define - news - alert) network deployed in the enterprise doesn't necessarily mean a company is ready to start leveraging that wireless investment for voice applications. Wi-Fi telephony is a pretty demanding wireless application, combining the interoperability and performance requirements of both enterprise VoIP and real-time Wi-Fi networking. The good news is that Wi-Fi telephony is working well today in a wide variety of enterprise applications, so the products and deployment best practices already are available. But similar to any other leading-edge technology, there are several recommendations that could better prepare your company to run voice and data over a shared Wi-Fi infrastructure.

One of the first issues to consider before adding voice to an existing Wi-Fi network is coverage. A lot of enterprise Wi-Fi deployments today were built around the primary objective of providing network access to employees in common areas like conference rooms and cafeterias. That's fine for laptop users, who only need access when they are sitting in a meeting or taking a break. But Wi-Fi telephony users aren't going to stop in one place to talk on the phone, nor are they going to put up with their wireless telephones only working in certain parts of the facility. A Wi-Fi network designed for voice applications has to provide comprehensive coverage through all areas where employees can be expected to roam, and maybe even in some unexpected places. This may increase the cost of deploying a wireless LAN, but the benefit of making employees more mobile, responsive, and productive will outweigh the cost.

The next thing to look at is the shape of the overall IP network. Wi-Fi telephony is just a wireless version of enterprise VoIP, so it is important to make sure that the wired network infrastructure also is ready for IP telephony. Wi-Fi telephones have the same tolerance for latency and jitter as wired VoIP phones, so if the wired infrastructure can't support wired VoIP phones, it is probably insufficient for Wi-Fi telephony as well. Most enterprise telephone system providers offer services to audit networks for VoIP applications and can make recommendations for system upgrades.

Wi-Fi telephony is working well today in a wide variety of enterprise applications, so the products and deployment best practices already are available.

Are All APs Created Equal?

The fast pace of innovation in Wi-Fi technology has led to improvements in performance and lowered the cost of installing and maintaining wireless LANs. But not all Wi-Fi access points (APs) are the same in terms of how well they support voice applications. The Wi-Fi AP market has divided into two distinct segments: low-cost APs for residential use and high-feature APs for enterprise use. Low-cost residential APs typically don't offer any quality of service (QoS) (define - news - alert) capability to ensure that voice packets get through the network with minimal delay. They also don't provide many of the security, maintenance and administration features critical to large-scale enterprise applications. For that reason alone most enterprise IT managers don't consider residential-grade APs for their facilities, even though they may cost significantly less.

Although all enterprise-grade APs are designed to serve the same basic functionality, there are still areas where individual vendors can innovate and differentiate. Several vendors have specifically targeted Wi-Fi telephony applications and offer unique features or higher performance for voice applications. But without doing a rigorous evaluation or relying upon credible independent third-party test results, it is difficult to know whether or not a particular product is right for any given application.

Interoperability between Wi-Fi infrastructure and wireless telephone devices is not a black and white proposition. Something that may work well in a laboratory test environment may not work as well on a large network with variables such as roaming and data traffic thrown into the mix. The Wi-Fi Alliance has formed a task group to look into the requirements for Wi-Fi telephony interoperability testing, but a test program is not yet available. Some Wi-Fi equipment vendors are taking it upon themselves to provide interoperability testing and verification.

Good For Data, Bad For Voice

The biggest impediment to widespread enterprise adoption of Wi-Fi has been security concerns. Just when the market was really heating up a few years ago, there were reports that standards-based Wi-Fi security was insufficient for enterprise use. Fortunately the industry came together to develop new standards that met enterprise security requirements for protecting wireless networks from eavesdropping and unauthorized access. But these data-oriented security standards can cause

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Go To Table of Contents | Go To Ad Index

problems for real-time applications like Wi-Fi telephony, particularly as users roam between APs.

Every time a wireless client device tries to connect to a different AP, it needs to go through a process of authentication or verification that it is authorized to connect to the network. This is the same process used when connecting our PCs to a wired LAN or through a VPN connection. The user credentials — username and password, depending on the security level — are verified against a security database. If everything checks out, the client device is given access to the AP and can start transmitting and receiving data. This process can take a tenth of a second or possibly as long as a few seconds depending on the network security infrastructure and number of devices trying to authenticate at the same time. For a data application, an interruption of a second or two won't be noticeable to the user. For a real-time voice application, a second or two of "dead air" is not only annoying, but also can mean losing an important word or syllable in a conversation. A Wi-Fi telephone user might change APs and have to re-authenticate several times just while walking down a hallway or staircase. Therefore, a more efficient AP authentication method is required for enterprise-grade Wi-Fi telephony.

Voice-friendly Authentication

There are several solutions to maintaining a high level of Wi-Fi security without sacrificing voice quality in enterprise applications. First is using different security mechanisms for voice and data applications. This approach is based on the premise that less stringent security is acceptable for voice applications, as long as voice devices and servers are segmented into a separate network. Many Wi-Fi APs support virtual LANs (VLANs), which allow different networks to be set up using the same infrastructure. In this scenario, voice clients can operate on a VLAN (define - news - alert) that does not require re-authentication every time a user roams to another AP, while data devices can be on a separate VLAN with full re-authentication required.

Several proprietary solutions are also available to maintain security without jeopardizing voice quality. Depending on the approach, some proprietary solutions may require cooperation between the Wi-Fi infrastructure vendor and the Wi-Fi client vendor to implement the same protocols.

Lastly, Wi-Fi security standards are available today that facilitate Wi-Fi telephony with even more robust solutions in development. The Wi-Fi Alliance's Wi-Fi Protected Access (WPA)(define - news - alert) standard is supported by most Wi-Fi vendors today, and it offers two different modes of authentication: enterprise mode and personal mode. Enterprise mode uses the same data application-oriented approach as previously described. Personal mode provides authentication without requiring a network security database, making it simpler to implement and easier on real-time appli-

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cations like voice. Personal mode uses a "pre-shared key," or common password, entered into all devices authorized to use the Wi-Fi network. The pre-shared key is never transmitted over the air, so there is no risk of it being intercepted. When a wireless client attempts to access an AP, the two devices verify that they share the same key and use it to generate their session encryption keys. However, a drawback with personal mode authentication is that the pre-shared key must be entered into each device, which makes it difficult to use in large-scale Wi-Fi deployments.

Just as it did with the original Wi-Fi security issue, the industry is working on a standardized solution for meeting enterprise requirements for voice quality and secure roaming. The IEEE 802.11 standards body has formed Task Group R specifically to address this issue. A forthcoming 802.11r standard will provide a secure roaming protocol that will eliminate the need for proprietary solutions or compromises.

Wi-Fi technology is rapidly maturing and vendors now recognize voice as a key application for enterprise use.

Deploying The Voice-ready Wi-Fi Network

In summary, implementing Wi-Fi telephony today is not as complex as it may sound. Sure, there are some things to watch out for like covering all the nooks and crannies where users may roam and determining how to deal with security. But Wi-Fi technology is rapidly maturing and vendors now recognize voice as a key application for enterprise use. Standards to address voice-specific issues such as secure roaming are on the way — although the standards process never seems to move fast enough. Finally, Wi-Fi telephony vendors see the value in working together to guarantee not just interoperability, but also to provide multi-vendor solutions that meet all of the implementation and performance criteria of enterprise users. IT

Ben Guderian is director of market strategy at SpectraLink Corp. For more information, please visit the company online at http://www.spectralink.com.

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By Dan Dearing & Michael Khalilian

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IMS And The Challenges Of Convergence

In the not-so-distant past carriers had ample time to develop and deploy new technologies and services for their customers using the PSTN; however in today's world, with the advent of the Internet and increased competition among providers, carriers face significantly compressed timelines to react and roll out new and differentiating services. Few would argue that service velocity, the ability to move quickly to deliver new services, is crucial to compete, and ultimately survive in today's competitive environment. The IP Multimedia Subsystem (IMS) architecture is emerging as the service delivery framework offering service providers a robust and flexible method for introducing new services and applications.

However, there may be some pros and cons that carriers must consider when deploying IMS. One of the biggest pros is that the IMS framework allows carriers to build a single IP network infrastructure that enables maximum service velocity for any real-time service. This approach gives carriers the potential to realize enormous CAPEX (define - news - alert) and OPEX (define - news - alert) savings because they can reduce the amount of equipment in the network, reduce the number of people needed to support the network, and respond more quickly to new business opportunities with revenue generating applications. In addition, larger carriers that have both wireline and wireless assets can use the IMS framework to converge their businesses into one cohesive network.

IMS allows carriers to build a single IP network infrastructure that enables maximum service velocity for any real-time service.

On the flip side, the IMS framework does create some challenges. Carriers want to leverage user mobility and the flexibility of IMS but they need to also provide their customers with PSTN-like reliability, availability, and quality. There is no "network busy" in the Internet. This presents a dilemma for many carriers since service assurance within this IP environment is a chief concern for their users. VoIP and other real-time IP services present new and unique challenges for carriers. The use of a single network to support multiple applications creates opportunities for resource contention. QoS for latency-sensitive applications, such as VoIP, is impacted by limited bandwidth in the access network, device over-utilization in the services layer (e.g., media gateways, applications servers) and congestion conditions in the network core due to outages. User mobility and the multi-service nature of enterprise applications complicates the carrier's task of meeting these challenges as users expect service flexibility and reliability at work, home and on the road.

So how do carriers control and manage VoIP and other forms of real-time IP traffic within the converged IMS?

Today many carriers are deploying VoIP technologies, such as softswitches and session border controllers (SBCs), to create a network overlay that facilitates the transport of voice traffic over their existing packet infrastructure. While softswitch technology provides carriers with a way to bridge between the PSTN and their IP network, the SBC manages and controls VoIP connections to enterprise and consumer customers and service provider partners. The SBC (define - news - alert)provides sophisticated session management — including call admission control, user authentication, and user authorization — needed to manage real-time IP traffic. SBCs are endpoint aware and can determine what network resources are needed for a session based on the user's service subscription, their location, and the type of VoIP endpoint used.

Carriers that have taken the traffic management approach to ensure PSTN-like reliability and predictability for VoIP — having already spent billions of dollars in MPLS (define - news - alert)router technologies to provide deterministic performance for IP services — do not have the session management granularity needed to manage a diverse set of applications within a converged network. Simply put there is no direct interaction between the IMS and the MPLS transport network, so a greater operational burden is being placed on the carrier to properly traffic engineer their

Go To Table of Contents I Go To Ad Index

network.

The operational burden can be eliminated by providing a control link between the SBC and edge MPLS router, so that the SBC can dynamically determine what applications or services a user is requesting and interface with the transport network and services layer to determine whether resources are sufficient to service the call with the needed QoS. In effect, the SBC becomes a microflow manager with the intelligence to configure and manage the forwarding plane of the MPLS layer.

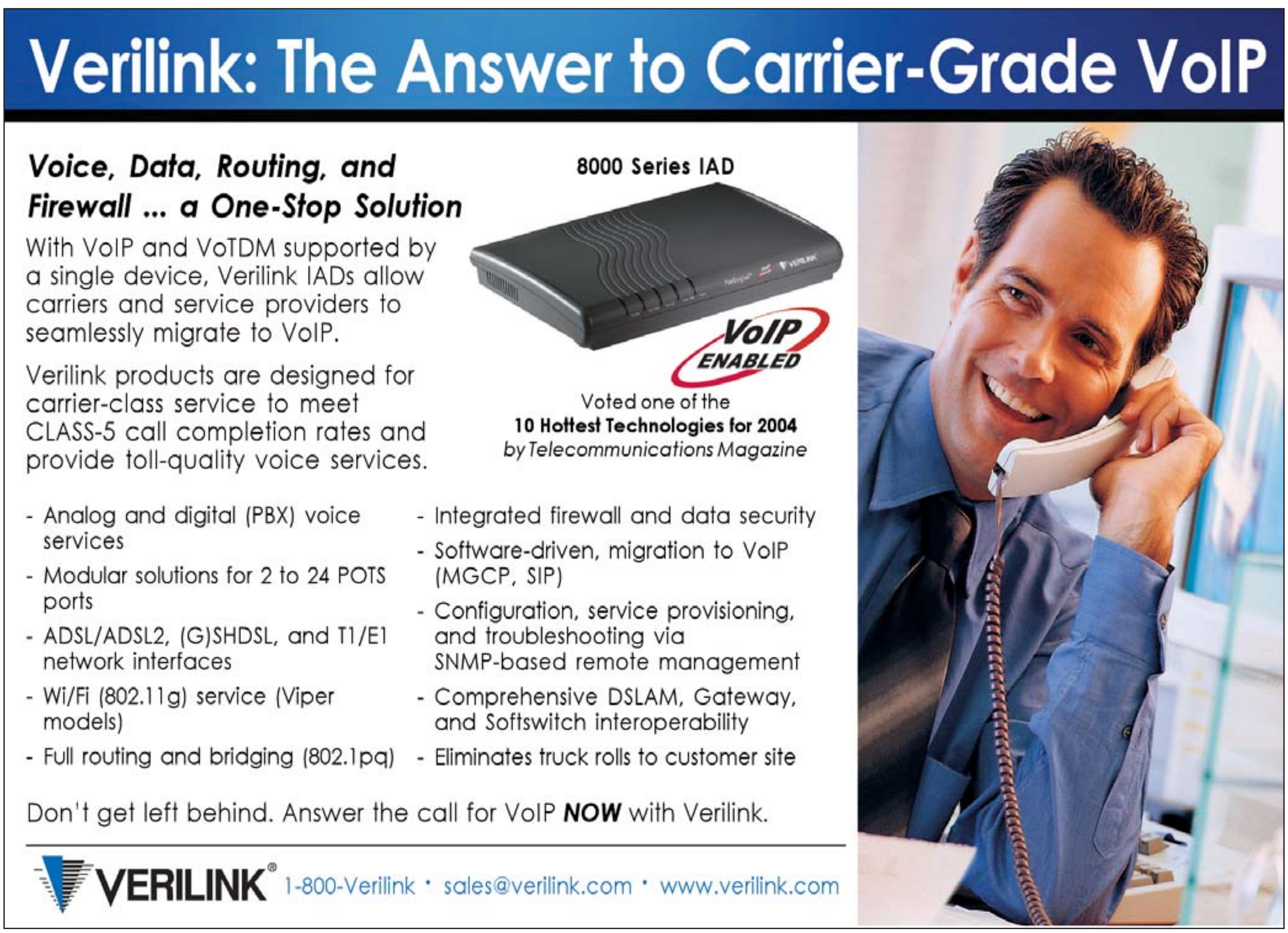
How Does This Work?

In MPLS networks, data transmission occurs on label-switched paths (LSPs). LSP is a virtual circuit over a fully meshed IP/MPLS network. At the ingress of the MPLS network, the edge MPLS router classifies traffic using tagging methods such as Diffserv and VLAN 802.1P to put those packets on traffic engineered paths. Unfortunately carriers are staking their futures on QoS schemes that are not suitable for the large scale and rapid introduction of real-time services for a couple of reasons. First, Diffserv is best suited for a large number of short flows typical of Web traffic. Second, these tagging methods separate traffic in bulk, best-effort and real-time, and cannot distinguish between types of real-time traffic that need different guarantees per flow. In short carriers can have more control over their IP traffic by deploying a hybrid solution.

A hybrid approach utilizes bulk methods of traffic classification to distinguish real-time traffic from Web traffic and employs SBCs for more granular classification of real-time traffic on an endpoint, user, and application basis. Operating within the real-time services signaling layer, the SBC gleans QoS (define - news - alert) information from the SIP signaling stream and used endpoint and user information associated with the user's registration to associate other policies, such as security, with a given session. This information is conveyed to the edge router via a control protocol and enables the router to dynamically and granularly map traffic to LSPs uniquely traffic engineered for each type of real-time session.

In summary, IMS is poised to revolutionize the architecture of the telecommunications industry to use IP to deliver a multitude of real-time services. SBCs, acting as microflow managers, provide carriers with the missing ingredient for implementing the IMS framework with PSTN-like quality, scale, and availability. The good news is today many carriers have already invested in the technology needed to retool their networks for IMS. The new opportunity for carriers, particularly their CEOs, is discovering — and then capitalizing on — the new services and applications available for their customer base. **IT**

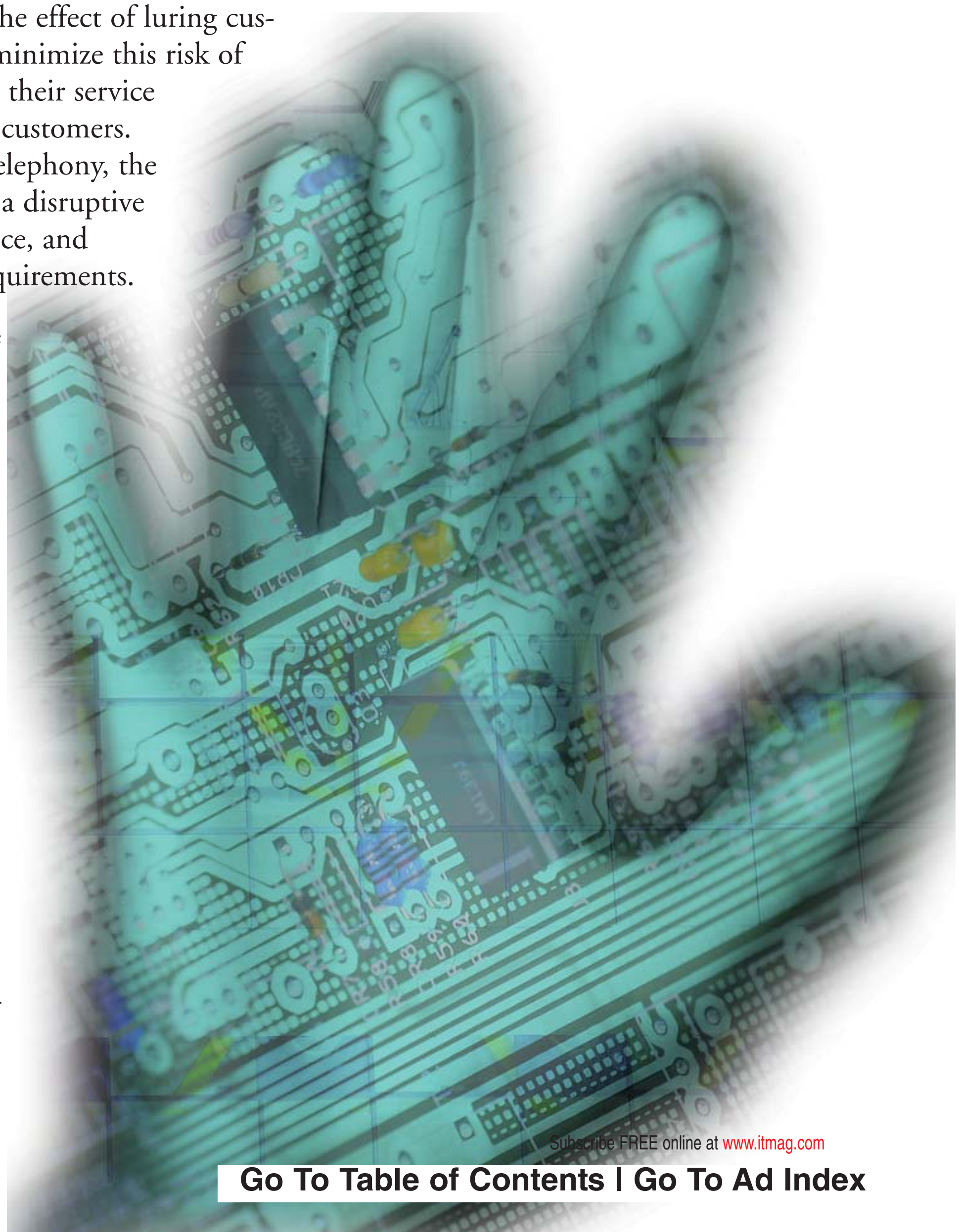
Dan Dearing is vice president of marketing at NexTone. Michael Khalilian is chairman and president of the IPCC, an industry consortium of carriers and solutions providers advancing packet-based communication technologies. For more information please visit http://www.packetcomm.org



Part III: Next Generation Networks: Should You Build Or Buy?

By Shawn Lewis

As the wireless industry has shown, disruptive new technology with better product and service features has the effect of luring customers to regularly change carriers. To minimize this risk of churn, carriers must continually expand their service offering in order to retain their existing customers. With the proven acceptance of packet telephony, the incumbent carriers are again faced with a disruptive technology that has a lower cost of service, and heavy CAPEX (define - news - alert) requirements.



Legacy incumbent service providers have huge capital investments in large class-five equipment geared toward transporting traditional circuit-switched voice calls. They are quickly finding their technology is outdated. The technology they invested in does not address the convergence of voice and data technologies at all, or without significant CAPEX expansion. The market; however, is demanding that they offer new IP-based services. They are therefore forced to invest in and build entirely new, next generation networks, which delays their market entry, or to purchase the services wholesale. By purchasing services wholesale, the capital investment is eliminated, and the service provider can spend its time, energy, and financial resources on acquiring customers and adding profitable revenue, rather than building its own next generation network to gain market share.

There are many benefits to buying next generation network services from wholesale providers versus building your own next generation network.

So, the question is… Should you build or buy your next generation network?

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In this third of a five-part series, we will describe how one answer is a low-cost, high-quality Voice Over Internet Protocol (VoIP) solution that speeds time to market, assures carrier-grade quality, and enables early market entry without the CAPEX requirements associated with building a new network.

CAPEX considerations: Should you invest in facilities now, later, or never?

The Telecommunications Act of 1996 was created to increase competition while expanding technology development at reduced costs for consumers. The competitive drive was to build class-five circuit-switched facilities to enable competition with the Regional Bell Operating Companies (RBOCs). As it turned out, building expensive networks to compete with legacy services was not a cost-effective investment. Several issues plagued these new Competitive Local Exchange Carriers (CLECs):

1) The profit margins did not prove to be very attractive;

2) The equity markets dried up before most CLECs (define - news - alert) completed their network build-out;

3) Winning customers with the same product at a moderately reduced price was not enough, in many cases.

Today, more and more carriers and service providers are focusing on building profitable revenue with VoIP services using wholesale network partners prior to making large CAPEX commitments. With the technology curve rapidly changing, it is difficult at best to manage and finance a large-scale build-out and focus on properly marketing retail services. With numerous wholesale partners available in the market today, the best answer for most service providers is to consider going to market immediately by purchasing the VoIP transport and application services from a wholesale carrier — at least initially. Once they have their product defined and begin a large-scale rollout, they can then analyze whether it makes sense to build new network facilities to address their new customers.

Avoid investing in rapidly outdated technology

To compete long-term in this industry, service providers must be able to react quickly and efficiently to customers' demands for new applications and services. The days of leading with price alone are over. To succeed, service providers must provide the sticky applications that customers are demanding and deliver them as the low-cost leader. Delivering these applications requires technical expertise in software applications.

Transport, enhanced features and application services in the VoIP world are radically different than in the circuit-switched, TDM (define - news - alert) world. The technology landscape is rapidly changing, making many of the technologies outdated before they have provided a positive return on investment. With the driving force behind VoIP being enhanced services and applications such as unified messaging, IP Centrex, and IP PBX, a facilities-based service provider is required to not only invest in the backbone network infrastructure but also the application and feature servers. This is not only a large financial commitment, but it pushes the service provider well outside of its area of core competence.

Invest in marketing to acquire customers first

The days of "build it and they will come" are long gone. Billions of dollars of investment in the telecommunications sector evaporated in the past several years, largely due to fundamentally unsound business strategies. The most successful service providers in the industry share one common mindshare; they invested in marketing a profitable set of products and services to acquire customers rather than building their own infrastructure. We need look no further than to a few key players:

America Online (AOL) (quote - news - alert), Microsoft Network (MSN) and EarthLink abandoned the idea of operating their own network backbone, opting for a wholesale-managed solution early on in their growth phase. The ensuing result is the three service providers quickly became the clear winners from a customer base and a profitability perspective.

Building expensive networks to compete with legacy services was not a cost-effective investment.

VoIP industry and early ISP parallels

We can easily draw parallels between the ISP industry of the late 1990s and the current VoIP marketplace today. There is a mad rush by service providers to fill demand, both real and perceived. Many providers are opting for putting the bare minimum infrastructure in place so they can enter the market with little investment. These providers are plagued with Quality of Service (QoS) issues that often taint customer expectations about VoIP technologies. This scenario is not dissimilar to the early stage ISPs that evolved from being bulletin board operators with banks of low-end analog modems. They quickly either sold off their customer base during the ISP rollup or they simply failed as a business.

At the end of the day, carriers and service providers have to ask themselves whether or not they are better off spending their financial and operational resources on acquiring profitable customers or are they better off building a next generation network and services delivery platform. Doing both is quite a tall order. IT

Shawn Lewis is the CEO of Volo Communications, a wholesale provider of advanced voice and data services and applications including broadband VoIP service. Mr. Lewis also wrote the first two patents for softswitch and media gateway technologies. For more information, please visit http://www.volocommunications.com.


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Boston Ballet Keeps Communications En Pointe

By Richard A. Johnson

Now in its 41st season, Boston Ballet has built an international reputation for its dancing and its education and outreach programs. Under the leadership of Artistic Director Mikko Nissinen, the Company garners outstanding reviews and its annual production of The Nutcracker is one of the most beloved in the world. Boston Ballet School is the largest in North America, serving over 2,000 students in three locations. The Boston Ballet Center for Dance Education produces a range of education and outreach programs, including Citydance, which has brought the joy of dance and movement into Boston public school classrooms for 14 years.

However, pulling off flawless productions requires much more than superior talent and training. Behind the school and every performance, a staff of about 120 handles everything from ticket sales to marketing to costume design. Ensuring that the school and ballet company run smoothly depends on seamless communication with ballet patrons, and among staff, dancers, teachers, and parents. This is particularly important during the Company's annual production of The Nutcracker, which features approximately 260 children dancing as toys, snowflakes, and dolls.

Boston Ballet's phone system is a critical part of its behind-the-scenes efforts and its service to the public. A few years ago, its phone system at the time offered little flexibility for managing calls, messages, and extensions as the Company's needs changed throughout the year, particularly during productions. Moreover, the outdated system consumed a significant part of its annual budget, requiring ongoing costs for maintenance and support.

Critical Flexibility With IP Telephony

Frustrated, Boston Ballet began looking for another solution. The Company's long-time network vendor, Guardian Information Technology, introduced the Company to the ShoreTel digital voice communications system. As an IP telephony system, ShoreTel promised to reduce Boston Ballet's monthly line charges and recurring maintenance costs. Plus, the browser-based interface would allow the in-house staff to take control of even major moves, adds, and changes to the various extensions.

Guardian assisted in implementing the system, consisting of ShoreGear 60/12 and 120/24 voice switches, which run an embedded, real-time operating system, and one T1 interface unit. The T1 interface allows Boston Ballet to eliminate a significant number of individual POTS lines and consolidate serv-

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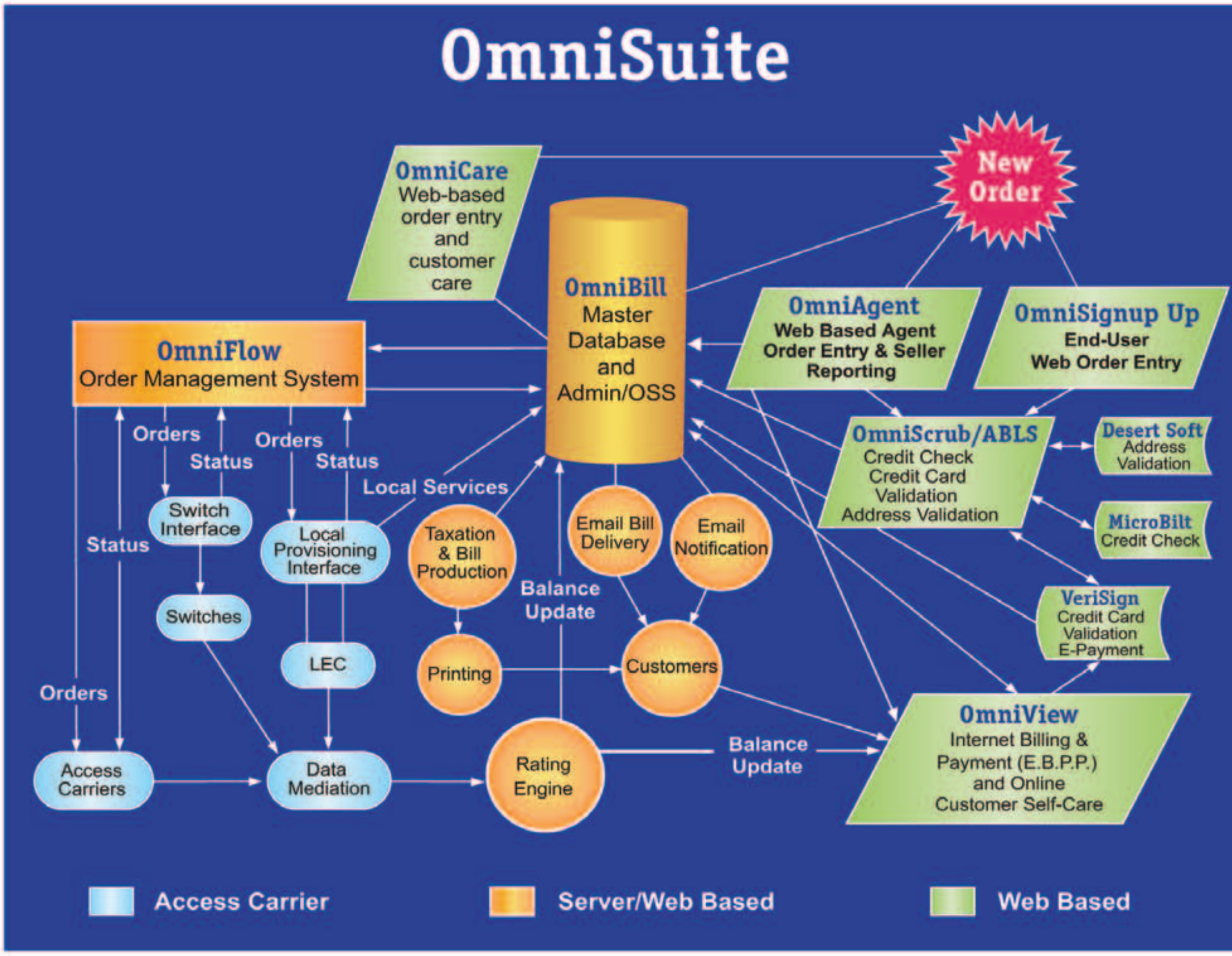

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"I truly appreciate the system's ease of administration..."

-Michael Kernochan, IT coordinator, Boston Ballet

ice onto a single, cost-effective and reliable circuit.

At the school, Guardian installed a switch connected to a 512K frame-relay circuit used to carry existing data traffic. There, Boston Ballet was able to eliminate five $180 per-month OPX lines. With ShoreTel in both locations, Boston Ballet reduced its monthly phone bill by almost 50 percent.

Reducing Costs, Complexity With Advanced Features

About 150 staff members and dancers now use the ShoreTel system, with some on phone extensions and others just taking advantage of softswitch voice mail capabilities. The Company's 60 dancers, who do not have individual phones, can dial in anytime, from any location to access their voice mail messages.

With ShoreTel and various menus configured for the box office, ballet company, and school, callers can easily select and get to the right individual's phone or voice mail. Over time, as staffing has shifted, Boston Ballet has found it no longer needs receptionists, saving at least $30,000 a year.

Call-handling features let Boston Ballet staff shift calls to ring at another extension, such as an external number or cell phone. If Boston is hit by one of its infamous snow storms, staff who are unable to make it to the office call the ShoreTel system from home to redirect calls to their homes and keep communication flowing. Most of the staff also access voice mail messages on their PCs via integration with Microsoft Outlook. With a unified mailbox, they can retrieve their e-mail and voice mail in a single location, and easily forward messages to co-workers.

Features like conferencing, caller ID, call history, voice mail forwarding and direct calling from PCs further enhance the ballet's ability to communicate internally and externally.

Grace Under Production Pressure

During productions, advanced features of the ShoreTel phone system allow Boston Ballet to handle an increasing number of calls and extensions, as well as forward calls to other phones as the crew moves among the school, offices, and theater.

The box office takes advantage of ShoreTel workgroup features to round-robin calls among the staff and ensure that patrons reach the next available customer service representative to purchase tickets. When someone goes to lunch, they simply log out of their extension. "Workgroups have been infinitely helpful with routing traffic," says Michael Kernochan, IT coordinator at Boston Ballet. "In the past, it was very rare that a customer would call in and immediately get the person they needed to speak to."

When The Nutcracker season rolls around each year, Boston Ballet easily configures the system with a Nutcracker information hotline. The children's coordinator for the beloved holiday tradition, responsible for managing the 260 young dancers in the show, easily forwards her extension to ring at her office, her cell phone, or at the theater. While their children are rehearsing, parents only have to remember one number, instead of the multiple numbers in productions past.

"If there's a change in plans, a snowstorm or cancellation, at a moment's notice the Nutcracker coordinator can change her voice mail recording instantaneously from anywhere, which helps us organize the students and keep parents informed and happy," comments Kernochan.

Likewise, the system supports the school by allowing teachers to route their calls to the theater and record informative messages for parents on their voice mail, from offsite.

Saving $30,000 to $40,000 Annually

By not having to call in outside help for changes, the ballet company can respond more quickly to user needs while reducing its costs for onsite technical support. "I truly appreciate the system's ease of administration, and simplicity in configuring call menus, adds, moves, and changes — all possible from a single graphical user interface," Kernochan said.

Though the large nonprofit operates with a $20 million annual operating budget, with funding from donations, sponsors, and ticket sales, it must continuously keep a close eye on its operational efficiency. In total, the ShoreTel system saves Boston Ballet $30,000 to $40,000 annually and helps productions go off more smoothly in the public's eyes and behind the curtain. IT

Richard A. Johnson is Chief Financial Officer, Boston Ballet.

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Conicelli Auto Group Drives Revenues With VoIP

Located in Southeastern Pennsylvania, Conicelli Auto Group is a multi-franchise automobile dealership that sells and services Honda, Mitsubishi, Nissan, Scion, and Toyota vehicles. A full service, bustling enterprise, Conicelli has earned a reputation for providing personal, expedient, and thorough care to its customers, striving to make the purchasing and ownership of its vehicles pleasant, seamless, and trouble free.

Conicelli (news - alert) sells about 8,000 new and used vehicles per year, and services over twice as many autos and trucks in a twelve-month period. The dealership requires a staff of about 100 professionals to keep operations and sales running smoothly and efficiently.

While some of Conicelli's employees may work at desktops, a significant number of sales professionals, mechanics, warehouse staff, and other team members are constantly on the go. In such a fast-paced environment, where every moment counts, fast, reliable communication is integral to the company's success.

When the company began planning a new showroom for its Mitsubishi franchise, Conicelli Auto Group sat down with Expert Technology Associates (ETA), an Inter-Tel Authorized Provider, to seek out options to address the company's communications challenges.

"We found ourselves in a unique situation," recalled Jack Monteleone, an executive with Conicelli. "We were planning to move the sales division to a new location, yet keep our service and parts department at a facility about a mile away. This posed a series of logistical obstacles, not the least of which was finding a cost-effective, reliable, and flexible means to communicate between locations. In addition, we needed a system that could grow and expand as economically as possible. After we met with ETA, we knew immediately that VoIP was clearly our best option."

Craig Marowitz, ETA's president, recalls the meeting they held with Conicelli to share their recommendation that Inter-Tel's VoIP (define - news - alert) solution made the most sense to address their short- and long-term needs.

"We sat down with Conicelli to explain everything — why IP was the right solution and how it would work. They already had a point-to-point T1 and routers, so we realized we could easily leverage their existing infrastructure to maximize cost-efficiency even more. VoIP proved to be the best option, from a technical and economic standpoint."

ETA's plan of action was simple and direct: Network two of Inter-Tel's Axxess communications platforms at two of Conicelli's dealerships, and utilize IP transport to link the new Mitsubishi franchise with the rest of the network.

"The distributed architecture of the Axxess platform results in a seamless and transparent network that links all of the dealer's sites," explained Marowitz. "As a result, calls coming into any one of the dealership's offices are easily routed to any of the other Conicelli locations, significantly increasing the efficiency in handling customer calls. In an industry where customers demand the highest level of service and care, this is of critical importance."

In addition to its IP-enabled Axxess platforms, Conicelli also acquired Inter-Tel's Model 8600 IP endpoints for deployment in its new Mitsubishi dealership. By leveraging much of its existing hardware and deploying IP where it makes business sense, the dealership's transition into the world of VoIP was smooth, orderly, and above all, cost-efficient.

As it turned out, technology was just one urgent issue that Inter-Tel (quote - news -alert) and ETA (news - alert) addressed on behalf of Conicelli.

As it was ready to deploy its VoIP

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solution, the dealership received word that its local and long-distance carrier — the backbone of its business — was withdrawing from the market. Conicelli had thirty days to find another provider or risk an interruption of service that would no doubt wreak havoc on the business. The good news, Conicelli learned, was that ETA could leverage Inter-Tel's Managed Services program to deliver a complete end-to-end communications solution to the dealer. Through the Managed Services program, Conicelli opted for the system hardware, software, endpoints, and business-specific applications that enabled the company to maximize its productivity and serve its customers more efficiently. Most importantly, the Managed Services program also included local and long distance services, maintenance and service, even data network planning and provisioning, to relieve Conicelli of the burden of managing the communications system, allowing them to focus on what they do best — selling top-flight vehicles to a selective clientele.

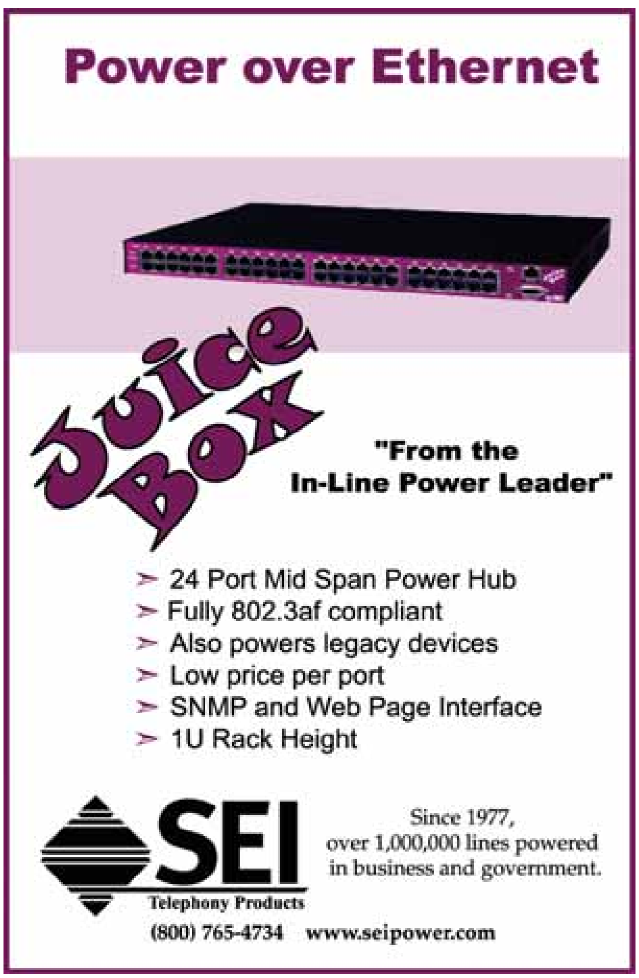
If increased efficiency and productivity were not enough, the Managed Services program also boosted Conicelli's bottom line by saving the company approximately $75,000 dollars over the course of one year.

"ETA and Inter-Tel came in and handled the situation quickly and correctly," said Monteleone. "They got involved immediately, made recommendations, and followed these through to the end. They clearly demonstrated that they thought of us as a partner instead of just a customer."

According to Marowitz, the real value ETA provides its customers is the managed services approach. "We tell all of our customers to look at us their very own internal communications department," he explained. "Our organization is equipped to handle all their needs, from systems to applications to carrier services to support. The vast majority of our customers are more than happy to hand over their communications issues to us so they can focus on operating their businesses."

For Conicelli, the benefits of working with ETA and Inter-Tel are even more obvious. "This partnership has helped us generate more business, serve our customers more efficiently, and reduce our costs," concluded Conicelli's Monteleone. "The outcome far exceeded our expectations." IT

The dealership's transition into the world of VoIP was smooth, orderly, and above all, cost-efficient.



IT EXPO Miami Shows The Sunniest Side Of VoIP 2.0

By Johanne Torres

Here at TMC we were all excited because we were going to escape the frigid Connecticut winter by packing our bags and heading down to sunny Miami, Florida to what was to become one of the most attended shows in our conference history: The Internet Telephony Conference & EXPO Spring 2005 at the Hyatt Regency Miami during the week of February 22–25, 2005.

We all had an idea of how many attendees were registered before we flew down there, but, personally I was incredibly overwhelmed at the massive amount of attendees spread out all over our conference sessions and our EXPO halls. I had planned one-on-one interviews with some of our exhibitors, and it literally took me anywhere from fifteen to twenty minutes to get through the super long booth lines to locate the executives I was to meet. Just how many attendees were there you ask? Over 6,200! It was amazing to also learn that representatives from over 70 different countries visited us in Miami; we even had simultaneous translations for our Latin American attendees in every keynote, making sure no one missed a beat!

We listened to the speakers kick off the conference sessions on Tuesday morning by delivering a pretty good overall look at IP telephony and its adoption in the military, contingency agencies, and contact centers all over the U.S. Discussions on SIP's role in open source, the future of enterprise network peering, and how to migrate to IP telephony also took place during the first day of one of the most attended shows in TMC's history.

A great highlight of day one was the head-to-head discussion panel presented to attendees at the IP Contact Center Shootout. During our panel discussion we got to see how four experts explained the advantages of deploying IP-based communication systems in the contact center corporate environment. The following panelists participated in our debate moderated by our own Internet Telephony magazine editorial director Greg Galitzine: Tod Famous, product manager for Cisco Systems (quote - news - alert); Jonathan Philips, solutions architect for Nortel; Oscar Alban, principal global market consultant for Witness Systems; and Joe McFadden, VP of corporate marketing for Nuasis.

Highlights for Wednesday included the keynote grand opening session with AT&T's (quote - news - alert) Cathy Martine and Lucent Technology's Stef van Aarle. Both speakers agreed that VoIP technology is currently changing the way we communicate both at home, as well as in the corporate world. They explained the really hot topic of cellular and IP network convergence aiding seamless telecommunications between employers and remote workers. This

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Go To Table of Contents I Go To Ad Index

topic seemed to be key in most conference and keynote sessions.

The conference's second day saw sessions filled to maximum capacity, turning the rooms into standing room only learning classrooms. It was also the day in which very interesting conference tracks took place such as UNE-P To VoIP Summit, VoIP solutions for the enterprise and government agencies, IP Telephony for developers and service providers, as well the conclusion of the IP Contact Center Summit. Wednesday also showcased Reseller Days, a track designed specifically for VoIP technology resellers where attendees learned how

"This show was completely jammed! I've never seen traffic like the first two hours of the exhibit hall on Wednesday night."
– Keith Weiner, DiamondWare

to make money reselling VoIP.

Wednesday ended with packed sessions on everything about security in the Internet telephony environment. Afternoon sessions highlights included telecom fraud, secure enterprise mobility and Quality of Service (QoS) in enterprise VoIP deployments. Keynoters Ken Epps from Bay Packets and Vertical's William Tauscher capped the second day with their presentations as a prelude to the highly acclaimed Future of Internet Telephony panel discussion that early evening.

Thursday's early conference sessions on WiFi technology, Enterprise VoIP

"The number of international service providers at Internet Telephony Expo looking for consumer and enterprise VoIP solutions for their subscribers is extremely impressive."
– Sarah Hofstetter

integration, and P2P VoIP kicked off day number three of the conference in full speed leading up to a very impressive keynote presentation piloted by Voiceglo's Ed Cespedes, Aspect Communications' Gary Barnett, and Mark Spencer from Digium.

Cespedes began his speech with a brief explanation of the current state of the VoIP industry. He described the pioneer-crowded industry as one with many challenges to be taken on by

"The conference was professionally run, and with so many tracks, and new tracks each day, there were always a choice of sessions I'd want to attend. There wasn't a moment wasted in the day… Thanks for a great conference."
– Karen Strouse, Management Solutions

organizations willing to invest significantly in the many new business opportunities available in the extensive market.

He went on to advise the audience to do their homework to determine which solution better fits their business goals. He warned professionals about solely depending on whatever is posted on message boards, and blog opinions and feedback. He recommended to instead conduct a thorough investigation by asking vendors to provide client contact information, so as to contact real clients who know best about the solutions they are implementing.

Cespedes concluded by expressing his gratitude to organizers of events like Internet Telephony Conference & EXPO, for offering professionals in our industry the opportunity to see these products and services in action. He noted the incredible difference in attendance and participation at these events, compared to two years ago.

Aspect Communications' Barnett began his presentation by describing how VoIP-based services and products in the corporate contact center sector are shaping the telecom industry. He emphasized the importance of the ability that organizations have when employing home-based contact center representatives in multiple locations all over the world. He said it was a great asset that companies could gain from the ability to outsource services in order to reduce operating costs.

Barnett also spoke about the importance of SIP-based products for the enterprise. He predicts that at least sixty percent of businesses will choose to integrate SIP-based products in the corporate telecom environment.

He mentioned that contact centers

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Go To Table of Contents | Go To Ad Index

"This show was unbelievable! It was like there was a sale going on in the exhibit hall with people everywhere. We go to other VoIP events and it's not this crowded and usually very calm. Not here. This is the happening VoIP event!"

– Ahsan B. Ali (Regional Sales Manager), SysMaster Corporation

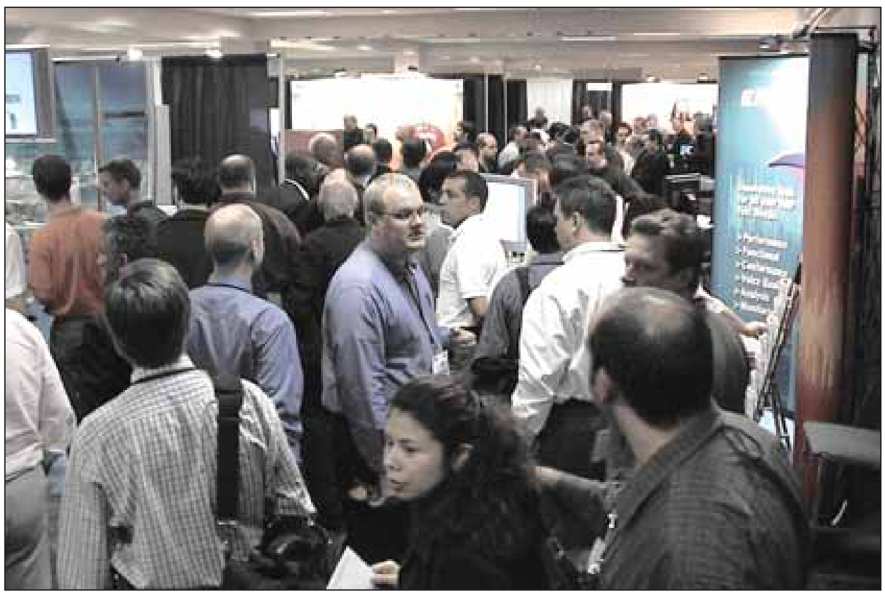
will definitely benefit from the integration of Web collaboration, e-mail, video, and voice services, and sees the integration to be as easy as Microsoft office is in PCs. The integration of PSTN and IP PBX (define - news - alert) systems will keep getting easier in the coming years. Barnett concluded by predicting that by 2008, most contact center representatives will be based out of their homes.

One of the most anticipated events taking place during the third day was The Open Source Debate between TMC's (news - alert) president Rich Tehrani and Digium's Mark Spencer. Tehrani asked everything from who fixes systems when they break to picking a Linux-based system when being clueless about that operating system.

Spencer explained the benefits of integrating systems that make it easy for corporations to fix problems that may arise. He said he believes that organizations should be able to fix bugs and perform upgrades through a Web-based interface, eliminating the need for having to contact resellers and a number of vendors responsible for offering the products and services. He said that the key benefit of integrating these systems involves transferring the telephony systems to the IT department of your organization, merging data and voice integrators.

Spencer concluded by describing VoIP as a huge business opportunity that may create challenges for businesses wanting to grow their services around the technology, and advised that the task won't be an easy one. He mentioned that companies should place the greatest value on finding technologically experienced management.

Nortel's Phil Edholm presented a



"From the opening of the exhibit hall until the time I left to catch my plane home, the exhibit area was completely full. The Miami event definitely established that VoIP is the most exciting technology in the telecommunications industry today. I was also very impressed with the international attendance. For any vendor targeting the Latin American market, this is the show."

– Max Schroeder, MSI Services

Thursday afternoon keynote presentation titled "Convergence Vision," and he gave a detailed glance at the state of the VoIP industry and its new developments to make telecommunications mobile in order to fit our busy lifestyle. He explained how important it is for workers to be reachable at all times everywhere they go, and described wireless accessibility as the new vision.

Edholm explained that SIP and packetized voice will indeed change the world, or at least the way we communicate. But not VoIP, because, as he explained, VoIP is already here; it is really up to us to adapt the technology to our telecom efforts. "SIP (define - news - alert) creates a change on how we implement systems," Edholm explained. Ultimately, these two technologies change the process from accessing devices to, in stead, accessing human beings. These multimedia communication systems will eventually create an inter-human Web, connecting all devices and making humans available practically everywhere at all times, he concluded.

Highlights of the fourth and final day of the conference included keynote sessions by Sansay's Andy Voss and Michael Rouleau from Time Warner Telecom. Friday's conference session tracks included more sessions on WiFi technology, solutions for the enterprise and government agencies, as well as workshops for service providers and SIP.

Overall, the conference attendees seemed to be more than satisfied with all the topics covered. I heard many exhibitors praise our team for providing them with a huge EXPO traffic turnout. Many of the exhibitors were very excited to meet industry executives from so many different countries in one place. I can hardly wait until we start packing again for out next show in Los Angeles Convention Center in Los Angeles, California this October 24–27, 2005… Hope to see you there! IT

"Once the … exhibit hall opened, our expectations were exceeded on attendee quantity and quality. There were several times that we were overwhelmed by the number of people passing our booth and stopping to talk to us. As we looked around, the halls of the show floor were wall-to-wall attendees in every direction and they were really looking at each booth as opposed to just strolling the floor. They were there to do business, not just walk the show."

– Todd Hirshorn, InPhonex

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Go To Table of Contents | Go To Ad Index

The editorial staff of INTERNET TELEPHONY® Magazine is pleased to announce the winners of the 2005 INTERNET TELEPHONY Conference & EXPO Best of Show awards for outstanding products and services demonstrated at the just-concluded event that was held in Miami, FL, February 23–25.

The awards emphasize innovative technology and creative product feature sets. Forward-looking products that will serve as a base upon which to build future developments in our industry were also awarded. Each winner submitted a thorough application and displayed and demonstrated their product on the INTERNET TELEPHONY Conference & EXPO exhibit floor.

We congratulate the winners and look forward to seeing you all at our next event in Los Angeles, October 24–27, 2005.

COMPANY NAME	PRODUCT NAME	WEB SITE
Comvoz	Telco Generator Portal	http://www.instatelco.com
EyeBill Interactive Solutions	External Dynamic Routing	http://www.eyebill.net
GlobalNet	Turn Key Customizable Consumer VoIP Platform for Service Providers	http://www.iDialip.com
Grandstream	GXP 2000	http://www.grandstream.com
i2Telecom	Voice Stick	http://www.i2telecom.com
Ingate Systems	Ingate VoIP Survival	http://www.ingate.com
Inphonex	The InPhonex Flash Phone	http://www.inphonex.com
Interstar Technologies	XMediusFAX SP	http://www.faxserver.com
MERA Networks	External Dynamic Routing	http://www.mera-voip.com
MKC Networks	7000 Communications Server	http://www.mkcnetworks.com
net.com	Shout900	http://www.net.com
NetFabric Corp.	FUS1ON 12:8 Intelligent Call Director	http://www.netfabric.com
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Viola Networks	NetAlly RealTime version 4.3	http://www.violanetworks.com

INTERNET TELEPHONY®
EDITORS' CHOICE AWARD
2005

Aastra VentureIP powered by Nimcat Networks

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http://www.aastra.com

Nimcat Networks
1135 Innovation Dr.
Ottawa, Ontario
Canada, K1Y 4T8
Tel.: 613-592-4343
http://www.nimcatnetworks.com

Price: Venture IP 480i IP Telephone Set — $379, Venture IP Gateway — $289

Aastra's (news - alert) VentureIP 480i and VentureIP Gateway devices leverage Nimcat Networks' embedded software. Nimcat's nimX is an embedded, P2P call processing software that takes the intelligence usually found at the PBX and distributes it to end-user telephone sets.

By placing the traditional PBX (define - news - alert) functionality such as voice mail on the actual phone, nimX-powered devices enable SMBs to "grow" their phone system requirements one set at a time. One such company, Aastra, has a nimX-powered phone called VentureIP. Actually, we really shouldn't call it a "phone" since each phone is a PBX or phone system unto itself — with minimum two phones of course! The phone system is powered by Nimcat Networks' nimX software and it enables SMBs (define - news - alert) to install, operate, and manage a full-featured IP-based phone system by simply plugging the VentureIP 480i phone into their network. The system scales phone by phone and can be connected to the local PSTN (define - news - alert) via a VentureIP Gateway.

The VentureIP 480i handsets independently recognize the other terminals (phones or gateways) on the network and instantly form a trusted, virtual exchange with the ability to interact with their peers or connect to the PSTN or any WAN VoIP channel. As users plug a handset into their network, the device automatically configures itself, allowing calls to be made and received without any complex setup or centralized equipment.

The VentureIP Gateway is also nimX-powered and it enables connectivity to the local PSTN. Each VentureIP Gateway supports up to four telephone lines, and multiple gateways can reside on the same network for additional trunk capacity. Just like the VentureIP 480i handset, the VentureIP Gateway is self-aware and automatically configures itself.

Installation

Installing the Aastra phones with the embedded Nimcat Networks P2P VoIP technology could not be any simpler — we didn't even use the manual. Each phone comes with a two-port (RJ-45) Ethernet power brick. One of the RJ-45 ports is labeled "Data" and the other "Power/Data." We made the correct assumption that the "Power/Data" connects to the Aastra phones and that the Data port connects to your network's hub or switch. We repeated the two connections for each of the three phones. Next, we connected a four-port analog gateway onto our LAN and connected four analog lines to a Teltone analog simulator. We then powered all the components and the phones booted up in seconds. Each of the phones has a large eight-lines-by-21-characters LCD display, which prompted us to enter in our last name. Using the phone's touchtone keypad, we entered in alphanumeric names. That's it. Done! The phones were connected and installed. The devices were automatically assigned an IP address via DHCP and the extensions were automatically assigned as well (x200, x201, x203).

RATINGS (0–5)
Installation: 5
Documentation:
Features: 4.75
GUI: 5
Overall: A

Operational Testing

For our first test we dialed the exten-

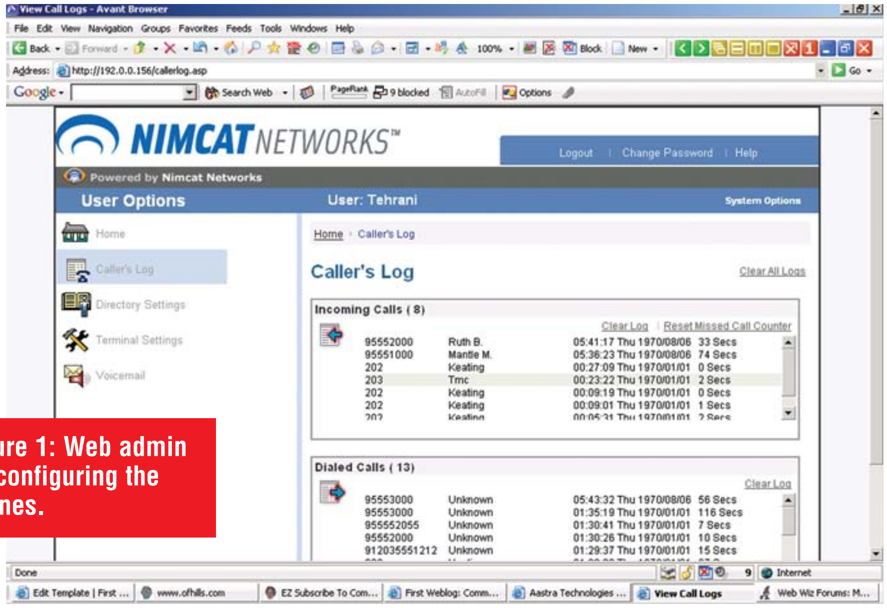


Figure 1: Web admin for configuring the phones.

sion of one of the Aastra phones. The phone rang, we picked up, and were immediately impressed with the voice quality and almost no perceivable latency. We tested the usual suspects such as putting a call on hold, call transfer, call conferencing, etc., and all tests passed with flying colors. Next, we tested the ability to access the four-port analog gateway to make PSTN calls. We pressed '9' to get an outside line, entered in a phone number (555-2000) for the second port on the Teltone analog simulator and we saw the first LED light for the Teltone simulator light up followed by the second port lighting up (the port we dialed). Since the second port was connected on the other end to the four-port gateway, the gateway picked up and played the Aastra auto-attendant greeting. From this point we were able to dial an extension or even dial by last name using the built-in directory assistance. Both CallerID name and number displays on the LCD screen.

One interesting "usability" feature is that the phone can obtain the correct time from incoming PSTN calls and update the system time automatically. Another nice usability feature is that you can navigate all of the features of the phone via the LCD screen. Everything from viewing the call records to recording your greeting can all be done via the LCD screen. One minor usability annoyance is that when you log on to your voicemail (20 minutes default storage), the LCD defaults to the voicemail's main menu which merely displays the number of new and old messages. You have to then click the down arrow to get to the first new message. We'd rather it default to immediately displaying the first new voice message so you can quickly play back a new message without an extra phone key press. Perhaps this could even be a user selectable option. Another nice usability feature is that when you are in a conference, you can drop any of the participants using the LCD display. Many other conferencing solutions only let you drop the last participant to the conference or the first participant. One really cool feature is that you can actually forward your voicemail to e-mail simply by specifying an SMTP (define - news - alert) server and an e-mail address. This is a feature you typically only see in very expensive voicemail or PBX solutions.

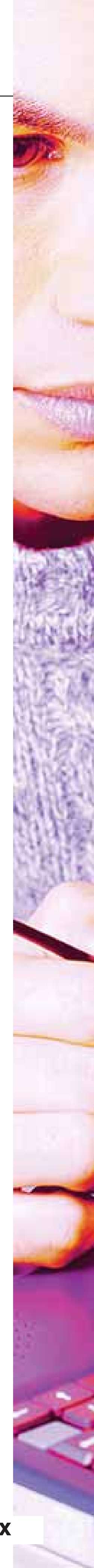
The voicemail system was quite comprehensive. It allows you to screen callers leaving a message and pull them out, offers single-button call return from voice mail, enables users to retrieve voicemail via the LCD or externally by dialing in, and even provides the ability to leave and retrieve voice mails for disconnected or unavailable sets. It's part of the whole peer-to-peer technology that allows you to still leave voicemail even when your phoneset is disconnected. This adds redundancy that you don't even see in traditional voicemail systems. When a traditional voicemail server is down, it's down for every phone!

Everything is preconfigured for you on the Aastra phones — the auto-attendant is set to a default greeting, the same with the voicemail, and the extensions are automatically assigned at first initialization. It's as close to a "plug and play" phone system as you will find and considering it's an IP-PBX targeting the SMB, this is one of its key selling points. The Web administration is very powerful. It allows administrators to log on to all of the phones and it grants users limited access to their features such as call forwarding, call logs, and more (Figure 1).

Not that most people would care what's under the hood, but the phones utilize SIP technology. Although it does use SIP and has no need for a SIP registrar or proxy — it's actually built into the nimX software. Since it is SIP-based,we were curious if we could make a SIP call using a SIP softphone. We contacted Nimcat Networks and then told us that they have done this with the Xten SIP client. Due to editorial deadlines we were not able to test this feature. They also informed us that in Q3 they will have the ability for you to double-click a phone number on your PC's screen and initiate an outbound call. We neglected to ask if an inbound screen-pop feature would also be available.

Conclusion

For SMBs, the option of going with an IP-PBX solution is often cost prohibitive. It requires a large upfront investment as well as additional maintenance costs. The advantage of this P2P VoIP system is that it is a "plug and play" affair, virtually eliminating costly service installation costs, and as previously mentioned, you can slowly "grow" into the system a few phone sets at a time. For instance, a 10-employee SMB can get a 10x4 (10 phones, four trunks) Aastra phone system for merely $4,079. The ease of installation and the ease of administration via the phone's Web interface are two more powerful reasons why this is a great solution for small to medium businesses and TMC Labs wholeheartedly recommends it. IT



Creating Value In Consumer VoIP

As the telecommunications industry thaws out from what many have called the telecom nuclear winter, the search for the "next big thing" has kicked into high gear. The voice communications industry has been craving a long overdue mid-life kicker and consumer VoIP (define - news - alert)seems to be just the right thing.

There is great promise in what consumer VoIP can deliver to the mass market and to the earnings reports of companies industry-wide. To date, much of the focus has been on technical matters around implementing VoIP infrastructure and PSTN (define - news - alert) interworking. IP networks are being upgraded, next-generation architectures built out, and component interoperability testing completed. However, often overlooked in the very eye of all this frenetic activity sits the most important deployment consideration of all — will people pay incremental dollars for consumer VoIP?

The good news is yes they will, as we've seen from recent market activity. Yet how much consumers will pay depends on the level of value they receive in return, and how the consumer defines value. Consumers can be both inconspicuous and contradictory in their buying habits. They can simultaneously spend indiscriminately in some categories while penny-pinching in others.

For example, U.S. consumers dutifully spend over $126 billion annually on telephone services across all categories including wireline local, long distance, and wireless. At the same time, these consumers limit their personal telephone expenses to approximately two percent of all household spending. Currently that equates to approximately $95 per month per household. This can be thought of as the consumer's Implicit Value of Voice, and currently cellular accounts for more than half of that value.

So if all consumer VoIP has to offer is plain 'voice' over IP, then the consumer has already spoken. 'Voice' is worth two percent of household expenses, as it has been for the past 20 years, and it will likely stay that way for the foreseeable future.

This value assignment has nothing to do with how the voice is delivered, whether it is voice over base band TDM (define - news - alert) over UTP (define - news - alert) , IP, CDMA or any other protocol or transmission medium. It does, however, account for the consumer's value of plain voice plus any other incremental value being delivered. For example, consider the incremental value of mobility. Surely consumers don't flock to cellular telephone service for its superior voice quality or its P.001 grade of service. Cellular is clearly inferior to wireline POTS (define - news - alert) on those metrics. But the consumer has assigned significant value to his freedom to roam and is willing to consistently pay incremental dollars for that incremental value.

This explains why the industry's initial POTS replacement consumer VoIP services do not fetch more than about $40 per month, which is the total wireline component of household telephone spending. It also explains why today's POTS replacement VoIP offerings, intent on garnering early market share, are aggressively priced well below that figure.

Ultimately, in order to get the consumer to spend dollars over and above the two percent spending allotted to voice, carriers must deliver "value

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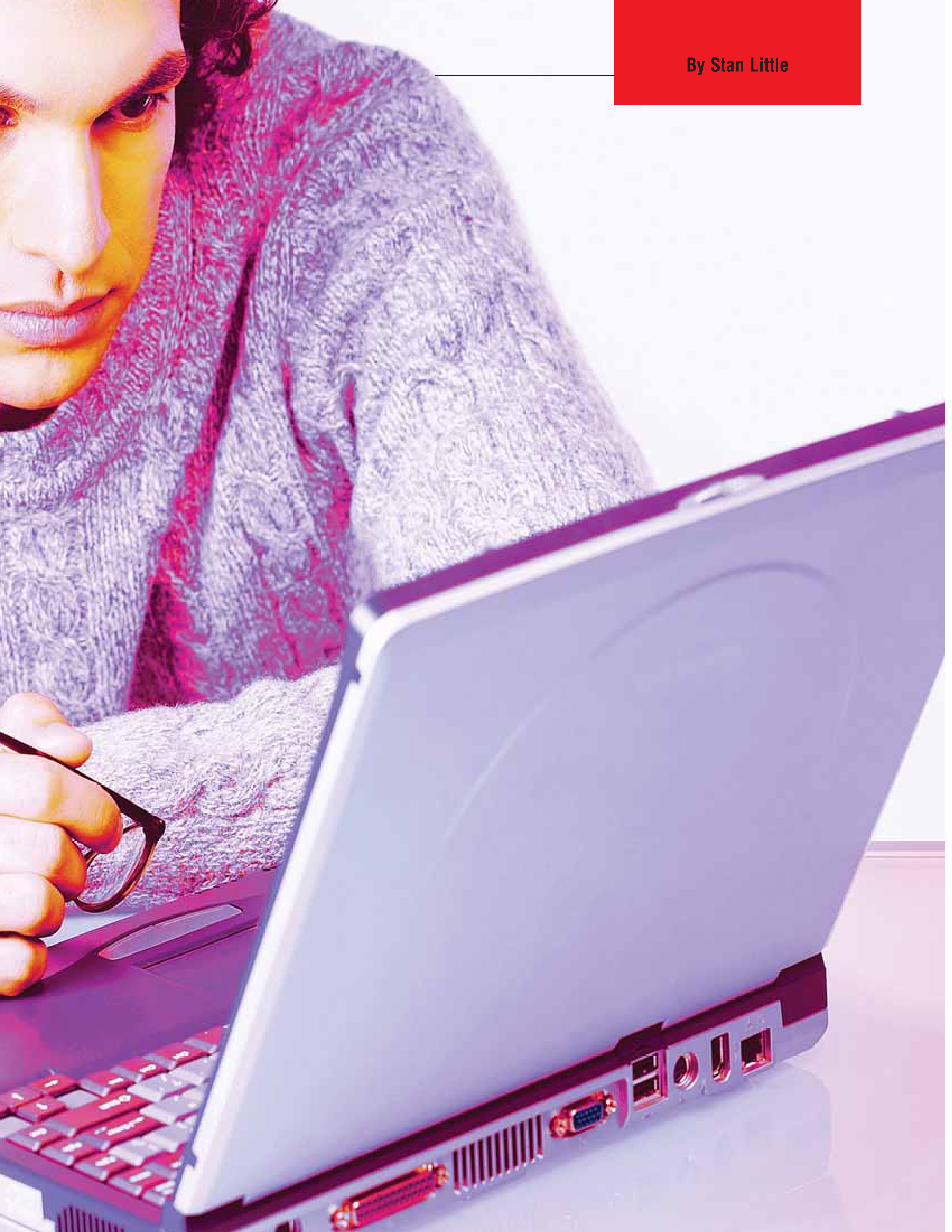
Go To Table of Contents | Go To Ad Index



By Stan Little

beyond voice.” That sounds pretty simple. However, consumers themselves define “Value,” but don’t always plainly articulate these value preferences. So if the consumer never speaks directly about these Explicit Value Preferences, how do we determine what services they’ll value and buy?

Consumers indicate their appreciation for value based on where they spend their money, time, and effort. Luckily, they are pretty good about leaving a clear and consistent value trail that can be applied to VoIP services. Without a doubt, consumers cherish their privacy and have been consistently willing to dedicate both time and money to protecting and enjoying it. They constantly demonstrate how much they value privacy, especially in regards to their telephone.

For example, consider that direct consumer outcry over privacy invasion resulted in the creation of the Federal Trade Commission’s National Do Not Call Registry (DNCR). By the end of the first eight months of the DNCR service, a staggering 55 million people had signed up. In a related move, the FTC instituted telemarketer sales rules requiring telemarketers to present Caller ID in all phone solicitations. Moreover, state public utilities commissions report a large and growing subscription to unlisted telephone numbers. In California alone, more than 50 percent of residents have unpublished numbers. There is an $800 million annual price tag for these California consumers to remove their numbers from the phonebook and 411 directory.

Nationally, the U.S. consumer pays almost 10 percent of his phone expenses on privacy features. To be a leader in consumer VoIP, carriers must cater to consumers’ demand for privacy, and will be rewarded by their willingness to pay to protect it. In fact, VoIP adds incremental value by adding dynamic next-generation services that give consumers greater privacy and call control based on who is calling them, when they are calling, why they are calling, or

Crossing The Chasm To Mainstream VoIP Adoption

By Cynthia Carpenter

Voice over IP (VoIP) has been on the Telecommunications radar since the early 1990s. As with most emergent technologies, early returns are mixed, and actual adoption is proceeding somewhat more slowly than the first enthusiasts predicted. The key question right now is: what will it take for VoIP to be adopted en masse? Research indicates that a majority of voice service customers (71 percent) are open to signing up for VoIP, but need a compelling reason to switch. For VoIP to emerge as a mainstream consumer application, VoIP service offerings must motivate this group. Initial VoIP awareness is critical—very few potential customers know what VoIP is or that it is an option. Once past the awareness hurdle, it appears that the most-likely motivators will be adequate discounts, offerings bundled with broadband, and the appeal of VoIP-specific features. An effective marketing strategy based around these variables, combined with a push from prominent consumer brand companies, could lead the VoIP deployment across the chasm that exists between early VoIP adopters and the mass market.

Where Is VoIP Deployment Right Now?

Popular press and industry trade journals would lead us to believe that VoIP is nearly ubiquitous, but the reality, according to the Yankee Group, is that there are currently approximately 1.2 million VoIP users. This research indicates further that only one in three households has the broadband connectivity that is the first prerequisite for VoIP deployment. Market research also reveals that those customers who have adopted VoIP are driven primarily by price, make 20 percent more long distance calls than average, are technologically savvy, and are likely to be dissatisfied with their incumbent service provider.

As VoIP emerges as a significant new technology, the initial crop of VoIP-focused new entrants will be joined by powerful incumbent providers ranging from the RBOCs to cable companies to the largest Internet service providers. As is true with most emerging technologies, new entrants are most motivated to capitalize on the first opportunities and most able to absorb the inevitable stresses and challenges of an imperfect platform. At this point, the companies that account for most of the over one million current VoIP subscribers are in the Enhanced Service Provider (ESP) and cable space and are not traditional phone companies.

The Market Opportunity Beyond The Chasm

The theory of high-tech market segmentation and product adoption that Geoffrey Moore outlines in Crossing the Chasm (1999) is particularly applicable to current VoIP adoption. Moore describes how a successful high-tech product moves through a fairly predictable adoption curve from Innovators to Early Adopters and then across a “Chasm” to the Early Majority. Applying this terminology, VoIP has made significant inroads into the Innovator and Early Adopter segments and has yet to obtain any real traction among the later groups which form the mass market.

While current adopters have given VoIP a good start, they will not be sufficient to sustain continued growth. We have reached the point where the only path to success is across the Chasm — the gap that exists between success with the Early Adopters and mass adoption of a new technology. Very little that the VoIP industry has done to enlist the Innovators and Early Adopters will work effectively to attract any substantial numbers in the Early Majority segment. While the Early Majority can still be strongly motivated by cost savings and access to useful new features, they expect all the service and support they are accustomed to getting from traditional telephony providers.

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Go To Table of Contents | Go To Ad Index

Two critical factors will enlist the Early Majority. First, the technology will have to reach the level of quality, reliability, and safety of traditional phone service, and second, these potential customers need compelling reasons to embrace the new technology.

Key VoIP Market Drivers

With a focus on addressing the key concerns and opportunities that will lead to mass market adoption, VoIP service providers can cross the Chasm. Research indicates that initial factors include awareness and attitudes toward current service providers. Once customers are interested in VoIP service, the major influencers of an actual purchase decision look to be cost savings, bundling with other communications services and new features enabled by the digital integration capabilities inherent to VoIP.

Awareness

One of the critical findings of this market research is that approximately 60 percent of surveyed consumers claim never to have heard of VoIP (or Internet Phone Service). However, after VoIP was defined to these consumers, 71 percent indicated that they would be open to signing up for VoIP if it was available. Clearly, a key issue for the untapped market is awareness. Providers cannot assume much base of understanding or awareness about VoIP at this point, so consumer education is critical.

Dissatisfaction With Current Provider

Market research indicated that those who demonstrated interest in signing up for VoIP services were also much more likely to be dissatisfied with their current phone provider. Those who were interested in VoIP also answered that they would respond more positively if the service were offered by an established phone company. The message here appears to be that, while dissatisfaction with current phone service can drive motivation to switch to VoIP, this does not mean that customers would not consider a VoIP offering from their incumbent provider. Dissatisfaction with the current provider could reside as much in dissatisfaction with traditional phone service options and costs as with the provider itself. There is a real opportunity here for incumbents to enter the market to meet needs for VoIP services and simultaneously to improve overall customer satisfaction.

Cost Savings

Early Adopters joined the market as they saw the opportunity for substantial savings over traditional telephone service. This was particularly compelling for segments where international and toll calling represent significant expenses. In surveys, respondents who indicated strong interest in VoIP become interested in price as a motivation to switch only when it was described as being 25 percent lower than the cost of traditional phone service. In addition, it is important to note that the prices that motivate customers, where there is at least a 25 percent savings over traditional phone service, remain at the higher end of the current VoIP price points. It is clear from this that quantifiable cost savings and the promise of consistent service quality, not "rock bottom prices" should remain the focus of successful marketing campaigns. In fact, prices that are dramatically lower than what consumers are currently paying may signal that VoIP is a poor quality substitute.

Bundled Offerings

Broadband providers have a huge opportunity that they have already begun to exploit. Particularly in the cable industry, VoIP/Broadband service bundles are proving very attractive to the installed base. According to research, this is the next most critical adoption factor after the 25 percent cost savings. Potential VoIP users indicated that a compelling offer would be VoIP bundled with a discount on Broadband services. Initial analysis also indicates that price and bundled offerings are much more important than the remaining potential market drivers. Disruptive pricing combined with broadband bundling will maximize demand for VoIP.

Digital Integration & Advanced Features

While price disruption and bundled value appear to be the largest motivators for potential consumers, the survey also reveals that there is significant interest in VoIP's ability to provide access to new phone features. Of those who said they were definitely interested in VoIP, 91 percent indicated that new feature availability would be a strong enticement to consider VoIP. When asked to indicate which features would be most compelling, messaging/screening features and enhanced wireless (WiFi) capabilities topped the list. Conversely, more specialized features such as vanity phone numbers, translation services, and video conferencing generated comparatively minimal interest. Service providers will need to promote the features most likely to appeal to the emerging market, while being careful not to overwhelm consumers with technologically intriguing features that might have little general utility.

The Path To Majority VoIP Deployment

Awareness is the most critical first target to drive majority VoIP deployment. Marketing efforts should focus on providing simple definitions of VoIP that combine with pricing, bundling, and new feature opportunities. The research indicates that the initial focus on advanced technology and features (which has been successful in gaining customers among Innovators and Early Adopters) would be ineffective for the Early Majority. The majority of potential customers will need to believe that the transition from traditional phone service to VoIP will be virtually transparent and that it will bring immediate and tangible benefits and savings. And finally, incumbent involvement in VoIP roll-out is key, as many potential customers will be more confident buying VoIP services from a known provider. IT

Cynthia Carpenter is Level 3's vice president of product marketing for wholesale and consumer voice services. For more information, please visit http://www.level3.com.

all of the above. And consumers will be able to configure these and other valuable capabilities through easy-to-use interfaces on their phone, computer, or TV set-top box, all with “set-it-and-forget-it” simplicity.

Another major driver for all consumers is the general pursuit of leisure time and entertainment to enhance their lifestyle. Many of the telecom industry’s most successful incremental services tend to be “lifestyle enhancing.” That is, consumers either derive gratification directly from the service or it enhances another part of their everyday life. The relationships that people have and the way they interact within them is central to their lifestyle and the quality of it. This is another area where VoIP adds value.

In general, consumers live well beyond the walls of a residence and value their participation in larger communities of interest and affinity groups. Common groups for the consumer exist around their kinships, their physical location, and how they spend their leisure time. It is these communities of interest that enhance, if not define, the consumer’s everyday life, and people spend a lot of their valuable time interacting within these non-work communities.

VoIP service providers now have the capability to bolster the value of their services by enabling consumers to set up and manage their communications more effectively by applying these types of communications relationships in private groups. For example friends and family members can enjoy special capabilities within the group, regardless of location, such as VoIP push-to-talk, intra-group calling priority, or voice/text/info group messaging. So to think that consumer VoIP means just residential voice is as limiting as using the term “car phone” to describe your feature-rich cellular voice and text messaging service.

In ever greater numbers, consumers are demonstrating that that they also value their gadgets, and why would VoIP consumers be any different? Accelerating growth into the next phase

Residential-Scale VoIP Testing: Why And How?

By Andy Huckridge

Quickly on the heels of the race to deliver VoIP services to the masses comes an even more heated race to deliver VoIP service so good that customers never leave. Toward that end, Internet telephony service providers (ITSPs) will soon apply carrier-class testing to truly validate network readiness and service quality.

Testing large-scale consumer VoIP will require test devices that can emulate legions of end-user terminals in the lab, and then monitor the highest capacity on-net traffic. Further, whatever subjective feedback end users offer, only quantitative measures of latency, dropped packets, jitter, and other objective quality metrics can provide the business and operational intelligence to engineer and maintain quality networks.

This kind of objective intelligence can prove extremely useful. Retail ITSPs, for example, need such objective measures to hold their wholesale network providers’ accountable. Simultaneously, objective measures can empower wholesalers to document their networks’ readiness and performance over time.

Additionally, some carriers may wish to benchmark quality to determine the true measure of the tradeoffs associated with certain equipment selections. One audio codec, for example, may yield lower sound quality yet fewer dropped packets, ultimately adding up to better call quality.

Testing architectures will matter too. A VoIP call may perform well from point A to B, yet poorly from A to C. A certain link may pass muster during one day-part but crumble under the traffic loads of another. Consequently, useful performance intelligence will require visibility across the network, end-to-end, link-by-link, terminal-to-terminal. This can best achieved through a test system that can be distributed at key junctures across the VoIP network while linking back to a single test user interface that provides both end-to-end and drill-down analysis.

Carrier-scale testing also will require test equipment capabilities to:

• Measure both signaling performance, for efficient call set-up and tear-down, and media session performance.

• Report on and analyze the inter-impact of multiple applications. As Johnny downloads a big file, what happens to Alice’s VoIP call? For this, reports should map side-by-side application results on a common graph.

• Verify codec capability to carry music as well as voice.

• Verify the terminal adapter’s ability to generate tones for short, 10-digit and international dialing, as well as touchtone activation of voice mail and the like.

• Measure the performance of services that are unique to each ITSP, such as star-codes used to activate certain value-added applications like teleworker access to enterprise VoIP.

As the residential VoIP gold rush continues on all fronts, ITSPs (define - news - alert) will need a reliable assayer to document how much gold quality exists in their network and service performance. As history suggests, today’s boomtown can become tomorrow’s tumbleweed junction for the entrepreneur who fails to verify the quality of his wares. IT

Andy Huckridge is product marketing manager, IP telephony, at Spirent Communications. For more information please visit http://www.spirentcom.com.

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Go To Table of Contents | Go To Ad Index

Will people pay incremental dollars for consumer VoIP? The good news is "yes" they will.

of consumer VoIP requires the rapid introduction of IP terminal devices coupled with mobility. Network-hosted VoIP, smart terminals and mobility bring together the best of all worlds where hosted user information, preferences, and contact data achieve additional value through more capable, portable, and usable devices.

There are common examples in other service industries that have benefited from enhanced devices. Consider the impact that PDA units, MP3 players, and similar devices have had on the related network services and applications that support them.

A compelling example of adding consumer value to voice services can be seen in today's cellular market. Providers and device manufacturers are steadily upgrading handsets to accommodate services that go well beyond voice. For example, these new devices make typing and addressing text messages simple. That move is one of the driving forces behind the 175 million U.S. cellular subscribers (over 1 billion globally), and the billions of SMS (define - news - alert) messages sent monthly.

Currently, the price points for IP devices are relatively high, and their costs impact consumer VoIP adoption. But the hard reality is, there's only so much that can be done with a VoIP-adapted $9 princess phone nailed to the wall in the kitchen. A manufacturer-subsidized program to provide low-cost/no-cost consumer IP phones would work just as it does in cellular. Carriers provide a high-end device, enabling the subscriber to purchase advanced services at low cost with a term contract. After all, these are the very same consumers who are fully conditioned to expect to sign a service contract to get the "next big thing" — device included. "Make mine a dual-mode WiFi/cellular with multiple addressable handsets and a base station… metallic blue, please."

Like many past innovations for the mass market, consumer VoIP comes with great promise to deliver just the right thing at just the right time. Early VoIP service providers have the pioneering task of establishing a new approach to delivering telephony without disrupting the way the consumer uses basic services. In this more basic POTS replacement phase of the market, it is understandable that these offers would be priced at or below similar TDM services. The consumer has already placed a value on POTS voice whether over TDM or IP.

It is the delivery of incremental value that really counts. However, it is not always apparent how the consumer reveals his Implicit Value Profile, let alone how they apply to VoIP services. But the examples are all there: privacy, affinity/group preferences, "set-it/forget-it" simplicity, freedom-to-roam, and others. These all represent incremental value to the consumer, and the consumer will pay incremental dollars for "Value" over IP. This is what the next phase of consumer VoIP is ready to deliver. IT

Stan Little is vice president, Consumer Solutions at Sylantro Systems. For more information, please visit http://www.sylantro.com.

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Converged IP Multimedia Conferencing:

Making The Vision A Reality

We've seen dramatic increases in the pace of business over the past decade, and multimedia conferencing is a good way to speed communications while reducing travel costs. With IP networks, PCs on every desktop, and inexpensive Web cams, it will be fairly easy to give users voice, video, and application-sharing capabilities. But despite all the advances we've made in basic networking and communications technologies, the vision of multimedia conferences as mainstream business tools has remained just that: a vision for the future. Now, new technologies are solving some of the age-old problems that prevented companywide multimedia conferencing.

A Pervasive Conferencing Vision

Bell Labs displayed the first prototype PicturePhone in 1956, and since then companies and vendors have been trying to reap the benefits of long-distance voice and videoconferencing. Small wonder. When companies can have fully productive meetings at which people can see and hear each other, exchange documents, and collaborate on common projects without being physically together, they can eliminate travel costs, minimize the security and health risks associated with travel, and greatly speed the flow of information. While such collaborations are normally associated with conference rooms, the ultimate vision is a world in which anyone with a desktop or laptop PC can have a rich multimedia conference with anyone else, from any location.

But even though multimedia conferencing offers huge potential for business improvement, very few companies use it at all. Before we look at the barriers to pervasive multimedia conferencing, let's define just what we mean by it. Any type of person-to-person communication can be seen as a conference, whether via voice, e-mail, chat, shared applications, or video, but true multimedia conferencing should encompass all of these facilities. Basic features should include:

- One-to-one, one-to-many, and many-to-many meeting capabilities;
- High quality, real-time voice and video communication;
- Application sharing and collaborative document editing;
- File transfer;
- E-mail and chat; and
- Ad hoc as well as pre-scheduled conferences.

The conferencing system should integrate the above features, but it should also present them in a way that is easy to use on any PC, and which delivers business-quality voice and video over a wide range of network connection speeds. After all, the people who may most need multimedia conferencing links will be those in remote locations where network links may well run at sub-T1 or DSL (define - news - alert) speeds.

Traditional Solutions, Ongoing Barriers

When we look at the traditional approaches to providing conferencing services, the barriers to a pervasive solution become pretty obvious. Essentially, companies wanting to implement conferencing services have had three ways to go:

- A hardware-based, video-centric solution.
- A software-based, application-centric solution.
- An outsourced conferencing service

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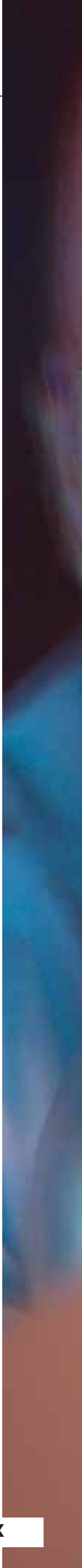
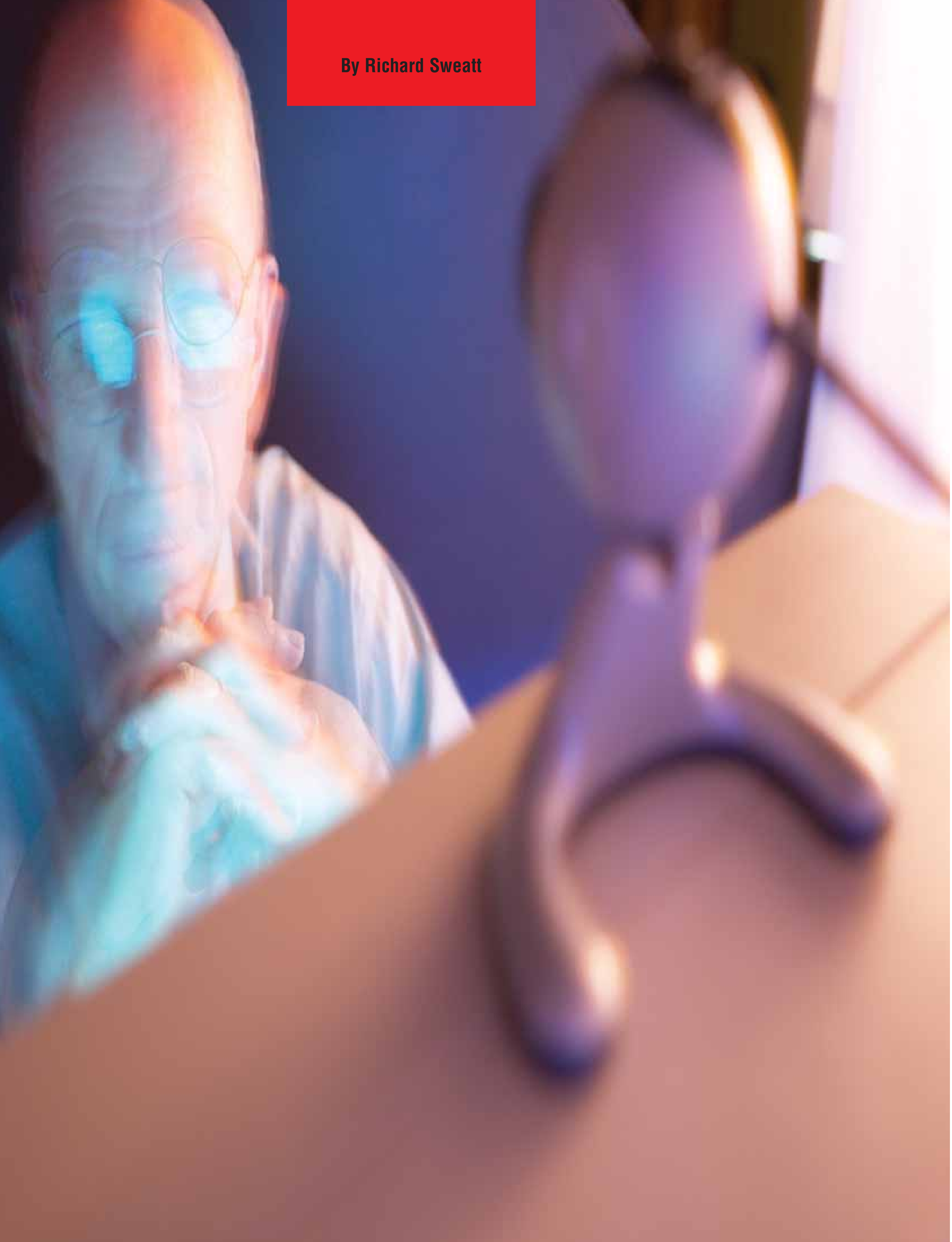


By Richard Sweatt



Multimedia conferencing must be available at every desktop, and users should be able to easily set up connections with one or several participants.

like WebEx.

Hardware-based videoconferencing has been around for decades, but it has always suffered from four basic problems: cost, complexity, integration, and quality. Together, these drawbacks have always made such conferencing seem like more trouble than it's worth. Hardware-based conferencing systems have traditionally been so expensive that even large companies could only afford a handful of them for conference rooms. Some vendors have lately rolled out scaled-down, workstation-sized varieties of their larger systems, but the cost is still too high for truly pervasive deployment.

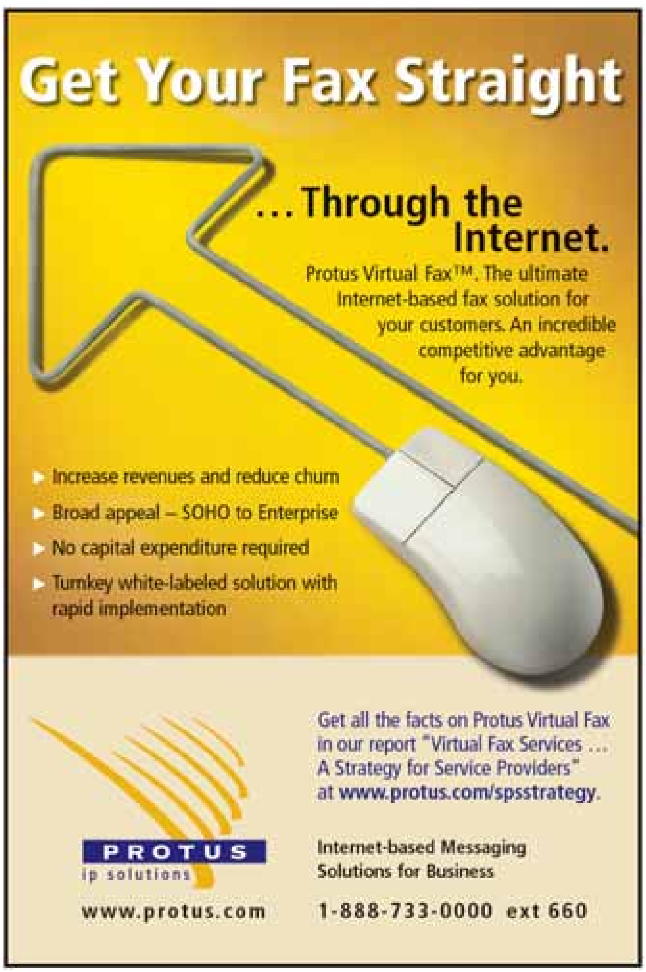
In addition, hardware-based systems have been a challenge to deploy and use. Conferencing servers have required specialized knowledge in protocols such as SIP and H.323, along with laborious integration with other network systems. Moreover, companies often have to increase their network's bandwidth to ensure smooth video transmissions. Few companies have been able to afford the dedicated, multi-megabit network links required for business-class video.

These systems have also been anything but intuitive to use. They suffer from call setup and control interfaces that looked almost nothing like the phone, e-mail, or instant messaging products employees are accustomed to. Typically, it's more trouble to set up and manage a conference that it's worth.

Another issue is that video-centric conferencing hardware typically has poor or non-existent application and data sharing features, so users are left to juggle the video/audio system and a separate PC — hardly a recipe for productive meetings.

More recently, products like Microsoft NetMeeting and LiveMeeting have promised to enable simple and pervasive conferencing through widely available software used with PC-mounted headsets and Web cameras. But while such software is inexpensive, it excels primarily at data sharing, and has a poor record when it comes to delivering quality audio and video. In addition, these systems are optimized for one-to-one conferences rather than larger meetings.

With the advent of the Web ten years ago, companies like WebEx have risen to prominence by enabling real-time meetings via ordinary Web browsers. By simply equipping users with Web cameras and headsets, these services eliminate the deployment headaches of hardware-based systems. However, the per-user, per-minute cost models for these services make them prohibitively expensive for broad organizational use.



Solution Requirements

With these challenges in mind, let's consider the requirements for truly universal, truly converged IP multimedia conferencing facilities.

Quality — To be truly universal, a multimedia conferencing system should support business-quality video, voice, and application sharing over a broad range of connections. The conferencing system should be able to dynamically adjust audio and video compression to compensate for the specific connection resources available to each conference participant.

Scalability — To become an integral part of normal communications, multimedia conferencing must be available at every desktop, and users should be able to easily set up connections with one or several participants. Even with adequate network bandwidth on tap to support dozens of simultaneous conference participants, most collaboration servers quickly become overloaded because they run on off-the-shelf hardware that was never designed to handle the demands of processing and routing multimedia traffic streams.

Deployment Simplicity — Hardware-based conferencing systems should not require the IT department to learn the ins and outs of multimedia networking protocols. IT should be able to consider multimedia conferencing as a discrete service that plugs into its existing network infrastructure with a minimum of hassle and reconfiguration. In addition, the hardware system should integrate with and leverage existing corporate user directories, groupware calendaring functions, and other features to eliminate duplication of systems and management effort.

Ease of Use — Ideally, starting, scheduling, and participating in a multimedia conference should be as intuitive as making a phone call, sending an instant message, or composing e-mail. Access to these services is as simple as point-and-click from your existing IP services, and now those existing IP services are multimedia enabled.

Cost-Effectiveness — For ubiquitous deployment on user desktops, the cost must come down to less than a few hundred dollars per user in a large scale network deployment.

New Solutions Drive New Vision

Today, these long-time barriers to productive multimedia collaboration are falling as vendors develop purpose-built systems to address them. Here are some of the most recent developments:

Multimedia Routing Servers

Vendors are now introducing a new generation of multimedia routing servers that are specifically designed to process, route, and compress multimedia traffic streams and distribute them to hundreds or thousands of endpoints. Rather than requiring IT staff to implement multimedia processing on standard servers and existing IP routers, these new routers are turnkey products that overlay existing network systems and internally handle all of the SIP (define - news - alert) and H.323 (define - news - alert) processing, video and voice compression, unicast and multicast requirements for conferences.

In addition, new multimedia routing servers can support quality video and audio by dynamically adjusting compression and each end user's video window sizes on an individual basis, depending on the speed of each user's connection. This ensures that every user will have a quality conferencing experience rather than jerky video or dropped calls.

Conference Provisioning Servers

Considerations For Conferencing And Collaboration Technology Purchases

Purchasing conferencing solutions is not what it used to be. Instead of the silo'd approach of the past when you purchased, deployed, and managed your voice conferencing separately from your video conferencing, separately from your data conferencing, IP now allows you to integrate them all together saving money, enhancing reliability, and making the experience seamless and intuitive for end users. Known as unified collaborative communications, these converged solutions allow presenters and participants to choose and combine modes of conferencing or collaboration that are most effective or convenient at the time (i.e., can support voice, video, and Web in the same conference in any combination). While these are good things in terms of the value and productivity-enhancing power that they bring to a business, they also require customers have a strategy to ensure they reap the maximum benefit and long-term value of the technology they have already installed as well as the new solutions they are researching to add.

Important Considerations:

What is the business need?

The business need should be the driving force of the purchasing decision. Companies should consider what they want the technology to accomplish and who they want to communicate/collaborate with, and whether those people are inside or outside the corporate firewall. The different types of solutions (voice, video, Web, and data) are ideal for different types of applications. For example, a video conference is ideal for face-to-face interaction, whereas a Web conference might be better suited for field training.

Will this technology work in my current (and future) network environment?

Customers should consider their existing (and planned) network environment. IP-based solutions can help a company maximize the ROI of their existing IP network investment. If a company has deployed, or is planning to deploy VoIP (define - news - alert), it should consider solutions that may interoperate with its VoIP platform, after all once a network is prepared for voice over IP, it is relatively simple to add real-time video. For example, some vendors offer standards-based VoIP phones that have been certified to work in different environments to provide integrated video telephony solutions from the desktop. Conferencing and collaboration solutions that are based on standards and that provide migration paths to future telephony initiatives will enhance the overall ROI of the investment.

Who do I want to collaborate with?

For technologies like video conferencing, conferencing with customers and partners outside your corporate firewall requires a strategy to ensure ease of use. Solutions are available through "bridges" or MCUs that provide gatekeeper functionality and through specific Network Address Translation (NAT)/firewall traversal solutions. NAT/firewall traversal solutions should be

To support broad scaling of collaboration facilities through large organizations, vendors are offering provisioning servers that handle the configuration and monitoring of multimedia routing servers and end user stations. These provisioning servers also maintain user address books and conference schedules via integration with Microsoft Outlook and other groupware programs, and they collect and store conference detail records for billing.

End User Systems

To drive down the cost of desktop conferencing and improve ease of use, the latest generation of multimedia conferencing systems leverages the native resources of desktop PCs to the fullest. These software-based products integrate with standard Web cameras and headsets, and allow users to schedule conferences using Outlook or other groupware tools. Users can begin conferences on-the-fly by selecting other users from a buddy list, and then view other users' video in separate windows on the PC screen. Application sharing is equally transparent, requiring users to simply open the application they wish to share.

Because they're software based, these end stations can be installed within fifteen minutes. Thanks to superior interface design and close integration with existing PC applications, calendars, and e-mail systems, users can learn to use the collaboration features them within minutes as well. With this level of transparency and price points of a few hundred dollars per user, these systems make pervasive desktop conferencing a true possibility.

The history of multimedia conferencing has been one of frustration, of a tantalizing vision of the benefits marred by the expensive and unwieldy reality of implementation. But today's new generation of conferencing systems brings new capabilities to the table, and may well be the catalyst for a new era of IP-based collaborative communication. IT

Richard Sweatt is vice president, sales and marketing at Amity Systems. For more information, please visit the company online at http://www.amitysystems.com.

Individual features may be important, however, savvy customers should look at the entire solution in context of their business needs.

based on standards and work with existing network systems to ensure they maintain network security. Some approaches like "tunneling" can actually create vulnerabilities.

Which interface is best for employees?

There are two primary user interfaces customers should consider — the telephony approach that centers around a phone (hard or soft), and PC platforms that provide an instant messaging (IM) interface with presence. Some vendors are tying these together by providing soft phone options that also include IM with presence. The phone approach is intuitive and enables users to leverage presence capabilities and to even migrate a voice call to video with the touch of a button. The other approach leverages the collaboration capabilities of the PC to create an environment in which users can launch a voice, video and/or Web collaboration session directly from their IM client.

Customers should evaluate solutions based on their existing and planned infrastructure needs and also the type of conferencing and collaboration interface that is best for employees — the phone or the PC.

How will I manage this on my network?

Conferencing and collaboration technologies have matured and established providers offer integrated scheduling and management solutions that help IT managers remotely deploy, provision, and troubleshoot devices on their network. Scheduling solutions offer device and resource scheduling (even bandwidth reservation) through the web or workgroup applications like Microsoft Outlook and IBM(quote - news - alert) Lotus Notes. Solutions should support management and scheduling of all standards-based endpoints to provide flexibility for the future.

What about the latest "features, speeds, and feeds"?

Individual features may be important, however, savvy customers should look at the entire solution in context of their business needs and network environment. Customers should evaluate a vendor's track record for providing support for new standards (like H.264 and H.239) in a timely manner, but in context of the entire solution. Customers should also consider the average life expectancy of a system or solution including whether vendors are able to deliver backwards compatible support for major new standards and features (such as SIP support) without requiring a hardware or complete system change. This will help customers determine which vendors offer the best lifetime ROI.

The Future Is Bright

The good news for customers is that conferencing and collaboration solutions are more powerful and more cost effective than ever. At the same time, the technology has matured and should be considered within the context of a company's overall network environment, which will help ensure long term benefits, value and ROI of any investment. IT

Steve Huey is chief marketing officer at Polycom.

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Looking Ahead:

VoIP Moves To Video And Wireless

2004 saw VoIP establish itself as a viable technology that is here to stay. In numbers larger than ever before, enterprises adopted VoIP for both new deployments and retrofits to traditional PBXs. Significant numbers of residential users subscribed to VoIP services through providers such as AT&T (quote -news -alert), Vonage,(news -alert) and Skype (news - alert). Much of this growth was driven by the cost advantages enabled by VoIP. However, for VoIP to truly reach the point of mass market adoption, and for the market to continue as a viable one for the service providers involved, it must move beyond "cheap long distance" and deliver upon the promise of advanced services. 2005 promises to be the year when the technology becomes available that enables these advanced services; for example, seamless VoIP over wireless and personal video conferencing are both on the cusp of moving from experimentation to viable commercial solutions.

Voice Over WiFi

WiFi has become ubiquitous in both residential and enterprise environments. This fact allows an increasing number of applications to take part in the mobile, networked world. Voice-over-IP becomes a natural technology to ride the wireless wave. While both residential and enterprise users can benefit from VoWiFi (voice-over-WiFi), each realizes a different set of benefits of this emerging technology.

Near term solutions for wireless VoIP within the home may take the form of cordless or DECT (define - news - alert) enabled handsets with an integrated ATA, Bluetooth connected handsets, or true VoWiFi handset, where VoIP connectivity is provided via WiFi. VoWiFi allows VoIP to move from the PC or a telephone connected to an Analog Terminal Adapter (ATA)(define - news - alert) , and become more integrated into our day-to-day lives. Whichever form of wireless enabled VoIP is utilized in the immediate future, these devices provide the benefit of mobility to the residential user, strengthening the usability factor of VoIP within the home.

Enterprise users also benefit from mobility, and in fact this may be one of the strongest drivers to a return on investment for VoWiFi use in the enterprise environment. The ability to contact individuals within the workplace on a single mobile device can eliminate costly paging or cellular charges when workers are within the reach of the enterprise's wireless network. In the enterprise environment, the VoWiFi device may need to take a slightly different form than that in the home. Support for "Presence" is a stronger requirement in an enterprise VoWiFi phone so that the user could indicate his or her ability to accept calls at a given time — anything from a minor benefit to an absolute necessity, depending upon one's individual tolerance for workplace meeting etiquette.

For both the residential and the enterprise user, the VoWiFi "Holy Grail" is that of a converged VoWiFi and Cellular device — the "dual-mode" phone that can seamlessly roam between WiFi and cellular networks. By integrating a VoWiFi and cellular phone into a single device, users have one less gadget to carry and one less battery charger (and car charger) to keep up with. But can this single converged device live up to our desire of a

Go To Table of Contents | Go To Ad Index

By Michael Ward

true unified device? In the immediate term, these devices will be able to indicate the presence of cellular and WiFi networks and allow the user to select which network to place a call. However, the ultimate goal of seamlessly roaming from WiFi (define - news - alert) to cellular and back may still be a way off. However, progress is being made today toward enabling this level of mobility, and 2005 will see a number of early-stage solutions deployed to move toward the reality of seamless mobility.

The ability to place calls over a wireless VoIP handset wherever a WiFi network exists is creating concerns of lost cellular revenue amongst certain cellular carriers. However, ironically, it is perhaps this concern that will serve to accelerate the ability to seamlessly roam across WiFi and cellular networks. The more progressive carriers will recognize the inevitability of VoWiFi and move to create business models that actually embrace this technology as an extension to their cellular networks. Enterprise and residential customers alike could then benefit from a truly converged device. If embraced rather than feared, carriers can actually leverage VoWiFi to increase revenues and begin to offer additional services that take advantage of the increased data bandwidth that exists within WiFi networks.

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Personal Video Conferencing

While IP-based video conferencing solutions via ISDN (define - news - alert) or other dedicated network links have existed for some time in the enterprise domain, the availability of this technology for the individual user has not been either viable or affordable. However, as the saying goes, "a picture is worth a thousand words," and the ability to communicate with others both visually and audibly conveys so much more than a voice-only conversation. Time and time again, the promise of the video phone has arisen, but failed to deliver. Blocky, stuttered video was tolerated only long enough for the novelty to fade. However, as broadband becomes pervasive and audio and video encoding technologies evolve, there now exists the ability to provide high-quality, full-motion video conferencing to the individual user for a reasonable cost.

When examined from a U.S.-centric perspective, personal video conferencing or communicating via an IP video phone as a daily occurrence are questionable, in large part because of privacy issues. However, when viewed from a global perspective, one can look to the rapid adoption of cellular camera phones and video services in Japan and parts of Europe to recognize that a high-quality personal video conference service may be just the "killer application" that can accelerate the adoption of VoIP and perhaps even more rapidly accelerate the proliferation of broadband worldwide. Trends in the U.S. cellular market, with recent announcements from major carriers regarding the rollout of mobile TV and the increased number of handsets with both video playback and record capabilities, lend credence to the view that our own inhibitions about video-based communications may be easing.

Perhaps one of the most important aspects of video- and voice-over-IP

Progressive carriers will recognize the inevitability of VoWiFi and move to embrace this technology.

(V2IP) is the fact that it enables a new set of revenue-generating services to be provided to the consumer. If VoIP is to reach its full potential, extending beyond "free long distance," the industry must deliver services that consumers are willing to pay a premium for. Many indications are that personal video conferencing may be just that service. VoIP service providers, such as Packet8, are marketing a video phone service for a $10 premium over the standard voice-only service. Telecom providers worldwide are beginning to roll out video phone services — France Telecom is but one example of a telephony provider rolling out video phone service, with more expected to launch services throughout 2005. While the initial round of video phones is taking the form of traditional wired desktop phones, one can look for personal video terminals to begin to take more innovative forms, including wireless versions.

Enabling Technology Trends

A number of technologies are converging to enable both voice-over-wireless and personal video conferencing to be technically feasible and accessible to the consumer. These applications have a number of challenges that must be addressed to drive these devices to full adoption. The portable devices among this mix must also deliver standby and talk-times on a par with the traditional devices that they are replacing. As these devices play into a very dynamic market, they must be designed with sufficient flexibility to accommodate changing and emerging standards, as well as deliver any new functionality desired by consumers. Perhaps the largest challenge is one of reaching cost targets that allow these devices to be readily purchased by the mass market.

Changes in the base technology architectures used to develop these devices are occurring. Traditionally, VoIP endpoints, including VoWiFi handsets and video phones, utilized architectures that included both digital signal processors (DSPs) and applications processors. The DSPs handled the "heavy lifting" of VoIP — the transformation of voice into packet data and back again — while the applications processors handled the signaling stacks (such as SIP or H.323), the device's user interface and other control applications. As pressure has mounted to deploy VoIP- or V2IP-enabled devices in smaller and more cost-effective solutions that help carriers increase revenues while decreasing customer acquisition costs, the need to explore alternative architectures has arisen.

A solution to the need to reduce system cost and footprint is to leverage the increasing processing power of the applications processor to handle all aspects of the VoIP or V2IP device. By running the audio-to-packet conversion along with the call control and user interface software on the applications processor, the system designer can remove the DSP and reduce both cost and footprint from the product, while increasing battery life in mobile designs. This "DSP-free" model simplifies the design for the OEM, allowing more rapid development and deployment of products, which results in a quicker ability for service providers to begin to capitalize on the new revenues streams enabled by these devices.

As VoIP continues to gain momentum, it will do so through the growth and adoption of devices such as voice-over-WiFi handsets and personal video terminals. It is devices such as these that will allow providers to deliver new and innovative revenue-generating services. The mobility and personal connectivity enabled by these devices will allow a new level of productivity to be realized by enterprises, while at the same time reducing their overall telephony expenditures. The technology exists today that enables the development of these products, which will in turn help to invigorate the VoIP market as a whole. IT

Michael Ward is director of product line management at Trinity Convergence. For more information, please visit http://www.trinityconvergence.com.

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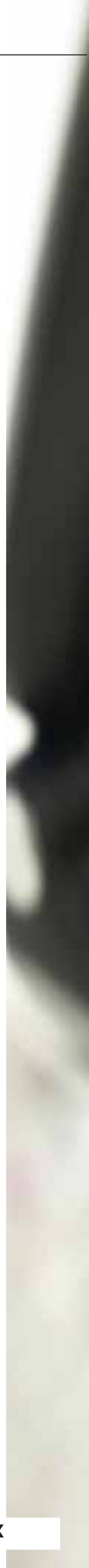

Converged Wi-Fi / Cellular Voice:

Ready For Residential Market

Telecommunications as we know it today may be on a verge of a crisis, that is, the crisis of complexity. Multiple types of network access — voice and data, fixed and mobile, wireless and wireline — present the user with an increasingly hard to manage variety of communications options. Add to the picture new communications methods — Instant Messaging, Push-to-talk, SMS (define - news - alert), video telephony — and multiple devices needed for connectivity — laptops, traditional and IP phones, Blackberries, cellular phones and Wi-Fi-enabled PDAs — and it becomes easy to appreciate the degree of user frustration. Therefore, it is imperative for the service provider looking for continual growth in their network usage and revenues, to deploy the technology unifying their service offerings and drastically streamlining the user experience.

Imagine being able to *simplify* your communications experience while at home, at work, or traveling, with a single access-independent service and one device accessible via a single number. This vision promises the return of connectivity convenience even as communication methods continue to proliferate. The good news is that the technology making this new vision a reality is being rapidly developed by the industry based on the recently completed standards such as IP multimedia subsystem (IMS)(define - news - alert) and Unlicensed Mobile Access (UMA) (define - news - alert). This technology promises access-independent converged service — such as a combination of voice over Wi-Fi (VoWi-Fi), broadband VoIP, and cellular telephony — now increasingly referred to as *Dual Mode* VoWi-Fi/cellular service.

Helping the rapid introduction of this technology is the fact that the end-user devices and the majority of infrastructure components capable of supporting such functionality already exist. The recipe is seemingly simple. Take a latest PDA-based cellular phone add a Wi-Fi card, throw in proper client software and *voila*! You've got a dual mode device. In reality, however, the main complexity of offering such service lies in both the end-user device software and the core network, where the effort required to properly architect and implement such services is quite significant.

Dual Mode Consumer Service: Luxury Or Necessity?

Service providers around the world more and more consider the proliferation of Wi-Fi networking and widening availability of residential broadband data service as a chance to both offer new attractive services and complement their existing infrastructures. They are not discouraged by growing pains of VoWi-Fi technology expected to be resolved in coming years. In fact some of them can even be tackled by careful selection of addressable market segments. For example, many of the limitations of VoWi-Fi, such as fast handoffs between Access Points, security, and capacity, may not be as critical in the residential networks, making this market an ideal proving ground for trial deployments and limited rollouts of dual-mode service.

But what makes dual mode so attractive for service providers? First, the need to address consumer demands in converged communications. With dual mode VoWi-Fi/Cellular handsets

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By Alex Shneyderman

Dual mode service appears to be very attractive to residential customers.

becoming available in 2005 and affordable by the end of 2006, the consumers will expect the services making use of their handset capability — for example allowing them to use the same device and number for both cellular and in-house cordless service. Such offerings will allow wireless operators to better address the competitive threats of low-cost independent VoIP operators. The second benefit of such service would be an offload of off-pick cellular minutes to lees expensive broadband VoIP network. Finally, dual mode service will allow for the cellular features — such as Push-To-Talk and SMS — to be offered to broadband VoIP consumers.

Dual mode service also appears to be very attractive to residential customers. Most importantly it makes the communications experience a whole lot more convenient and user friendly while preserving the existing familiar user experience and service models. The consumer would have to purchase — and also charge, maintain, and learn how to use — fewer devices and take care of one bill for all voice services. That would directly lead to both service and equipment cost savings making the dual mode service offerings even more attractive. Additionally, such service would address indoors coverage issues and resulting inadequate cellular service quality.

With the need for a consumer dual mode service thus clearly identified, the majority of the service providers today either have definite plans or already took initial steps to introduce dual-mode service to their residential customers. The deployment through enhancement to the existing VoIP infrastructures, currently favored by the majority of operators, promises to minimize the initial investments and guarantee the service stability and reliability from both support and customer satisfaction perspective.

The Dual Nature Of Dual Mode

From a strategic perspective, enabling real-time voice communications over the same network that handles data transmission is increasingly viewed by service providers as a key long-term advantage and an important stepping stone to next-generation infrastructure, such as IMS. Offering it, however, will require complex converged architectures encompassing VoIP domain seamlessly integrated with cellular domain, significant investments in dual-mode handset programs, and creating new alliances and partnerships for joint development, billing reconciliation, and multi-access roaming.

While residential dual mode service can be based on a variety of technologies, among the most promising are the two standards-based approaches built around IMS (define - news - alert) and SIP-based VoIP, and UMA recently standardized as a convergence standard for GSM systems. Both approaches make VoWi-Fi calls appear as cellular calls to the network "hiding" the Wi-Fi access media and signaling from the network. However, that's where the similarity ends. While the SIP-based VoIP approach places the interpretation and mapping functionality in the core network in the form of media and signaling translation gateways and allows for the majority of the traffic to be offloaded from the cellular network, the UMA standard enables the Wi-Fi infrastructure with the base station controller making Wi-Fi appear as just another

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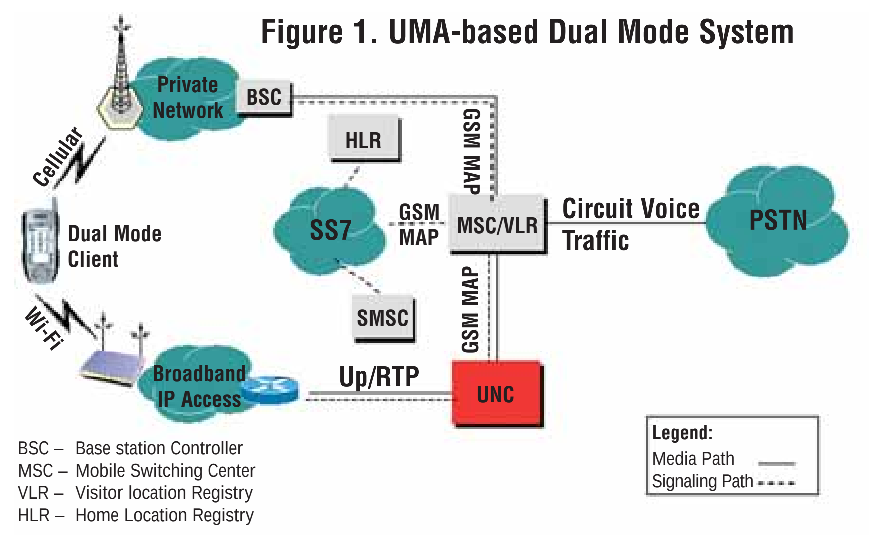
Figure 1. UMA-based Dual Mode System



air interface and requiring all traffic to traverse GSM network.

UMA Approach

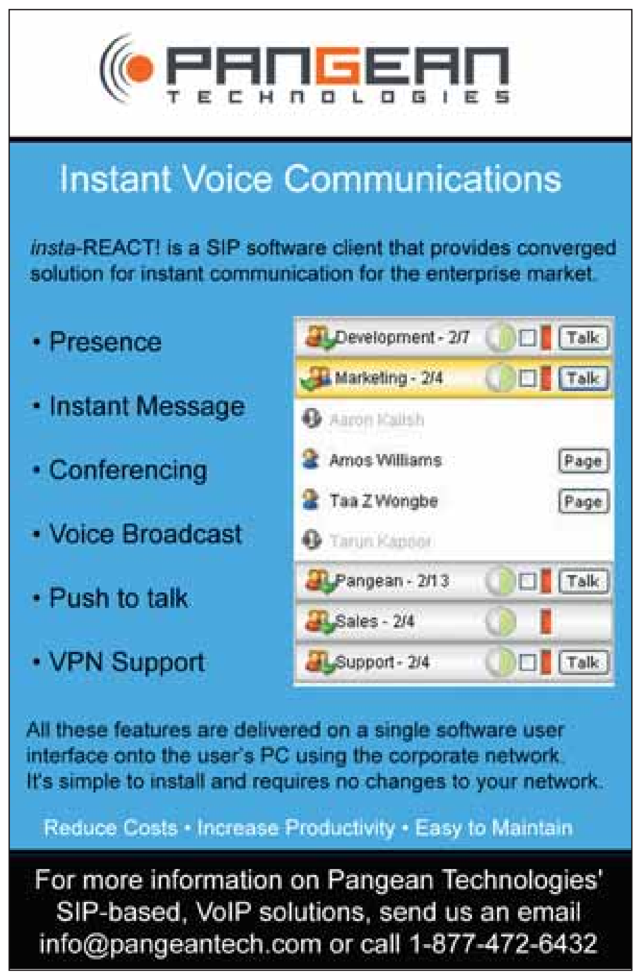
The UMA specifications enable GSM/GPRS handsets equipped with Wi-Fi or Bluetooth radio to access GSM service over arbitrary combination of Wi-Fi and broadband access infrastructure such as hotspots, and home networks. This approach essentially extends the GSM/GPRS services over the Wi-Fi air interface through a blend of VoIP technology and UMA-defined tunneling and signaling protocols. UMA is a GSM-specific technology and therefore can be used only in conjunction with the GSM/GPRS (define - news - alert), GSM EDGE Radio Access Network (GERAN), and Universal Terrestrial Radio Access Network (UTRAN) cellular systems defined by 3GPP.

To achieve the goal of seamless integration with 3GPP networks the UMA defines the UMA Network (UMAN) radio access network based on Wi-Fi, requiring broadband and Wi-Fi infrastructure and terminals to support GSM and GERAN defined signaling and media. This objective is accomplished in UMA by introduction of a new element, UMA Network Controller (UNC) merging the IP and cellular domains, and a new protocol, Up interface (Figure 1).

The UNC terminates RTP media streams and GSM signaling tunneled over the Up interface to and from GSM terminal equipped with Wi-Fi radio and appropriate software, essentially appearing to the core network as a GERAN base station controller. The UNC can be either located within Wi-Fi infrastructure in enterprises and hotspots or owned by wireless carriers or broadband access providers to offer *GSM over Wi-Fi* service to residential customers.

By supporting GSM voice and services over Wi-Fi or Bluetooth UMA provides logical extension to the existing GSM-based systems requiring relatively small investment from carriers. However, UMA specifications do circumvent the VoIP-based IMS, which is gaining widespread support among both wireless and broadband operators as a standard for next-generation networks. Other potential drawbacks of UMA are its limited usefulness with other cellular technologies and the need for new types of terminals, likely more costly then comparable SIP-based VoIP devices.

IMS And SIP-Based VoIP Approach



The SIP-based approach is not limited to GSM.

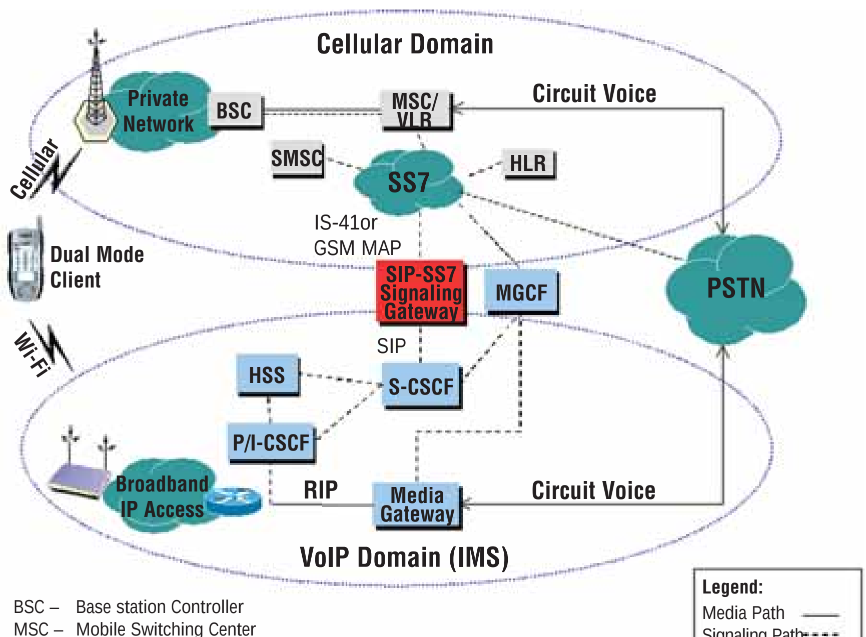
This approach is based on the combination of VoWi-Fi access and traditional SIP-based VoIP broadband technology defined by Packet Cable and IMS specifications. In addition proprietary architectures and components mapping VoIP signaling and RTP media into cellular circuit voice protocols need to be implemented. This type of dual mode service can be offered to any residential subscriber with access to a broadband connection, Wi-Fi network, and off-the-shelf handsets equipped with Wi-Fi and cellular radios, VoIP clients, and dual mode software.

The SIP-based approach is not limited to GSM and can be used with any cellular system albeit at the expense of likely customization. The bulk of VoIP media and signaling traffic can be routed directly to PSTN gateways or other VoIP networks freeing expensive cellular core network capacity. The call routing, establishing of RTP media paths and cellular voice telephony features are all supported in VoIP domain making the end user device appear an extension to cellular network (Figure 2). The mapping of VoIP traffic into circuit domain is accomplished by Wi-Fi/Cellular gateway serving as a hub for both SS7 and SIP signaling allowing for call establishment between different types of terminals and handling of services and calls placed to VoWi-Fi clients through cellular infrastructure.

The VoWi-Fi/Cellular Gateway translates and maps the SS7 messaging, such as ANSI-41 for CDMA (define - news - alert) and GSM MAP for GSM-based cellular systems to SIP messages and visa versa, treating VoWi-Fi calls as cellular calls.

The SIP-based VoIP approach provides truly universal architecture not only enabling the support for dual mode service in the majority of cellular systems but also providing an evolution path towards IMS compliant infrastructure. Unlike in UMA, the bulk of the calls here is handled entirely within VoIP domain offloading traffic from cellular infrastructure. Moreover this approach is equally appealing to broadband service providers and MVNOs (define - news - alert)providing for easier partnering with cellular carriers and offering logical next step on the way to the next generation networking. That makes SIP-based dual mode service a perfect vehicle for establishing service relationships between cellular and broadband operators, which both can combine their existing infrastructures to provide converged service.

Figure 2. SIP-based Dual Mode System



BSC – Base station Controller
MSC – Mobile Switching Center
VLR – Visitor location Registry
HLR – Home Location Registry
CSCF – Call/Session control function
MGCF – Media Gateway Control Function
HSS – Home Subscriber Server

Is It For Real?

So what can we say about the technology which has significant consumer market appeal, offers clear business case to service providers, can serve as a logical foundation for the Next Generation Networks and mostly relies on the existing proven technology? It is likely to start happening very soon! Service providers with the existing cellular and broadband VoIP networks and alliances between broadband and cellular network operators are in the best position to offer dual mode services to their residential customers early on.

We believe that the operators carefully selecting experienced vendors will be able to successfully address the obstacles to deployment of converged services such as Wi-Fi technological immaturity, potential architectural complexity and the challenge of integration of different network infrastructures. IT

Alex Shneyderman is product manager, BSABU, at Cisco Systems. For more information, please visit http://www.cisco.com

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Voice Security In An IP World

Security may be among the most obvious concerns for those considering the deployment of voice over IP (VoIP) communications for their business. Beyond concerns about service quality and stability that have been largely addressed by vendor product improvements over the past few years, firms are more acutely aware of the service disruptions and the financial impact of security threats. VoIP and broader IP telephony application technologies can reduce voice costs, enhance staff productivity, increase the utility of IP networks and improve end user experience. They do also, at least in the voice world, raise the stakes on security. In particular, VoIP challenges firms to rethink the implications of voice Confidentiality, Integrity and Availability (CIA).

Traditional voice networks have had to address toll fraud, misuse and switch stability issues over the years, however most have found their traditional voice networks to be a haven relatively unaffected by viruses, malicious codes, and operating system vulnerabilities. Not so in a VoIP (define - news - alert) network. VoIP and IP telephony (IPT) environments are subject to all the security threats that impact IP networking environments.

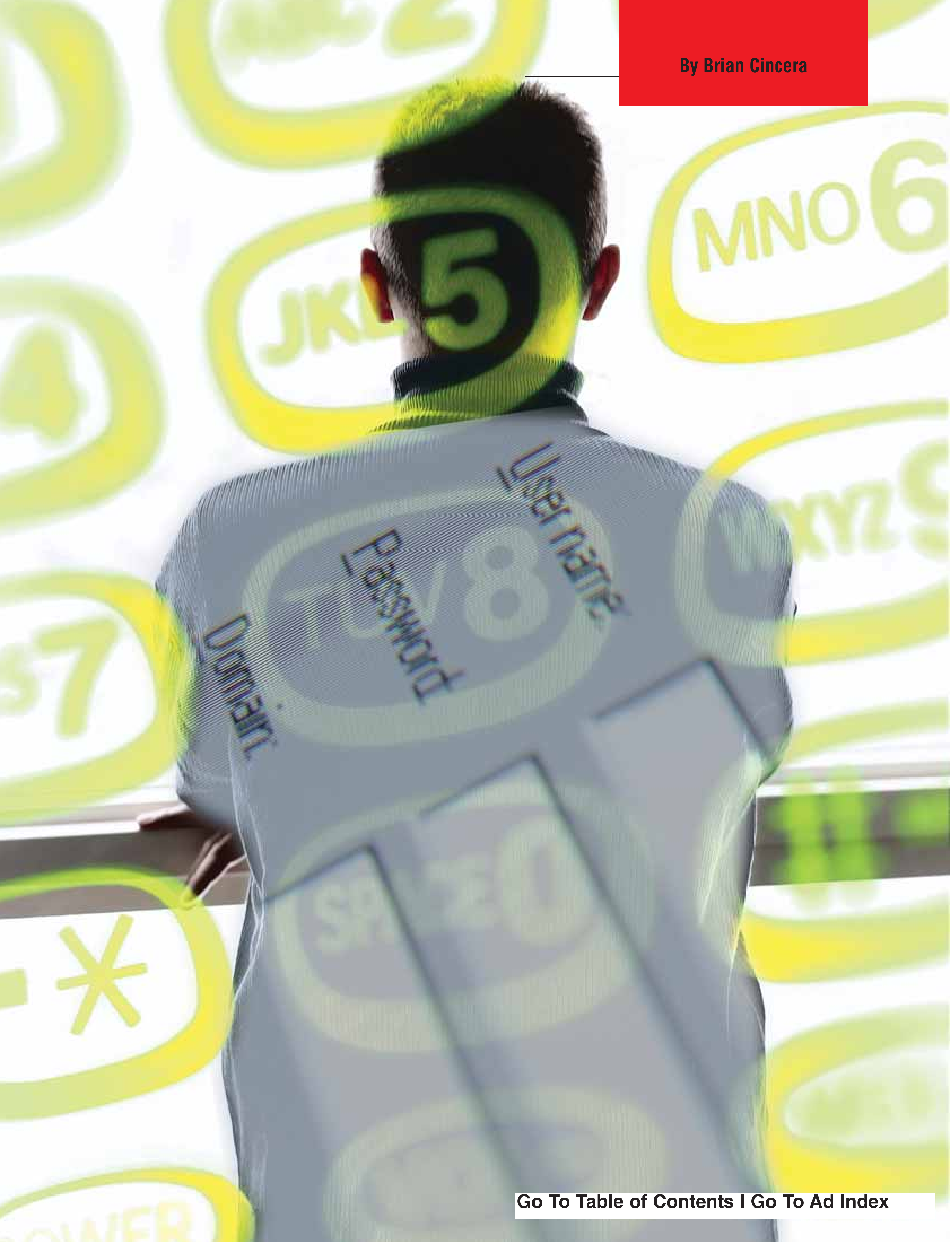
Confidentiality threats affect VoIP communications in much the same way as they affect e-mail on network. An attacker on a LAN might record a sensitive voice call by capturing and reassembling packets. Rather than needing to physically tap a phone circuit, an attacker needs only a packet sniffer to intercept a VoIP phone conversation. Reconstructing the packets could produce the entire call in a form suitable for posting to the Internet or e-mailing to a third party.

Integrity threats impact the content of a VoIP call or the call management system. However unlikely, in theory, a man in the middle attacker could intercept voice packets, alter them in some way and then forward them to the intended destination. While replacing selected words in a live conversation seems implausible, other IPT (define - news - alert) applications like time and attendance tracking performed on an IP phone are subject to integrity attacks where data input by the sender is altered in transit to its destination.

Availability attacks are often the most concerning to businesses deploying VoIP. Denial of service attacks, buffer overflows, viruses, and malicious codes can add delay or jitter to IP calls that render them unintelligible. The same attacks can disable individual phones, voicemail servers or the central call manager servers. Without a call manager, an IP voice system is completely unusable. Availability concerns rightly rank highest among those looking to protect their VoIP investments.

The threats to CIA are rooted in the very nature of IP communications. When we packetize a voice phone conversation, we expose that conversation (now the contents of IP packets) to all the threats that affect any Internet Protocol communication. The bad news is that to the malicious hacker, unwitting end user or application flaw, a VoIP packet is no different than one containing content from a Web page or one containing an e-mail from a friend. The good news is that many of the protections available to secure traditional IP networks are very effective at securing VoIP networks. With a few notable exceptions, VoIP security recommenda-

By Brian Cincera



tions mirror those that might be made to any firm for providing protections for their IP data infrastructure. For those seeking to deploy VoIP systems, here are a number of steps that can be taken to increase VoIP confidentiality, integrity and availability.

• *Engineer security into the service.* While security rarely trumps good functionality, when security is built into the IP voice system design it is typically easier to maintain and less disruptive of the functionality for which the system was deployed than if security has to be added later on. Voice applications increase the pressure for low latency on an IP network, at the same time that they decrease the tolerance for unavailability. A 500 millisecond latency in an account record lookup is an annoyance. 500ms of latency in a voice call makes people hang up. (C,I,A)

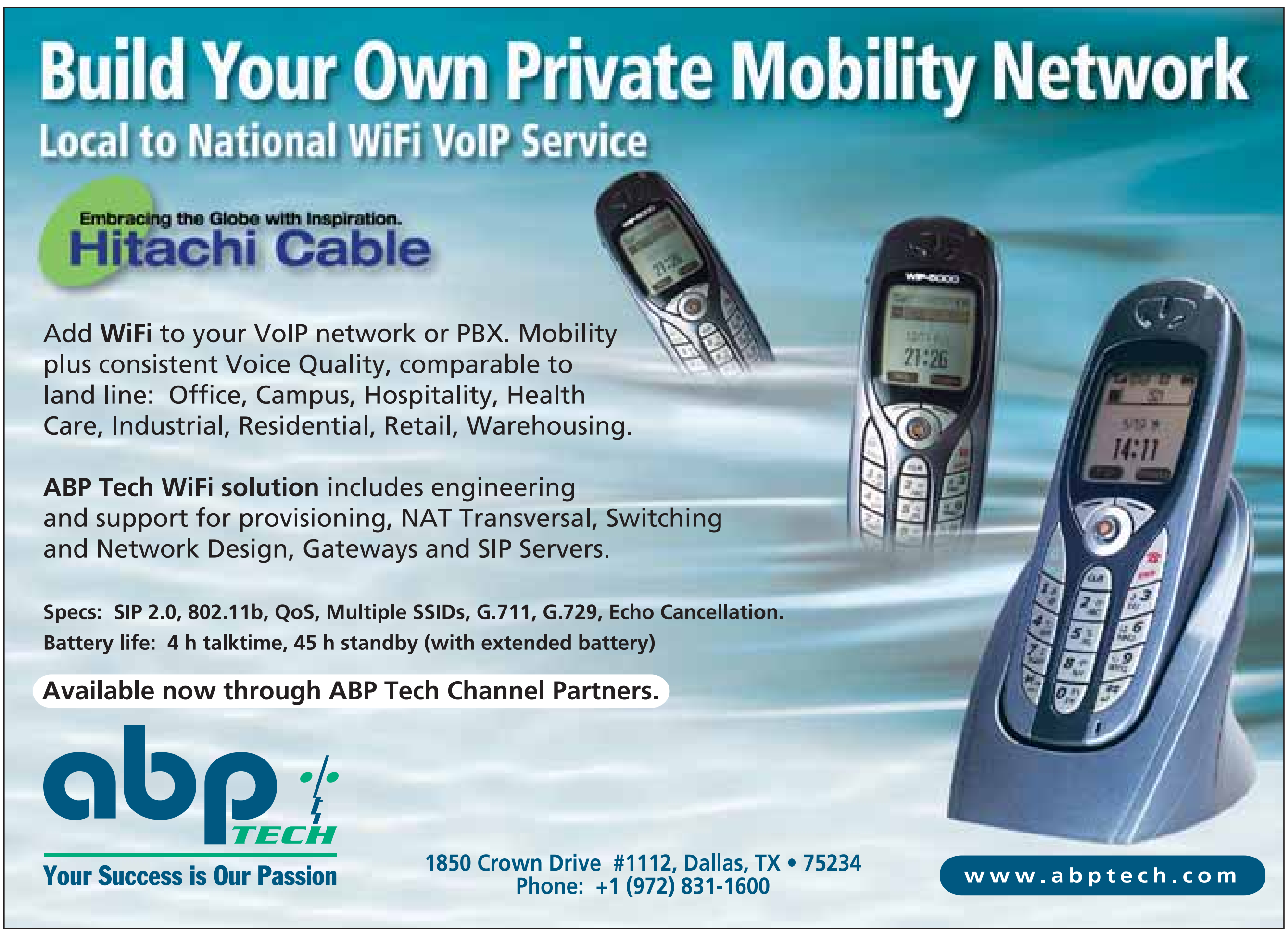
• *Keep voice and traditional data on separate LANs.* This potentially means more wires to each desk area and it means disabling the PC port commonly found on IP phones. Commingling voice and data traffic makes it harder to implement other voice security recommendations (see below) and increases that chances that data traffic creates availability problems for voice calls. In a shared LAN, voice calls could easily be disrupted by bandwidth intensive applications like music downloads and file sharing. (A)

• *Encrypt VoIP traffic to the call manager*. While encryption increases the performance requirements of the call manager, it is the easiest way to decrease the chances that sensitive conversations from executive offices get intercepted, recorded, and replayed on the Internet. (C)

• *Harden the call manager and voice mail hosts.* While most vendors have already taken some effort to lock down the operating system of these critical devices, they are still general purpose operating systems that require maintenance, patching, and proper configuration. All the hardening rules that apply to any other business system server should apply to VoIP hosts. Close unnecessary ports and disable unneeded host operating system services. (I, A)

• *Set traffic rate limits that are below system performance limits.* VoIP traffic is very network sensitive. By setting appropriate traffic rate limits, denial of service attacks are less likely to be successful. The impact of capping the total number of simultaneous calls is far less than the impact of a system wide outage. (A)

• *Deploy redundant backend tech-*



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nologies. Router, switch, and call controller redundancy are critically important. Most businesses can survive a day's outage of an e-mail system. Few can recover the losses incurred from a day without phones. (A)

• *Separate management traffic from voice call traffic*. By separating the management traffic to its own network, it is less likely that the call manager could be attacked by an end user of the voice service. (A,I)

• *Authenticate phones, IPT devices, and users*. Device authentication assures that only legitimately deployed IP phones can use the voice service. Doing so addresses attacks and toll fraud from illegitimate PC-based soft phones. User authentication adds another layer of protection from "walk up" fraudulent use of a VoIP phone. Most users are unaccustomed to authenticating before using a phone, but firms that deploy sensitive applications to IPT phones like time and attendance often find user authentication is necessary. (A,I)

• *Deploy monitoring, alerting, auditing and response technologies and processes as on any other IP network*. Maintaining a secure posture includes dealing with inevitable vulnerabilities uncovered in computer technologies over time. Vigilant monitoring for breaches and new threats combined with well developed processes for addressing events and managing change increase the chances that security gaps are filled before the bad guys find them. (C,I,A)

• *Train administrators and end users, not only on how to use functionality, but also on the steps they can take to secure the voice over IP technologies they use*. A post-it note with a userID and password stuck to the phone can easily defeat the world's best security technology. When users understand the potential impact of their behavior on their business, most are all too happy to go the extra mile to protect the firm.

Voice over IP security raises everyone's game. For those accustomed to traditional voice networks, a whole new world of IP threats is unleashed on a normally stable voice environment. For those accustomed to traditional data networks, a whole new world of specialized applications and high-stakes availability emerges in an environment that works acceptably well despite availability gaps. Most planning and investment is focused on addressing availability concerns, with confidentiality and integrity ranking second and third, respectively. Successful and secure VoIP operations are those that address the specialized confidentiality, integrity, and availability concerns of voice communications in the context of an IP network. Successful users:

Understand the impact of voice on IP design, and they understand the threats of IP networking to voice communications, and they account for both in their design.

Build and configure technologies and applications with the understanding that service protection runs a close second to functionality in terms of importance.

Implement tools, processes and human training that weave security maintenance into the fabric of their organization's infrastructure operation.

When done properly, a secure VoIP network easily delivers on the promise of lowering voice call costs, minimizing fraud, expanding applications to end users, improving personal productivity and enhancing the end user experience. IT

The good news is that many of the protections available to secure traditional IP networks are very effective at securing VoIP networks.

Brian Cincera is vice president, Security Solutions, at Greenwich Technology Partners (GTP). For more information, please visit http://www.greenwichtech.com.

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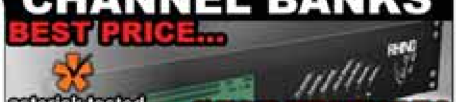

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CEO Spotlight

Dr. Elon A. Ganor
Chief Executive Officer and
Chairman of the Board of Directors
VocalTec Communications.

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 Dr. Elon A. Ganor, Chief Executive Officer and Chairman of the Board of Directors of VocalTec Communications.

GG: What is VocalTec's mission?

EG: VocalTec's (quote - news - alert) mission is to continue offering innovation and excellence in the changing telecommunications market. We will continue to successfully adapt to changes, in order to further company growth and take our place in the burgeoning VoIP (define - news - alert) industry.

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

EG: VoIP is finally maturing, perhaps it has taken a little longer than we originally thought, but it is nonetheless maturing, and VocalTec can now capitalize on this maturation process. VocalTec has a history of successfully reinventing itself, adapting to changing climates and market conditions. Over the course of the past year, we have done this again, in what we believe will be the most successful reinvention yet, and now offer a scalable, service-rich enterprise hosting and Class 5 alternative. Our current challenge is to encourage the market to recognize the new VocalTec, and learn more about its new offerings.

GG: Describe your view of the VoIP market today, and how it's changed in the past 10 years since VocalTec pioneered the industry with its Internet Phone?

EG: VoIP and Broadband are tied together. When VocalTec originally offered the market's first commercial offering, the challenge was then to offer reasonable quality voice over narrow bandwidth. Later in time the market took advantage of improved IP routing, providing arbitrage opportunities in the International Long Distance and Class 4 core markets.

Now, with the rapid deployment of Broadband to the home, VoIP is becoming a mature solution. The impact is dramatic. Within a few years the telephony of about twenty percent of developed countries will be transported end-to-end by packets. Voice will become just another, albeit important, service.

> **Voice will become "just another" service.**
>
> **-Elon Ganor**

GG: Now that it appears that growth and opportunity are once again the trends in the VoIP industry, what possible hurdles do you see that might upset this momentum?

EG: Regulation may assist incumbent carriers in slowing the process down — in countries such as Germany and Israel unbundling is still not occurring, and this of course slows the process down. However, while it does slow it down, it cannot stop the process from happening.

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

EG: True scalability and extraordinary efficiency of call control, over standard platforms is where VocalTec excels. Add to that the issue of security, where we also stand out. We believe that our excellence in these two areas will stand us in good stead in the near future, as networks with hundreds or even thousands of switches will be replaced with an efficient centralized mega-softswitch. Regarding security, Internet technology has already proven beyond doubt, that security is a crucial technology in the deployment of any packetized network.

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

EG: It is my view that voice will become "just another" service, part of the set of multimedia services to the home and to the enterprise. Voice will not be billed separately and per minute charges and long distance tolls will disappear. However voice will remain a "must-have" in any network. IT

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