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INTERNET TELEPHONY

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Speaking With... CompTel's Earl Comstock

Page 74

The VoIP Authority Since 1998™

VoIP INVESTMENT HEATS UP

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

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The VoIP Authority

By Greg Galitzine

What's Hot?

Conferencing & Collaboration

We've all heard Rich Tehrani talk about Just in Time Communications referring to the tremendous productivity gains we will see because of the latest communications technology. Two of the areas of most rapid growth will be conferencing and collaboration, and we're hearing hot news from companies in this space almost daily.

Skype (news - alert) is getting in on the action, by virtue of its recently announced relationship with Logitech. Logitech and Skype worked in concert to certify Logitech´s QuickCam Web cams and PC headsets for use with the newly announced Skype Video feature. The companies worked closely together on usability and performance, to ensure a fully synchronized, high-quality video and audio communications experience over the Internet. Skype Video, which includes a full-screen video mode, is an integrated part of Skype 2.0.

According to Frost and Sullivan research analyst Har Yen Yen, "With the ongoing transition from ISDN to IP, video conferencing is set to witness a greater convergence of applications as varied as instant messaging, streaming, and Web collaboration.

Factor in the proliferation of conferencing applications designed for 3G networks and what you get is a rapidly expanding market for conferencing and collaboration services.

It seems like conferencing and Web collaboration are on everybody's "hot list" these days. In fact, Inter-Tel is releasing a brand new Audio and Web Conferencing solution this month; I was lucky enough to get a sneak preview of their latest offering.

Essentially, this is a premise-based conferencing solution that allows users to schedule and attend remote meetings while mitigating the high costs usually associated with ASP-based conferencing solutions.

Here are just some of the features they told me about:

- Scales to 96 parties per session
- Two software licenses — audio only and audio and Web conferencing
- Software solution runs on a standard Linux server, using a SIP audio conference stream
- Integrates with Outlook e-mail, calendaring, and address book
- Provides robust conference session recording capabilities
- Integrated reporting tools capture all call activity for efficient management
- Has an outbound dialing feature, which eliminates the need for participants to remember dial-in information
- Users can schedule one-time, recurring, and reservation-less conferences.

Of course if you're interested in learning more about the future of conferencing and collaboration, you can choose to shed your winter doldrums and head down to sunny Ft. Lauderdale for the first major VoIP event of the year: Internet Telephony Conference & EXPO. The conference is taking place from January 24–27 and will feature a wealth of educational opportunities that you simply cannot find anywhere else. Included among the educational offerings is a Conferencing and Collaboration Summit, which will take place on Tuesday January 24. The Summit will feature sessions on the following topics:

- Introduction to Collaboration and Conferencing
- Adding Video to Enhance the Collaboration Experience
- Adding VoIP to Web Conferencing and Collaboration
- IP-based Collaboration for Teleworking
- Presence & Unified Communications, and more.

For more information or to register, please visit www.itexpo.com.

Greg

INTERNET TELEPHONY®
Volume 9/ Number 1 January 2006

Contents

118

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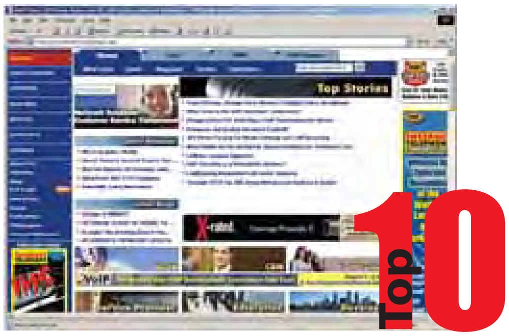
Contents



Top 10 Visitors to TMCnet.com (By U.S. City)

1. Herndon, Virginia
2. Marina Del Rey, California
3. Middletown, New Jersey
4. New York, New York
5. Washington, D.C.
6. San Jose, California
7. Cambridge, Massachusetts
8. Mountain View, California
9. Atlanta, Georgia
10. Boston, Massachusetts

QUOTE OF THE MONTH:

"In Mr. Whitacre's assessment (an assessment apparently shared by other Bell company CEOs), it is not enough that their companies get paid for the service they actually provide — namely, transmission. Rather, they believe that they should get some or all of the value of the content that is transmitted over their networks. That is what Congress should not allow to happen — transmission providers, whether incumbents or competitors, should not be able to leverage their ownership or control of transmission networks to dominate, or extract a surcharge from, the complementary markets for goods and services that use those transmission networks to reach business and residential consumers.

Further, the reality is that VoIP subscribers have already paid SBC for the use of its transmission networks through the purchase of DSL service, and the common carrier that delivers the VoIP traffic to SBC has likewise paid SBC for termination of the VoIP traffic on its network under the FCC's current access charge rules, so no one is using SBC's "pipes" for free."

– Earl Comstock (page 74)

WHAT'S ON TMCNET.COM RIGHT NOW

To stay current and to keep up-to-date with all that's happening in the fast-paced world of IP telephony, just point your browser to http://www.tmcnet.com for all the latest news and analysis. With more than 5.9 million unique page views per month, translating into more than 700,000 visitors, TMCnet.com is where you need to be if you want to know what's happening in VoIP.

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

Vonage: Any U.S. Customer Dialing 911 Will Get Help

Vonage announced that all of its customers now have access to 911 services. Today, any Vonage customer in the U.S. who dials 911, will get help when they need it most. http://tmcnet.com/218.1

theglobe.com Hires Bankers to Exit VoIP Business

Only days after the company was the subject of a highly speculative article in the influential financial weekly, Barron's, theglobe.com confirmed it has engaged Kaufman Brothers as investment advisors to evaluate partnership or sale opportunities for its VoIP subsidiary. http://tmcnet.com/219.1

A Day in the Life of the Enterprise

Travel is expensive, and it's not getting any cheaper. Companies of all sizes from SMBs to large enterprises face a similar challenge these days: How to communicate and collaborate with their own offices, customers and partners throughout the globe in the most economical and effective way? Enhanced multimedia communications services that utilize IP networks have already emerged as a viable option to meet this collaboration challenge. http://tmcnet.com/220.1

Bridging The Distance Between Analytics And Operations

Many companies have found improvements to the call center process with analytics. Analytics allow companies to make decisions, sometimes in real-time, to change call center procedures to better accommodate customers. One problem many companies have had with analytics, however, is the disconnect between the analytics software and the real-time data it needs to make intelligent judgments. http://tmcnet.com/221.1

IP Centrex-Hosted PBX Market About to Explode

The IP Centrex/Hosted PBX market is poised for "explosive growth" over the next four years, with revenues reaching approximately $1.3 billion by 2009, according to the crystal ball over at In-Stat.
http://tmcnet.com/222.1

TMC's IP PBX Channel

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

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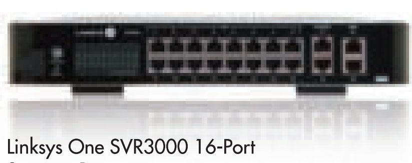

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Cisco Systems

By Rich Tehrani

Top VoIP Investments/Trends

Last year in response to many requests I started a Top 10 investments list in which I outlined some of the best investments for 2005. Here is an update on last year's list as well as some new additions. As a bonus I have added some top trends.

Peer to Peer

I led with P2P last year and lo and behold, two out of the three companies mentioned, Nimcat Networks and Skype were purchased for CA$46 million and over $2 billion respectively. P2P is still a super-hot market. Expect much more activity in this segment in the next 12 months.

VoIP Peering

There are rumors that some of the interconnection facilities may be acquired soon. These are the buildings that actually allow the physical peering to take place. Telx is a leading voice peering facility. The Voice Peering Fabric (VPF) is a Layer 2 Ethernet exchange allowing VoIP (define - news - alert) peering to take place as well as the resale of ASP services. The VPF is owned and run by Stealth Communications. As VoIP peering is generally a small part of what Telx does, the more pure play companies in VoIP peering are session border control companies as well as those that do transcoding such as Ditech Communications whose stock has appreciated 50 percent in the last quarter.

Open Source Telephony

This continues to be a rapidly growing space but it is unclear where the investment opportunities are. Companies like Digium or Pingtel are obvious plays but Sangoma, a company making high-end open source compatible hardware, is less obvious. Still, all of these companies are private at the moment which presents a problem for the casual investor.

Government Suppliers

This is still a hot area and the government will likely be one of the largest purchasers of VoIP equipment in the world. There are a few companies here worth watching such as NET.com and TeleCommunications Systems. The former used to also sell products to service providers but recently changed focus to selling enterprise VoIP solutions. In addition they have had a recent management overhaul. As a result the stock is battered and at a three-year low.

The FCC will continue to make life difficult for VoIP providers.

Triple-Play

This is still a very hot space and like last year it is the holy grail of service providers. Sales in this space are still brisk and will only get hotter. Cisco (quote - news - alert) recently acquired set-top box maker Scientific Atlanta to further entrench themselves in video and paid just under $7 billion to do so. The IPTV market is also seeing massive growth. I don't expect either of these spaces to slow for the foreseeable future.

Consumer Electronics

As VoIP gets more entrenched in our everyday lives the consumer electronics market will have more and more VoIP embedded in every day products. I am still waiting for the Apple VoIPod.

The companies that may benefit here are service providers such as Vonage or Skype that can get great licensing deals in place with these companies.

VoIP Chips and HMP

These technologies are actually at odds with one another but there are certain devices where HMP makes no sense and in these areas highly specialized VoIP processors are way to go and represent a great opportunity.

In larger VoIP systems, look to Intel and Aculab as some of the major players in the HMP market. No, unfortunately Aculab is not public.

SIP

SIP is hotter today than it was a year ago and TMC is launching *SIP Magazine* (www.sipmag.com) in print this month in order to help educate the market on the tremendous opportunities available in the space. Anything relating to SIP has legs at the moment and service providers and enterprise customers want more SIP products now.

As more SIP products are released, the desire for products and services based on this standard will only grow. Expect companies to make a nice living this year providing SIP trunks for IP PBXs.

That wraps it up for last year's list. Here are some new thoughts:

VoIP Service Providers

In the past few weeks 8x8/Packet8 worked out a deal to provide VoIP service to BellSouth and their stock price doubled overnight. Then Vonage announced it received $250 million in convertible debt! Whether this makes Vonage a good investment or not is unclear but obviously there are many people who want to back this company at all costs.

I would have to say that the position that Packet8 is in is highly enviable as they are wholesaling their service quite effectively overseas and domestically. In the past six months they have turned me into a believer in their strategy. Prior to this point they were a niche player… Now they are in an amazing position. Yes Vonage is gaining tremendous mindshare and their brand is synonymous with VoIP. Packet8 on


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the other hand has a low-risk, high-reward strategy of growing their wholesale business and reaping the future rewards of someone else's marketing.

IMS

IMS is one of the most exciting technologies I have ever been involved with and it promises to help service providers roll out new services faster than ever before and furthermore allows these providers to make more money from the networks they own. Finally it allows wired and wireless networks to work seamlessly together. At the moment anything having to do with IMS is in high demand making this another investment opportunity for the foreseeable future. TMC will be launching *IMS Magazine* (www.ims-mag.com) in print next month.

WiFi Telephony, Dual Mode

These markets will be huge. There is no doubt in my mind. Companies making money here will be those selling equipment as well as service. Even Skype (news - alert) will make a killing as they continue to license their protocol for WiFi telephony phones.

There is additional opportunity in connecting WiFi telephony devices to corporate PBXs. SpectraLink and Symbol Technologies are some of the players in this market.

Hosted VoIP

This market will grow more slowly than expected due to tremendous competition. Corporate America is not as eager to go the hosted route as the hosted community hopes. This means the IP PBX vendors will still do well for a while longer. Tremendous education still needs so take place to get more corporations comfortable with hosting.

The overall picture for VoIP is very bright. There are few segments of the market that aren't growing rapidly. If you are interested in hearing more about the financial angle regarding VoIP, don't miss the opportunity to hear investment guru and CNBC anchor Ron Insana as he keynotes our upcoming Internet Telephony Conference & EXPO this January 24–27. We are truly honored to have Mr. Insana as part of the event and I for one am very much looking forward to his illuminating comments.

Enough with the investment thoughts; Here are a few predictions that I've been mulling over:

Two-Tiered Internet

The FCC will continue to make life difficult for VoIP providers. In addition they will side with the LECs on a two-tiered Internet system meaning LECs will be able to block competitive content. For this, the Federal government will come down hard on Kevin Martin and in so doing reduce his ability to get anything accomplished in the FCC.

TV Wars

Cable companies and LECs will have brutal wars with one another about what rules the ILECs must follow to provide TV in areas where cable companies are dominant. Bribes and government wooing will take place on both sides of this conflict and 90 percent of the effort will be on killing off each other and 10 percent will be on providing services consumers actually want.

The E-911 Opportunity

In our company's insider parlance, an opportunity is often a euphemism for a "problem." This makes for some interesting turns of phrase when things go wrong and my colleagues approach me with, for example, a three-alarm opportunity. And yet sometimes an opportunity is just that: a chance to do some good, and maybe make some money while you're at it. I recently spent some time with companies in Colorado, the heartland of this country's 9-1-1 technology. A great deal of the United States' 9-1-1 development comes from Colorado, a state better known for skiing than helping to complete emergency calls to the nation's response centers.

Intrado is one of the leaders in this space and their business is booming. On their map for the future is providing next-generation 9-1-1 services to PSAP and helping enterprise customers route 9-1-1 calls. One of the challenges with business VoIP is that an employee using a VoIP phone on a VPN could be identified as calling from where the corporation has its IP PBX and not the address where the employee actually is located.

Level 3 is a carrier's carrier. This provider not only helps other service providers by providing an underlying network, they also provide bulletproof 9-1-1 service so long as the number is one that Level 3 provides.

For those service providers concerned about the expense of the above two solutions or want 9-1-1 coverage for any number from any provider, there are yet other solutions worth exploring. For example a company called Telefinity Dash911 is bringing 9-1-1 service to the realm of the smallest VoIP providers as inexpensively as possible. Gregory Giagnocavo is the company's founder and he has a number of successful startups under his belt.

Gregory saw an opportunity when he realized VoIP service providers (VSP) who wish to connect to a Tier 1 E911 provider such as Level 3, TCS, or Intrado would have lots of hard work and a great deal of expense — both upfront and ongoing — to deal with in order to get implemented. They need engineers and telecom experts to interconnect properly. In addition they will have to pay recurring monthly expenses.

Dash911 allows interconnectivity to the Intrado network at a fraction of the cost. Starting at $395/month you pay according to your needs at $1.45 per telephone number per month. This includes 24x7 access to Dash911's information and update call center and a fair measure of handholding for starters. This call center accepts change of address phone calls for a VSP's subscribers, which meets the FCC mandates requiring a VSP to provide a way for a subscriber to update his address by telephone, over the Web. Gregory says Dash911 decided to go with Intrado as a 9-1-1 calling services provider because he believes they are marketplace leaders and provide the maximum flexibility in services and coverage.

Talking to Gregory reminds me of what I love about this business. I get to see entrepreneurs come up with novel ideas that just make so much sense you just know they are onto something. Whether Dash911 will become the next Wall Street darling is unclear at the moment but certainly the ideas and services the company is rolling out seem to be perfect for the time being.

Getting back to how the system works. Dash911 can set up a Web page for you with your logo, corporate colors and URL to allow your customers to register and/or change their address, also referred to as Registered Location for 9-1-1 service. The branded Web page can be set up within a few days, requiring very little effort on your part. If you prefer, the company also provides a SOAP API so you can seamlessly integrate their system with your own Web site and applications.

Dash911 will soon be offering directory assistance, a tried and true, albeit boring service that many customers want and will pay for. And Q2 will see the implementation of v911 TTY/TDD service for the hearing impaired — an often overlooked but absolutely critical segment of the market.

Another unique application is the new opt-in SmartLink911 service that is coming soon. Using this solution, whenever one

In the meantime Yahoo!, Apple, and others will become TV stations providing broadband television to consumers at such a rapid clip that everyone will be forced to rethink their Internet TV strategy.

Finally a VoIP Phone Our Kids WANT To Use

I have been asking the VoIP community to come up with the iPod of VoIP phones for some time now. No one has listened. I asked for SMS and other next-gen features to be built in. No one listened. Now Hasbro (yes the people that make the Nerf football) has a device that is part walkie-talkie and part IM device. The "toy" is aptly named ChatNow and is selling in record numbers. On eBay it is currently selling at more than twice the list price of $80. I now predict someone will listen and develop a VoIP phone with SMS/IM built-in. Hopefully it will look good too.

Designer Phones Will Take Over

In true all or nothing fashion we will not only get really cool wireless VoIP phones but we will see more designer phones from the likes of companies like Bang & Olufsen and others. The telephone will go the way of the iPod and have to look stylish to be considered by ever-snobbier (more demanding) consumers.

More FCC Mess

of your customers dials 9-1-1 the call is actually recorded by Dash911. As part of this service your customer can program up to five emergency contacts in advance so that when 9-1-1 is dialed, these five contacts are immediately notified via e-mail, SMS or phone.

The people that are notified have the option of listening to the 9-1-1 call. Imagine how crucial such a service can be to someone with elderly parents. If your parent calls 9-1-1 and goes to the hospital today you may not find out for 24 hours or more that this call took place. If a parent is out, they might want to know why the babysitter called 9-1-1 and what was said. There are many scenarios in which this service can be useful.

Another useful feature is the "saved address" database. The Dash911 service allows your subscribers to save up to seven pre-validated addresses for 9-1-1 service. Why would you want to do this? If you travel frequently, your 'frequently used addresses' can be pre-saved as pre-verified addresses, allowing you to quickly and easily update your address to your new location.

Why does it take so long to update an address into the national 9-1-1 database? Credit the 9-1-1 databases that are finicky and tough to deal with, and PSAPs with their own rules and standards. When entered the way you understand and normally refer to it, your address, called the civic or postal address, may not be exactly in the format required by the PSAP's Master Street Address Guide, or MSAG, database. This guide contains street name, direction information such as north or south and suffix details such as avenue or road (there are 32 different suffixes!)

Another service Dash911provides is checking addresses as they are entered. If there isn't an exact match the system suggests a few options and the user decides which address is correct. Addresses entered that pass this system are expected to

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The FCC will force all phone providers to pay into the Universal Service Fund and as a result Skype, Google, and others will see a massive upswing in their subscriptions as people sprint as fast as possible away from telephone numbers.

FCC Enforcement

Starting in Q1 expect enforcement of 911 rules. The FCC will get very tough with service providers that aren't providing adequate 911 service. For the record, I am 100 percent behind them on this.

Blackberry Loses Large Amounts Of Device Share

A slew of new devices from companies like Motorola will eat into Blackberry share of market and devastate the company. These devices will be VoIP and presence enabled, will allow seamless browsing, and look sleek and new-age.

More importantly, the mobile device market will really take off as processors get faster, wireless networks speed up and applications take advantage of small screen size. It will finally be almost redundant to have a laptop unless you need access to large graphics.

Web 2.0 and VoIP 2.0 Finally Merge

The opening up of the GoogleTalk API creates a slew of new products and services leveraging Google's open API to allow developers to voice enable everything on the Internet. Every Web site now can easily take advantage of the amazing potential of adding voice.

Google Launches GoogleTalk "Out"

On the heels of the "GoogleTalk Out" announcement, eBay/Skype will sue Google to get them to change the name to something else. Google changes the name but gets so annoyed they vow internally to take eBay out. Google subsequently begins to integrate tighter with Amazon and eBay makes nice with Yahoo!

Regardless of what the name becomes, GoogleTalk Out allows any Web application to be connected with any and every phone. Overnight, new business models appear on specialized Web sites allowing these sites to send phone calls to retail establishments on a paid per call basis.

Google (quote - news - alert) will get involved in this market allowing the pay per call ads to be placed on a variety of Web sites like Google ads are distributed today. Google will get a cut of every call as will the owner of all the specialty sites.

VoIP Continues Growth

Lastly, I predict that VoIP, IMS, and SIP will continue to grow rapidly and will blaze past many of the most optimistic market forecasts. IT

TMC's Internet Telephony Magazine is working more closely with the industry's leading VoIP association — the IPCC — to become a sounding board for service providers and regulatory bodies in order to assist the service provider and carrier migration to safe and secure VoIP in a technology neutral and unbiased fashion. For more information please visit http://ipccforum.org.

reduce error rates that VSPs must correct to less than five percent using this service, according to the company.

Now it probably makes sense to keep seven addresses that are "MSAG approved" on file. As you can imagine the company also keeps a record of voice and Web activity of all changes in case they are needed.

Our conversation then went into how the 9-1-1 system in our country needs a drastic overhaul operationally and regulatory. One obvious area needing improvement is that operators at some public safety answering points, otherwise known as PSAPs or 9-1-1 call centers, can receive extra pertinent information via a 80-character field while other PSAPs can only receive a 30-character field. However, due to variations in what PSAPs can receive and use, this field is rarely populated with extra and critical life-saving information.

The extra information that could be made available along with an emergency call could reduce confusion in an emergency, decrease response times and easily save lives. For example what if someone who is mute calls 9-1-1, how do they respond to requests from emergency workers if a door is locked? What if a wheelchair-bound person is upstairs when a fire breaks out — how would anyone know? What about a guard dog? Will it attack? Do emergency workers shoot the dog? Or, is it a harmless canine whose bark is louder than its bite? For these and many more situations, it would be much more helpful and possibly critically important to emergency responders if they had this information when the 9-1-1 call comes in. Our national 9-1-1 system needs major enhancements and uniformity; there are too many disparate systems being used across the USA.

Other concepts and ideas we discussed were how there doesn't seem to be a single way to immediately communicate with all the more than 7,000 PSAPs in the country. As previously mentioned the 9-1-1 and PSAP system is highly fragmented and standards, equipment, procedures, policies and methodologies vary widely. For something so critically important, surely we as a nation can come up with a better and more efficient system.

Tom Ridge to Speak at ITEXPO

I am looking forward to hearing Tom Ridge's comments at the upcoming Internet Telephony Conference & EXPO this month in Ft. Lauderdale Florida (January 24–27). Secretary Ridge will present his thoughts on a number of issues ranging from security to how we can effectively improve the state of 9-1-1 service in this country. The event will also feature an E-911/Regulation Summit, which is a must-attend for anyone interested in learning more about the pertinent regulatory issues facing VoIP companies today.

Some of the session titles that will be featured in Ft. Lauderdale include:

- VoIP's Role in Military & First-Responder Communications;
- VoIP Regulatory Update; and
- E-911 Technical Session

These sessions will serve as an update on VoIP regulatory and compliance issues. Attendees can get up to speed on the latest situation regarding the support of CALEA, E-911, and taxation issues. Recent FCC rulings will be explored and the ramifications for the industry will be addressed. Some topics covered will include where carriers stand in their ability to support 9-1-1, wiretapping, and the touchy issue of taxing the regulated providers to pay to support these initiatives.

Topics will also range to discuss the technological advantages of using a VoIP-based solution in times of crisis such as the recent hurricane strikes that damaged traditional telecommunications networks. If you need to know how today's solutions work, and how they might work in your network, you need to attend this Summit. If you are at all affected by our emergency response system — and I imagine that's pretty much everyone who reads this column — you will likely want to be at the show to hear what Mr. Ridge has to say. I look forward to seeing you there. IT

Go To Table of Contents | Go To Ad Index

Contents

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Intel Continues to Push HMP Envelope

By Greg Galitzine
Internet Telephony Magazine

Intel host media processing (HMP) solutions are helping customers of all sizes achieve results. Across the industry, different groups, including service providers, ISVs and enterprises alike are all benefiting from the latest HMP releases from Intel. Service providers are increasing their average revenue per user (ARPU) by offering exciting new enhanced services. ISVs are generating more for less, by increasingly offering higher-density applications while shortening their time to market with those apps. And enterprise customers are benefiting by deploying converged IP applications such as IP PBXs, and saving money in the process.

For those new to the term, host media processing is a technology used to perform media processing tasks on general-purpose servers with standard processors but without specialized digital signal processing (DSP) hardware. Host media processing technology provides media services that can be used to build flexible, scalable, and cost-effective next generation media servers.

Because HMP is a software-only media processing solution, costs are lower for procurement, development, deployment, and maintenance. Since HMP is standards-based and does not require specialized hardware, solutions using HMP are faster and easier to develop and deploy, easily scalable, and very flexible.

Intel NetStructure® Host Media Processing Software combines the flexibility of software-based IP media processing with powerful high-density multimedia capabilities that run on a server's host processor. In today's highly competitive marketplace, offering multimedia services can be a key differentiator for telephony solutions.

Release 1.5 for Linux

The company recently announced new hardware and software products for simplified deployment of IP services such as VoIP and multimedia services in pure IP and hybrid enterprise networks. The announcement centered on Intel NetStructure Host Media Processing Software Release 1.5 for Linux, as well as a reference design for a Converged Application Platform for distributed enterprises.

The Intel NetStructure Software enables up to 120 channels of video as well as revenue-generating services like video mail, video caller ID, video portals, and video color ringback.

When deployed in an IP network, Intel NetStructure HMP supports the initiation and termination of a multimedia (audio/video) call, which includes SIP-based call control and H.263 video format. Intel NetStructure HMP provides synchronization between voice and video streams for playback on IP video phones and video-enabled soft clients and connection to a 3G-324M gateway on a 3G network.

Service providers the world over are thrilled.

According to Suntek CTO Zhang Shaowen, "The release of Intel NetStructure Host Media Processing Software 1.5 will enable Intel customers to build video portals, voice and video messaging, and other value added services. Suntek deployed network KTV and video portal solutions over a NGN network in Chengdu China and achieved overwhelming customer satisfaction."

Tristan Dessain-Gelinet, preisdent of Asniere, France-based Tetco Technologies agrees. "TETCO Technologies has chosen Intel NetStructure Host Media Processing for its flexibility and easy scalability. Our Next Generation platform based on HMP from Intel has been able to provide hundreds of media channels for value added services such as Next Gen Voicemail, Video, Ring Back Tones, with lower CPU consumption and a limited footprint."

Since the solution is implemented as a software-only product, there are numerous cost benefits. There is a smaller up-front investment and lower development, deployment, and ongoing operational costs. Since the Intel solution is compatible with both the ITU's H.323 standard and the IETF's Session Initiation Protocol (SIP), developers gain access to a wide array of gateways, gatekeepers, and IP endpoints through standards-fueled interoperability. Speaking of interoperability, the fact that Intel HMP Release 1.5 for Linux supports streaming media over RTP (real-time transfer protocol) using a variety of codecs (G.711, G.723.1, G.729a and G.729ab…) means that developers can create products that will work with a wide variety of industry standard IP gateways.

Release 2.0 for Windows

Intel NetStructure Host Media Processing Software is also being used to help enterprises move to VoIP technology while protecting their investments in legacy equipment. The company recently announced the Intel NetStructure Host Media Processing Software Release 2.0 for Windows and the Intel NetStructure Digital Network and Station Interface Boards to aid enterprises in deploying IP-based services in a hybrid network, combining legacy TDM with IP infrastructures.

Intel is also providing help for small to medium-sized enterprises (SME) interested in deploying multimedia services by releasing a reference design for a Converged Application Platform for the Distributed Enterprise.

The reference design for a Converged Application Platform for distributed enterprises will help SMEs in deploying multimedia services. The design uses processors and host media processing from Intel, enabling integration of separate pieces of equipment (like PBXs, switches, routers and VPN/firewalls) into one device. Service providers may offer the platform as a hosted service, allowing IT managers to easily deploy, maintain and upgrade their networks for support of a variety of multimedia applications like fax, instant messaging and multimedia conferencing.

According to Rusty Cone, President, Alliance Systems, "Alliance Systems is using the Converged Access Platform reference design from Intel to offer an integrated data and telephony network. Our customers will quickly be able to benefit from the combination of Intel processors and Intel NetStructure Host Media Processing, including scalability and high performance. With this approach, enterprise networks can be managed in a cost effective and simple way."

The Intel NetStructure Host Media Processing Software Release 2.0 for Windows supports up to 400 channels of rich media processing on various Intel platforms, including those under the Pentium (III, 4, and M), Celeron, and Xeon banners. The latest software release facilitates the development of many advanced applications such as conferencing and voice applications such as IVR and voice mail through support for such functionality as voice record/play with tone detection and generation and conferencing features such as coaching, active talker notification, tone clamping, echo cancellation, and scalability of up to 400 parties per system.

Intel HMP solutions - both the Release 1.5 for Linux and the Release 2.0 for Windows - enable a number of applications including the following:

- IP media gateways
- IVR and announcements
- IP PBX
- Video messaging server
- Video server
- Voice mail server
- Unified messaging server
- Converged PBX
- Prepaid/debit card services
- Enhanced services
- Conferencing server
- Speech enabled applications

Greg Galitzine is editorial director of INTERNET TELEPHONY® magazine.

Multimedia Services - Faster, Easier and More Affordable

Service providers want to offer multimedia services to their telecommunications customers to increase revenue, improve the end user experience, and compete more effectively. At the same time, enterprises are investigating multimedia and VoIP services to increase productivity and reduce operating costs while protecting infrastructure investment. But adding new services has often been costly and complex - until now. Intel NetStructure Host Media Processing Software is allowing developers to provide new IP-based services and solutions faster, more easily, and at a lower cost.

Enterprises that want to offer IP services but continue to use legacy equipment should consider using Release 2.0 of HMP software with Intel NetStructure Digital Interface and Station Interface Boards. These products let enterprises deploy IP-based services in their current environment, seamlessly integrating legacy TDM and newer IP infrastructures.

Service providers that want to enable new revenue-generating multimedia services (video mail, video color ring back, video caller ID, video portals and more) should investigate Release 1.5 of HMP software. The software supports multimedia in easily scalable high-density configurations.

Developers Offering Enhanced Solutions with Intel Products

Developers and their customers have eagerly adopted HMP technology for their solutions. Here are some recent announcements:

Edify

Edify, a leading global supplier of voice and speech solutions, recently announced its Edify Voice Interaction Platform, Release 9.5. This product provides the flexibility to implement the IP solution in a wide variety of environments IT investment and reduce operation and maintenance costs.

Paraxip Technologies

Paraxip recently announced a version of its gateway software that runs on a VoIP Media Gateway Reference Platform from Intel. The reference platform allows SIP application vendors and system integrators to free themselves from dependency on closed and proprietary gateway appliances.

Pronexus Inc.

Pronexus, an IVR and speech solutions innovator, has been selected by CenturiSoft, a developer of unified communications systems, to provide rapid application development tools for building IVR applications. Centuri Media Switch is their latest product and is built with VBVoice by Pronexus to enable voice and fax communications over the Internet and Intranets.

More Praise for HMP Software

"HMP software from Intel provides All New Video with an application development platform which meets our stringent requirements for a highly scalable, carrier-grade video-enabled IP media server," said Lee Woodland, CTO of All New Video. "This is a key component of our next generation of applications which include video content delivery and video messaging."

"Baytalkitec has selected Intel NetStructure Host Media Processing Software 1.5 for its IP solutions based on the rich multimedia services HMP enables and the performance when combined with Intel processors. Video telephony is the future for carriers and operators to increase revenues and HMP makes it possible for a rapid development and deployment of these services."

"We're committed to allowing our customers to quickly utilize the latest VoIP capabilities of Host Media Processing from Intel with our rapid development solutions, the Envox 6 Communications Development Platform and Envox CT ADE." said Mark D. Flanagan, president and CEO of Envox Worldwide. "Through this powerful software, which provides connections to both legacy hardware and new IP devices, our customers will be able to reduce the time, cost, and complexity of creating voice solutions and migrating those solutions to an IP or hybrid infrastructure."

To learn more about HMP and other IP products from Intel, visit www.intel.com/go/iptoday.

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Utilize the Intel Communications Alliance solutions directory and its extensive network of over 300 member companies to quickly locate products and solutions that can help you speed up development cycles, cut costs and solve design issues.

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TMCnet Traffic Analysis

Note: Alexa.com ranks Web sites to their proximity to being #1. The lower the number, the higher the ranking and therefore the greater the traffic. Yahoo!, the world's busiest Web site, is ranked #1 by Alexa.com

TMCnet.com Traffic vs. Technology/IT Web Sites

Web Site	Alexa Site Rank
TMCnet.com	2,381
eWeek.com	2,826
Computerworld	4,671
InfoWorld	6,618
Network World	8,394
Light Reading	14,655
Pulver.com	36,063
Wireless Week	40,701
Destination CRM	48,598
Telephony Online	58,251
VoIP News	76,801
Telephony World	121,573
Call Center Magazine	183,448
America's Network	185,033
Telecomweb	204,159
CommWeb	249,258
Wireless Review	317,334
Communications News	984,904

TMCnet.com Traffic vs. Business Magazine Web Sites

Web Site	Alexa Site Rank
TMCnet.com	2,381
Fortune Magazine	2,484
Smart Money	2,980
Inc. Magazine	4,984
Fast Company	5,259
Business 2.0	5,986
Barron's Online	6,560
Weekly Standard	8,996
Technology Review	9,624
CIO Magazine	11,330
BtoB Online	23,419
Worth Magazine Online	174,723

TMCnet.com Traffic vs. Prominent Web Sites

Web Site	Alexa Site Rank
TMCnet.com	2,381
Sharper Image	4,152
Volkswagen	4,258
Nokia USA	4,351
Coca-Cola	7,670
Brookstone	10,045
GE Appliances	11,058
Brooks Brothers	14,899
JVC	15,692
Black & Decker	32,061

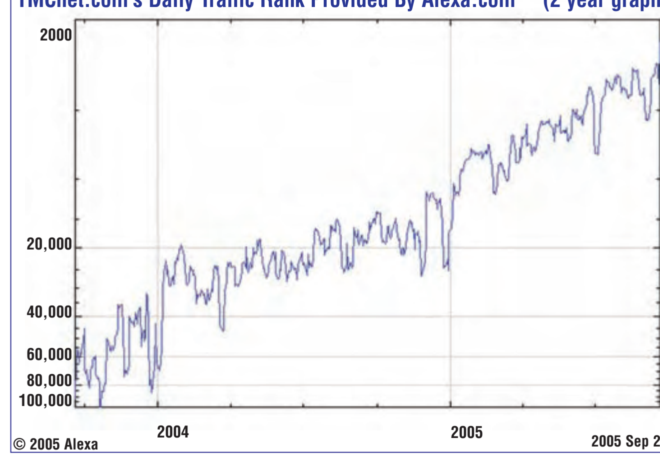
Source: Alexa.com 12/5/05

• Over 7.4 Million Page Views*
• Over 629,000 Unique Vistiors'*

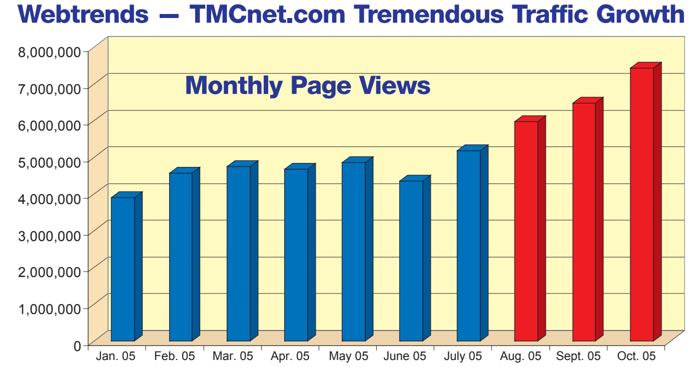
*Source: Web Trends October, 2005

No Other Communications Site Even Comes Close!

TMCnet.com's Daily Traffic Rank Provided By Alexa.com** (2 year graph)



Webtrends — TMCnet.com Tremendous Traffic Growth



**Source: Alexa.com ranks Web sites by traffic. The number indicates a site's proximity to being the number one most visited Web site. Date: 12/5/05 Alexa is an Amazon.com Company. Neither Alexa.com nor Amazon.com endorse, or are affiliated with, TMCnet.com in any way.

The Changing Face of TV

News Analysis By Robert Liu

Qualcomm (quote - news - alert) efforts to launch its own multicast network took a major step forward with the recent decision by Verizon Wireless to deploy its proprietary MediaFLO platform. The announcement underscores the seismic changes that could grip hold of the TV world over the course of the next 12 months.

FLO (which stands for Forward Link Only) is a multicast technology developed by the Code Division Multiple Access (CDMA) innovator and delivered over the 700 MHz spectrum that Qualcomm currently owns. As part of the agreement, Verizon hopes to offer broadcast TV via MediaFLO as a complement to its VCast and BroadbandAccess on-demand video services in half of its EV-DO footprint by the fourth quarter of 2006.

While it's unclear if Verizon's existing VCast agreements governing on-demand content also applies to live broadcasts, Qualcomm is taking no chances. As a wholly owned subsidiary, MediaFLO has been hard at work negotiating its own relationships with media giants over the use of their content.

"It's certainly critically important to the consumer what content is on the handset," said Jeff Lorbeck, senior vice president at Qualcomm and general manager for MediaFLO USA. "We've been talking to the content providers for a number of months. We've been talking to virtually all major media brands."

But content rights are just one of several factors that could affect the rollout of mobile TV. And that's exactly why analysts like Linda Barrabee at Yankee Group predict we could be well into 2007 or beyond by the time cell phones start replacing TV sets.

"At the end of the day, it's about handset replacements. You've got to go through the typical handset replacement cycles," the senior analyst explained. Of the 193.6 million cell phone users, less than one-half of one percent takes advantage of video multimedia content.

Barrabee believes handset makers and operators aren't likely to throw their weight behind mobile TV until kinks like technological standards and network performance can be shaken out of the first-generation offerings. Qualcomm's MediaFLO is positioned against a competing open standard called DVB-H (Digital Video Broadcasting for Handhelds), which Crown Castle International has opted to deploy in the 1.67 GHz band. Nokia, Motorola and Samsung (using Microsoft software) have already agreed to conduct trials using DVB-H.

Jeff Lorbeck is senior vice president at Qualcomm and general manager for MediaFLO USA.

During a recent phone interview with *INTERNET TELEPHONY*, Lorbeck argued that MediaFLO operates more efficiently because Qualcomm is permitted to transmit its signal at a higher power than Crown Castle. And even if power levels remained at a constant, lower frequency transmissions generally disperse to a wider distance than higher frequency transmissions. Those factors, in turn, translate to lower capital expenditures because fewer transmitters are needed for any given market using the MediaFLO platform.

"Although it is too early to predict the outcome of this new battle, it is clear that mobile-TV has gained traction and is the focal point of many heavy hitters in the industry. The battlefield in the U.S. is clearly tilting toward Qualcomm's favor given the simple laws of physics and costs," Deutsche Bank analysts recently concluded in a report on Fixed Mobile Convergence.

Complicating matters for Qualcomm, though, is the fact that it has been unable to fully gain access to the 700 MHz spectrum. A number of independent TV stations still operate at or around that frequency, Channel 55. (Remember UHF?) Qualcomm has been negotiating with station operators in various markets to take control of those airwaves.

In addition, DVB-H and FLO (as its name implies) are both examples of

Go To Table of Contents | Go To Ad Index

"Although it is too early to predict the outcome of this new battle, it is clear that mobile-TV has gained traction and is the focal point of many heavy hitters in the industry.

multicasting technologies and only permit data to flow uni-directionally. Carriers like Sprint are deploying mobile TV based on unicast technologies, which would allow the bi-directional flow of data permitting services like remote DVR programming.

"DVB-H or other broadcast (as opposed to unicast) technologies are useful for providing a mass audience with live TV in order to prevent potential network bottlenecks, which may occur in future on unicast networks. They do not, however, facilitate interactivity or a return-path functionality which would allow commerce, voting or other two-way communication with a broadcaster," said Jason Taylor, Director of Corporate Communications, MobiTV. IT

Robert Liu is the Executive Editor of TMCnet, the news and information portal of Technology Marketing Corporation, and is a frequent contributor to INTERNET TELEPHONY magazine.

Robert's 15-year communications career spans from the print world to television and to the Internet. He has covered business and technology writing for Dow Jones, Bloomberg Business News, CNN, and Jupitermedia's internetnews.com. He has served as a producer at CNN, Headline News and A&E Television Networks. You may contact Robert at rliu@tmcnet.com.

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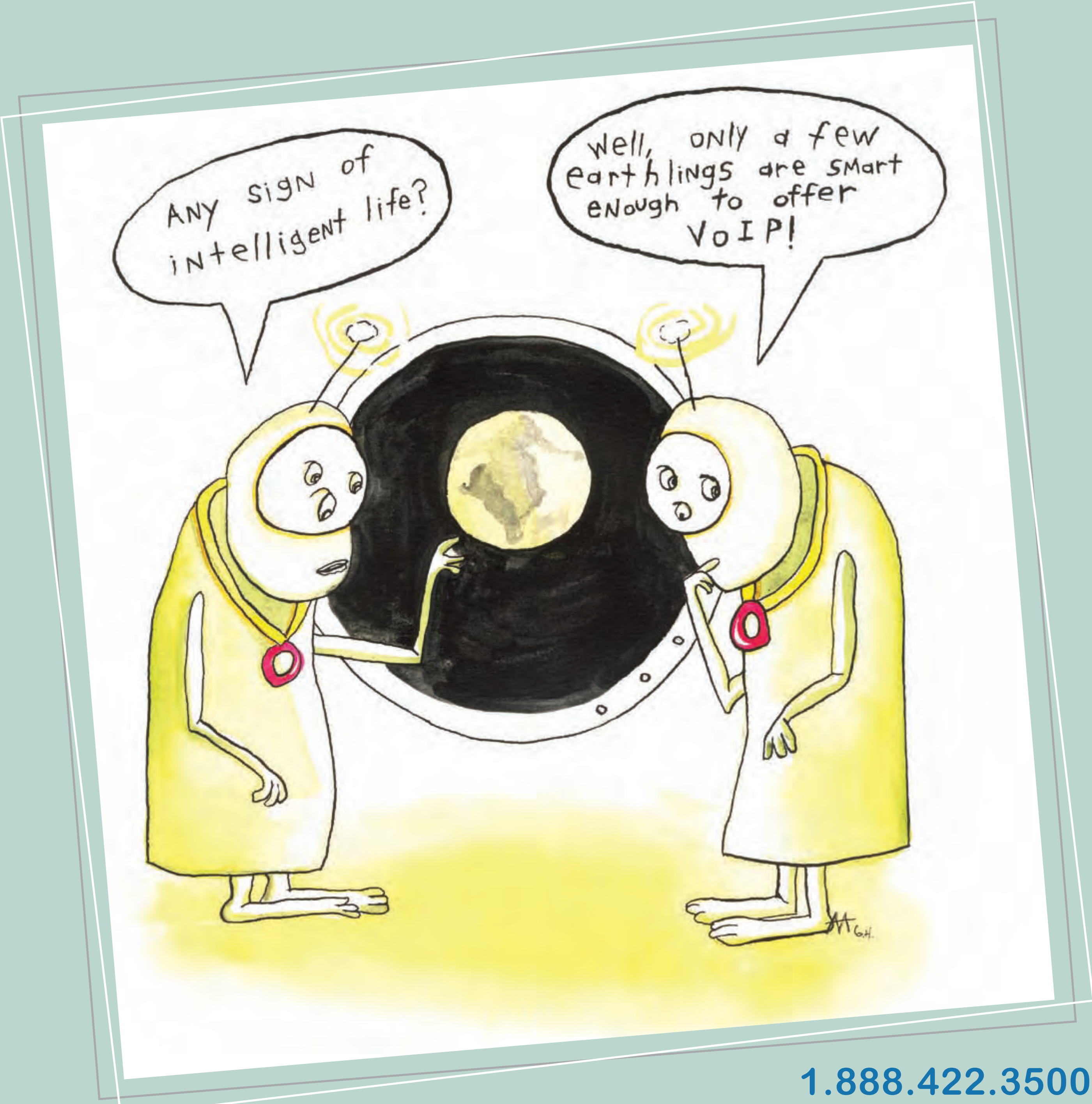

Any sign of intelligent life?
Well, only a few earthlings are smart enough to offer VoIP!

Shift Networks Expands Canadian VoIP Network with Common Voices

Business Subscriber Growth Drives Expansion of Next Generation Messaging Platform

Common Voices today announced that it has received a significant order for expansion of its NowMessage™ service from Calgary-based Shift Networks. (news - alert) Shift is using NowMessage to provide hosted Unified Messaging services to its small business customers in Canada.

"Shift has more than quadrupled its capacity since launching this service in 2004. This is a very rapid expansion and confirms the suitability of our NowMessage Unified Messaging services in the small- to medium-sized business environment," commented Todd Hasselbeck, CEO of Common Voices (news - alert). "We are very pleased to be working with a customer that is aggressively rolling out leading edge VoIP applications."

NowMessage is a call answering and unified messaging application that solves the issues with proprietary voicemail systems for wireline and wireless carriers. It provides leading-edge enhanced services business users require on a standards-based, VoiceXML software architecture, using off-the-shelf hardware components.

Shift is also using Common Voices OpenLink™ VoiceXML browser and will add licenses for its OpenLink offering, too.

Rick Unrau, Chief Operating Officer at Shift, said, "With NowMessage and OpenLink we have been able to continue to offer highly reliable hosted services to our customers and to expand our offerings quickly and efficiently. Using Common Voices' software-based solution, paired with our industry standard servers, we can respond to our customers' needs as our market grows and that's an important part of our offering." Shift Networks is the first company in Canada to provide "business class" local telephone lines for small and medium business via broadband connection. Its hosted multi-line, business class Voice over IP solution provides businesses with leading edge telephone system and services that are centrally hosted at Shift's carrier-class data center.

http://www.shiftnetworks.com
http://www.commonvoices.com

NetClarity Upgrades Line of Vulnerability Management Appliances; Auditor Now Provides Multi-Appliance Correlation and VoIP Security Testing

NetClarity (news - alert) announced significant enhancements to its Auditor line of vulnerability and intrusion management appliances, including multi-appliance correlation and scans of VoIP equipment. These new features build on NetClarity's patented clientless Vulnerability Quarantine System (VQS) and smart switch integration introduced in June 2005 to further enable customers to proactively strengthen their networks from the core to the edge. Auditor™ offers an industry-first zero footprint solution that helps protect against trusted high risk insiders and potentially malicious insiders including those with dirty laptops and rogue wireless devices.

Auditor's Command Center now enables customers to manage all of their Auditor appliances, including those at remote sites, from a central console. This streamlines the vulnerability audit, workflow and remediation processes, saving customers a significant amount of time and money. In addition, Auditor now has the capability of scanning VoIP network equipment for Common Vulnerabilities and Exposures (CVE), the systemic cause of over 95 percent of all network security breaches.

"If you are considering deploying VoIP on the same network as your desktop computers and servers, you are at high risk of poor call quality, denial of service, breaches of privacy, integrity and availability. By removing your CVEs, you can quickly mitigate much of this risk," said Gary Miliefsky, chief technology officer for NetClarity. "Because these packet-based networks are not very secure by default they are extremely susceptible to attacks such as Man in the Middle (eavesdropping and alerting) and Denial of Service (DoS). Auditor now enables customers to quickly find and remediate CVEs that may lead to these types of attacks."

http://www.netclarity.net

Citrix Improves Its Position in the Visionary Quadrant of Leading Industry Analyst Firm's SSL VPN Magic Quadrant

Citrix Systems, Inc., (news - alert) the global leader in access infrastructure solutions, today announced it has been positioned by Gartner, Inc. (news - alert) in the "visionary" quadrant in the "Magic Quadrant for SSL VPN, North America, 3Q05"(1) report. Gartner vice president and distinguished analyst John Girard evaluated 14 SSL VPN vendors based on their ability to execute and the completeness of their vision. Gartner is a research and advisory firm that helps more than 45,000 clients worldwide understand technology and drive business growth.

According to 2005 criteria, Gartner focused on seven areas within the parameters of vision and execution — market understanding, product strategy, innovation, overall viability, marketing execution, sales execution/pricing and customer experience. According to Gartner, vendors positioned in the "Visionaries" quadrant demonstrate a clear understanding of where the market is headed and how they intend to capitalize on market trends. Visionaries invest in the leading/bleeding edge features that will be significant in the next generation of products and give buyers early access to improved security and management. Visionaries can affect the course of technological developments in the market, but they lack the execution influence to outmaneuver challengers and leaders. Clients pick visionaries for best-of-breed features, and in the case of small vendors, they may enjoy more personal attention.

The report states, "SSL is both a new VPN remote access and a replacement market for legacy IPsec remote access. SSL is the most widely deployed virtual privacy system in the industry because it is integral to every browser and independent of platforms and operating systems. It is the ultimate VPN in terms of portability and is also the best known VPN method. Gartner estimates that by 2008, SSL VPNs will be the primary remote access method for more than two-thirds of business teleworking employees, more than three-quarters of contractors and for more than 90 percent of casual employee access."

"We believe being positioned in the SSL VPN visionary quadrant is a testament to the core capabilities of Citrix and recognition of the strategic market position that Citrix is taking in this high growth market," said Gordon Payne, vice president of marketing at Citrix. "The Citrix Access Gateway offers new customers as well as our installed base of more than 160,000 Citrix customers a compelling and easy to deploy SSL VPN solution for secure remote access to any enterprise resource. We are seeing rapid customer adoption."

http://www.citrix.com
http://www.gartner.com

Sonus Voice Portal Allows VoIP Consumers to Customize Services

By Patrick Barnard, TMCnet Associate Editor

Sonus Networks Inc. (news - alert) has released its new Sonus Voice Portal, which enables VoIP consumers to customize their service through a voice activated menu.

According to a company press release, the Sonus Voice Portal is the first application created with, and managed by, Sonus' recently released IMX application platform, a web-based multimedia environment that enables wireline and wireless service providers to rapidly develop, integrate, launch and manage enhanced telecommunication applications and services. The portal manages the preferences for a variety of residential services delivered on the IMX platform, including speed dialing, call forwarding, three-way calling and nuisance call blocking.

Jupiter Telecommunications Co., Ltd. is reportedly the first Sonus customer to deploy the new Voice Portal application.

"The Sonus Voice Portal is just the first of a number of new applications that leverage the power of the Sonus IMX application platform to reshape today's communications experience," said Mike Hluchyj, chief technology officer, Sonus Networks.

Hluchyj said the portal allows subscribers to customize their service "while they are on the move."

http://www.sonusnetworks.com

SKY Announces SKY-MAP 2.0 Mobile Applications Software Platform For Multimedia Handsets

By Laura Stotler, TMCnet IP Communications Columnist

In an announcement that will take multimedia handsets to the next level for delivering enhanced services, SKY MobileMedia, Inc. (news - alert) has announced its SKY-MAP 2.0 mobile applications software platform. The new platform offers an integration of services and engines for multimedia handsets, converged consumer electronics devices and mobile chipsets.

The latest platform offering from SKY features the Khronos-compliant SKY-MMF Multimedia Framework at its center. This portable, wireless framework adds the ability to create, edit, play, stream and share multimedia with almost any wireless multimedia device. It supports a number of file formats and codecs, and contributes to and complies with ratified Khronos standards for multimedia. SKY is a member of the Khronos group and plans to continue compliance with future Khronos standards. The enhanced framework enables SKY-MAP 2.0 to deliver multimedia-enabled IMS services, streaming web content, video messaging, camcorders and the purchase and playback of music and digital television.

SKY-MAP 2.0 also features the JTWI-compliant SKY-VM Java client and an enhanced SKY-MMI user interface framework. This helps wireless operators and manufacturers in deploying new phone features and multimedia services in compliance with industry standards. Other features include the growing SKY-APPS pre-integrated mobile applications suite, which consists of the SKY-MAIL Email Client, DRM and SKY-PoC Push-to-Talk Client.

"SKY-MAP has been in volume production worldwide since early 2005, and has rapidly proven itself as a leading software platform for feature phones," said Naser Partovi, CEO of SKY MobileMedia. "Our second generation solution, SKY-MAP 2.0, extends this platform to include high-end multimedia functionality and applications — enabling SKY's handset and consumer electronics customers to rapidly develop exciting next-generation wireless multimedia handsets and converged devices."

http://www.skymobilemedia.com

Aventail Adds Web Collaboration to SSL VPN Product

By Johanne Torres, TMCnet VoIP Minute Watch Columnist

SSL VPN technology provider Aventail Corp. (news - alert) announced on Monday it added a Web collaboration system to its remote access product. The new collaboration system features Remote Helpdesk, Web Conferencing, Instant Messaging, Secure VoIP chat, and other tools, all integrated with the Aventail SSL VPN. The system supports up to 500 concurrent users and multiple platforms, including Windows, Mac, Linux, and PocketPC.

The Secure Remote Helpdesk is a Web-based remote helpdesk system available through a portal from any device, platform and network connection that is VoIP and text enabled. The Secure Web Conferencing feature is a scalable Web-based conferencing system for meetings, training sessions, or demonstrations across multiple platforms and user groups. The Secure Teamwork feature is a set of collaboration tools that allow instant and impromptu conversations, scheduled meetings, and ad hoc collaboration between users and groups.

"Most companies will see a return on investment within three to six months of deploying Secure Collaboration with our SSL VPN," said Sarah Daniels, vice president of product management and marketing, Aventail. "The cost savings are realized in several areas, for example, companies will no longer need separate collaboration products, and they will receive additional productivity and security gains for collaboration functions such as VoIP and instant messaging."

The appliance comes standard with all access methods and feature add-ons, including Aventail Secure Desktop, Aventail Connect, Aventail OnDemand, and Aventail Mobile, making it the most cost effective enterprise solution on the market. Competing products require additional licensing fees for advanced features or access methods.

http://www.aventail.com

MCI Launches Industry's First Comprehensive Security Risk Management Service

MCI, Inc. (quote - news - alert) announced the launch of its NetSec Security Risk Management Service, a new managed solution that helps companies improve their security by quantifying, prioritizing, and remediating security risks across an enterprise. MCI's latest cloud-to-core offering enables companies to enhance their decision-making capabilities to take proactive and immediate action against threats and vulnerabilities, when and where it is needed most.

"MCI is changing the security risk management landscape as we know it," said Chris Sharp, vice president of NetSec Security Services for MCI. "Until now, organizations have been continually challenged knowing which systems are at risk at any specific time. With our new Security Risk Management Service, organizations can intelligently manage risk so they can better protect their critical assets and systems."

With NetSec Security Risk Management Service, MCI customers gain enhanced visibility into and centralized control over the security of their environment. Powered by Finium, the company's next-generation integrated services and delivery platform, this new offering correlates and calculates diverse threats enterprise-wide using a dynamic scorecard-based approach to provide prioritization and remediation capabilities. As a result, companies can strengthen their defense and enhance their security investments, while reducing strain on corporate resources.

With the next-generation Finium platform, MCI customers are able to easily navigate their IT environments through a secure and intuitive Web interface. Customers can view real-time risk information and navigate through associated threat descriptions, system and device configurations, remediation plans and reports. Extensive filtering and drilldown capabilities also can be tailored to meet an organization's policy and business model for more meaningful interpretation and better risk assessment.

http://http://www.mci.com

Peneo, A New Player on the VoIP Market

Launched in October 2005, Peneo Telecommunications (news - alert) enables anyone with high-speed Internet access to sign up for voice over internet protocol (VoIP) services. Users can sign up easily and they can enjoy the PC-to-PC telephony for free. Peneo offers challenging low rates when it comes to calling out to PSTN lines worldwide.

Peneo uses the widely spread SIP technology. All SIP-compatible standalone phone devices are supported, which means users can initiate or receive VoIP phone calls without a PC.

While free calls between users is common in the VoIP industry, what makes Peneo standing out is its wide range of services. Any Peneo user can connect two telephone lines anywhere on the world using an online interface and paying only Peneo's low rates. Web callback can be used from any PC and does not require any software to be downloaded or installed. You simply login, enter the two phone numbers that you want to be connected, and click 'connect.' Peneo's server will call you back on the first number, then after a short delay, connect the second number allowing you to talk.

There is also an option called SMS callback. Simply send an SMS to Peneo including your account login and the two phone numbers you wish to connect, and that's it. Both phones are ringing, and once again, even if you're on the other side of the world, you pay Peneo's low rates. All services are particularly useful if travelling or if you want to make low cost calls from a country where pc2phone services are blocked. Once you have a Peneo account, you automatically have access to our callback services using the same username and password.

http://www.peneo.com

Gartner Report Outlines Six Disruptive Trends in IT

By Patrick Barnard, TMCnet Associate Editor

A new study by Gartner Inc. (news - alert) predicts that 30 percent of U.S. households will use only cellular or Internet telephony by the year 2010.

According to Gartner's "Top Predictions for 2006 and Beyond," "growth in traditional wired voice connections will slow in North America, Western Europe and other developed markets as more people dedicate fixed phone lines to DSL links and switch to cellular or Internet telephony."

"U.S. consumers are just beginning to add voice over Internet Protocol (VoIP) services to their range of telephony options, but as they get more comfortable with the technology, and as VoIP services improve, they will start to abandon traditional phones," a press release concerning the new study states. "Mobile communications will remain the preference of developing countries, and as a result, wireless links will represent 99 percent of the world's new voice connections in 2009."

Ken Dulaney, vice president and analyst at Gartner, said in the press release that after 125 years, plain old telephone service "is now on the decline in the U.S."

"Top Predictions for 2006 and Beyond" is part of a series of reports, called "Gartner Predicts," to be released over the next year. The nearly 50 reports included in the series discuss the major trends that will affect IT users, vendors and most industries in 2006 and beyond. "Top Predictions for 2006 and Beyond" focuses on six IT industry trends that Gartner expects will cause "significant disruption and drive opportunity for business and the IT industry."

http://www.gartner.com

China Telecom Selects Alcatel's IP Solution to Enhance Triple Play Service Offering in Anhui Province

Alcatel (news - alert) announced that Anhui Telecom, a provincial subsidiary of China Telecom, has selected Alcatel to expand and optimize its metropolitan network across the Anhui Province to deliver higher quality and more reliable IP-based voice, video and data services. This contract was won through Alcatel Shanghai Bell, Alcatel's flagship Chinese company.

Anhui Telecom will deploy the Alcatel 7750 Service Router (SR), a purpose-built IP router, which will help the service provider deliver nonstop services with the highest levels of quality in seven major cities of the province including Bengbu, Anqing, Suzhou, Fuyang, Huangshan, Huaibei, and Tongling. Once the installation is complete in early 2006, Anhui Telecom will be able to provide more robust triple play services to both residential and enterprise users, and offer Service Level Agreement (SLA)-based networking services like Layer 2 and Layer 3 virtual private networking (VPN).

"This contract marks the 23rd province in China to utilize the Alcatel IP solution," said Basil Alwan, president of Alcatel's IP activities. "Transforming the network to deliver next-generation IP services is a notable trend; and we are pleased to be selected by so many discriminating service providers in China as they continue to build out their network to address this growing appetite for triple play services."

The Alcatel 7750 SR is the industry's first service router designed and optimized for the delivery of high performance carrier-grade data, voice and IPTV services. The platform has gained significant traction since it was introduced at the end of 2003. In China, the 7750 SR serves in the national, provincial and metro production networks of all the key service providers, including China Telecom, China Netcom, China Mobile, and China Unicom. Anhui Telecom joins a list of more than 90 customers in 40 countries around the world.

http://www.alcatel.com

Wi-Fi Chipset Sales to Surpass 120 Million in 2005

By Patrick Barnard, TMCnet Associate Editor

The Wi-Fi industry hit a major milestone this week, as sales of Wi-Fi chipsets surpassed 100 million for 2005. According to a news release from the Wi-Fi Alliance, supported by data provided by In-Stat, the 100 million mark is evidence that Wi-Fi is “now a staple of computer networking and a powerful presence in millions of homes and businesses” around the world.

“This is a significant milestone for such a young technology, and the Wi-Fi Alliance has played a strong role in helping Wi-Fi gain traction,” said Gemma Tedesco, senior analyst at In-Stat, a market research company to the telecom industry. Tedesco predicted that Wi-Fi chipset sales will exceed 120 million for 2005.

“The 64 percent average yearly growth rate [in Wi-Fi chipset sales] reflects the transforming nature of Wi-Fi and the value of interoperable, standards-based technology,” the news release states.

The Wi-Fi Alliance provides testing and certification programs to ensure the interoperability of products based on the IEEE 802.11 standard. More than 2,200 products have been Wi-Fi certified since March 2000.

”Wi-Fi has truly come of age,” said Frank Hanzlik, managing director of the Wi-Fi Alliance. “Today more than 90 percent of notebook computers are Wi-Fi enabled, and as we look ahead, there are seemingly limitless opportunities, due to the insatiable consumer and enterprise demand for the technology.”

http://www.wi-fi.org

Conexant’s Dual-Band, Multimode Solution Achieves Wi-Fi Alliance WMM Power Save Certification

New Power Management Techniques Extend Battery Life for Mobile and Handheld Devices

Conexant Systems, Inc., (news - alert) a worldwide leader in semiconductor solutions for broadband communications and the digital home, today announced that its 802.11a/b/g dual-band, multimode WorldRadio™ solution has been certified by the Wi-Fi Alliance (WFA) under the group’s new Wi-Fi CERTIFIED™ for WMM™ (Wi-Fi Multimedia) Power Save certification program. WMM Power Save includes several features that extend the operating time of battery-operated mobile devices such as multimedia cellular phones, enterprise handsets, PDAs, and digital cameras. Conexant’s chips will also be used by the WFA to validate other products attempting to achieve WMM Power Save certification. The Wi-Fi Alliance’s testing and certification programs ensure the interoperability of multi-vendor wireless local area networking (WLAN) products, providing consumers with the assurance that their Wi-Fi CERTIFIED products will work together seamlessly.

“Voice, video and audio applications running on mobile and handheld devices require increasingly robust power management features to maintain high performance and prolong battery life,” said Chee Kwan, vice president and general manager of Conexant’s Wireless Networking business. “WMM Power Save techniques deliver a significant improvement in power consumption, which is a key issue for many of our customers. Earning this notable certification gives them an added level of confidence that our products conform to industry interoperability and performance standards.”

Conexant’s WorldRadio includes the company’s innovative PowerSave power management technology, which significantly reduces power consumption. The combination of this technology and WMM Power Save features will help drive the proliferation of Wi-Fi-enabled mobile devices.

Conexant offers an extensive suite of WLAN solutions including IEEE 802.11a/b/g and dual-band (2.4 and 5 GHz) chipsets, firmware, software, drivers and reference designs. These wireless networking solutions are used by the world’s leading telecom, networking and computer companies in a wide range of products including access points/routers, client cards, desktops, notebooks, PDAs, digital cameras, MP3 players and other handheld networking appliances. They are available as standalone solutions or offered in conjunction with Conexant’s DSL and cable modem semiconductor system solutions, voice over IP (VoIP) chipsets and home network processors.

http://www.conexant.com

Vonage Claims To Offer All Subscribers 911 Access

Vonage America, (news - alert) a subsidiary of Vonage Holdings Corp., announced today that all of its customers now have access to 911 services. Today, any Vonage customer in the U.S. who dials 911, will get help when they need it most.

According to the company, here's how it works: Vonage currently offers E911 or enhanced 911 throughout the U.S., by sending the call, along with the customer's address and phone number, to the proper local emergency call center based on the caller's street address. The caller's information is then displayed on the dispatcher's screen whenever they dial the digits 9-1-1 from a Vonage phone. In the event local authorities cannot display the Vonage customer's phone number or address, Vonage offers basic 911. Basic 911 is a service in which the customer's emergency call is delivered through the traditional 911 network and the call is answered by a trained dispatcher in the local public safety answering point (PSAP), or 911 call center. Finally, Vonage provides an additional safety net to customers if traditional 911 methods fail or are not available: customers receive support from emergency trained personnel at Vonage's national 911 emergency response center, which is supported by APCO 33 trained personnel and serves customers throughout the U.S.

"Vonage shares a common goal with the FCC, Congress, public safety officials and regulators: to deploy E911 service for all subscribers as soon as possible," said Jeffrey A. Citron, Vonage's Chairman and CEO. "We are dedicating every resource at our command toward turning up a national E911 system — not only are we spending over $50 million, but we're working with public safety to create a framework to hasten this national deployment and remove any local roadblocks we've encountered."

http://www.vonage.com

New Avaya IP Solution Simplifies Business Communications for Telecommuters and Remote Workers

Avaya, (quote - news - alert) a leading global provider of business communications applications, systems and services, today announced a new telecommuter solution that embeds virtual private network (VPN) remote capabilities into Avaya's family of IP telephones. Businesses can use this solution to more easily and securely extend headquarters-quality communications to employees working from any home office, temporary work site or emergency location. By integrating new Avaya VPNremote(TM) software within the phone, telecommuters have an "always on" business-class IP telephone that contains all of the functions required for highly-productive and continuous enterprise voice communications.

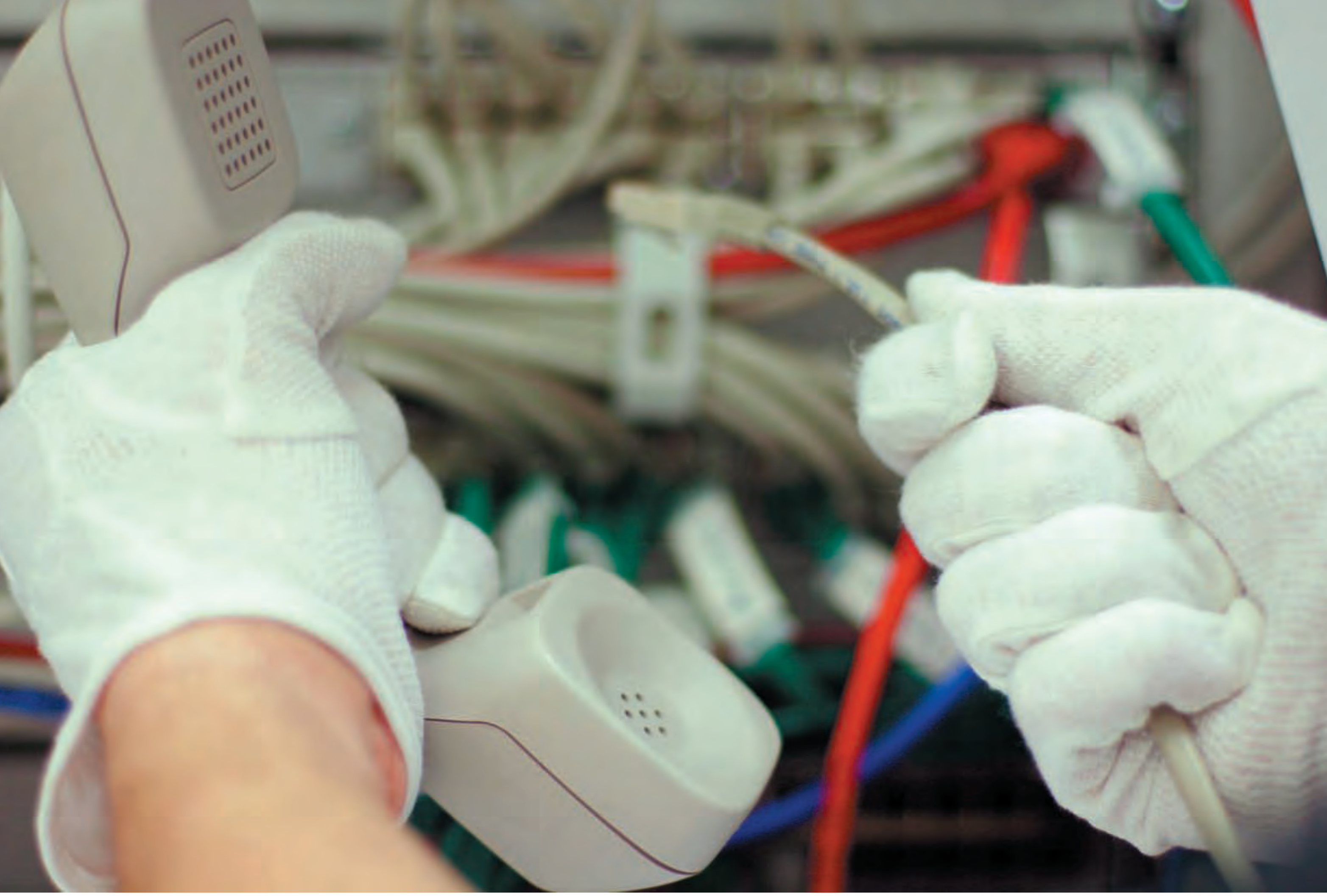
The solution — called VPNremote for Avaya 4600 IP Telephones — enables telecommuting employees to quickly and cost-effectively install the desktop IP phone in their home office with minimal equipment or IT assistance, similar to how a laptop PC is connected for remote access via Internet. No other equipment or software is required in the home office. This simplicity can also be applied to temporary remote deployments, such as when business teams gather at trade shows, corporate events or other temporary work locations.

This solution drives greater business continuity by enabling companies to swiftly deploy Avaya IP phones with VPNremote to necessary locations when responding to a disaster recovery scenario or disruption in network communications. An organization can cost-effectively deploy the phones to dispersed and remote workers anywhere, knitting together a secure communications network that keeps business running and workers productive under the most challenging conditions.

The features of the Avaya IP phones with VPNremote are identical to those found in a corporate headquarters or contact center, making them ideal for home-based contact center agents, telecommuting sales people or other remote employees that must access enterprise telephony capabilities daily.

"In order for telecommuters in a home office or remote location to be productive, they require a business communications experience that is secure, highly-functional and equal to that within a corporate headquarters," said Allan Sulkin, president of TEQConsult Group. "Avaya's new solution helps improve the performance capabilities of the telecommuter, and can be a lifesaver for executives who need to operate in a virtual workspace."

http://www.avaya.com



Zultys Announces Wireless Conference Phone — Unique Product Offers True Conferencing Over VoIP

Zultys Technologies (news - alert) today announced the BTC phone, a powerful wireless conference phone that is compatible with its range of SIP IP phones as well as mobile phones. The BTC (Bluetooth Conference Phone) provides unparalleled speech quality and removes clutter from the conference room table.

The BTC phone is incredibly easy to use, having just three buttons. The phone is paired with any Bluetooth phone such as the Zultys ZIP 4x5 IP phone. Calls can be originated on the ZIP 4x5 phone or on the BTC phone using voice activated dialling. Calls can be answered on either phone. A conference call with multiple parties can be made using the features of the ZIP 4x5 phone. With its unique Bluetooth capability, these features are also available in conjunction with any other Bluetooth phone, including cell phones.

The BTC phone is aesthetically designed to enhance any conference room. Its batteries power the unit for about a week, obviating the need for power cables and phone lines to be dragged across the conference room table. The portability and use of open standards permit the phone to be moved from room to room or to a remote facility-even your car. The phone has four microphones and a powerful DSP (Digital Signal Processor) ensuring clarity of both the listeners as well as those at the other end of the call. The phone's speaker output is best in its class and the acoustic echo cancellation permits a natural conversation.

This BTC phone rounds out a range of SIP phones from Zultys, which includes Ethernet and Wi-Fi sets. It permits an exclusively SIP installation to add conference phones without running wires to carry analog signals. The phone has four quad-color LEDs that display its status. The phone also is equipped with standard 3.5mm connectors (pink and green) so that it may be connected by wires to a desktop phone or a soft phone running on a PC. This enhances its flexibility for any application-wired, or wireless.

"When people think of conference room phones, they usually think of one supplier," said Patrick Ferriter, VP of Product Marketing at Zultys. "This phone will radically change the perception of what a conference phone should be like. Instead of stagnant features, Zultys has demonstrated clear technological leadership with this product."

http://www.zultys.com

SeaMobile Signs Agreement with Silversea Cruises to Deliver Unmatched Wireless Voice & Data Services to Personal Mobile Devices at Sea

SeaMobile Inc., (news - alert) a leading global provider of advanced at-sea wireless voice and data communications services, announced it has signed an agreement with Silversea Cruises, (news - alert) an award-winning, ultra-luxury cruise line, to provide never before offered communication solutions for its passengers throughout the world.

The first Silversea ship to deploy the SeaMobile advanced wireless services will be the Silver Whisper. Passengers and crew aboard the ship will be able to use their own cellular phones and wireless PDAs while at sea, just as they do on land. Charges for calls and data services while at sea will simply appear on the wireless customer's bill from their home carrier.

SeaMobile's leadership team includes proven business and technology leaders who have repeated success in using technology advantages to successfully deliver services to significant markets over the past two decades. They were the founders and growth leaders for McCaw Cellular Communications, AT&T Wireless Services, Microsoft, DSS Direct/DIRECTV, Nextel and other top telecommunications and software companies.

SeaMobile's proprietary technology is a sophisticated IP/software based solution that is "agnostic" to the type of phone (GSM, GPRS or CDMA) used by the wireless customer when accessing the SeaMobile network at sea. This allows virtually anyone aboard any vessel at sea to use voice and data services available through their wireless home carrier, just as they would on land. In addition, SeaMobile's solution can work with any satellite provider, including those currently offering services to the maritime industry. Worldwide roaming agreements established by SeaMobile provide transparent connectivity for wireless services. Coverage aboard the Silver Whisper can include the entire ship or select areas to enable "quiet zones."

"Personalized service, flexibility and convenience are all important elements of the ultra-luxury experience we offer our guests," said Silversea CEO Albert Peter. "We chose SeaMobile because of their ability to provide a superior wireless experience for our guests, giving them seamless and convenient cell phone access to family, friends and business associates not only on land, but also on the ship."

""We are thrilled to sign this agreement with Silversea to support their primary goal of offering an ultra-luxury experience for their guests by helping them stay connected," said SeaMobile President and CEO William D. Marks. "Cruising is a fast-growing industry where improved connectivity is becoming a key decision factor among consumers who choose a vacation destination based on their ability to stay connected for business and personal needs."

http://www.seamobile.com
http://www.silversea.com

CDC Deploys MobileAccess Universal Wireless Solution at Renovated Campus Headquarters

MobileAccess Networks, (news - alert) an enterprise wireless innovator, announced that the company's Universal Wireless Network has been installed at the newly constructed Centers for Disease Control and Prevention (CDC) campus in Atlanta, Ga. The MobileAccess solution provides seamless coverage for multiple wireless applications and services, including mobile phones, BlackBerry® devices, PDAs and Wi-Fi for over 600,000 square feet, which includes 19 floors in multiple buildings.

Totaling $1.5 billion, the rebuilt CDC campus enables the agency to house many of its 9,000 employees at a secure, modern facility that combines leading-edge technology and pragmatic creature comforts. As part of the renovation project, the MobileAccess Universal Wireless Network provides a next-generation platform that delivers connectivity for a broad range of wireless technologies throughout the campus to promote increased convenience and productivity for CDC workers. In keeping with the facility's progressive outlook, the MobileAccess solution features flexible and scalable common infrastructure that accommodates today's multi-service wireless needs yet readily adapts to the agency's evolving future requirements.

The MobileAccess Universal Wireless Network is ideally suited to address CDC's cost and capability requirements. By deploying best-of-breed technology solutions like MobileAccess, CDC is not only being fiscally responsible, but capable of providing real benefits to its employees, who in turn can work more efficiently and effectively.

"The MobileAccess Universal Wireless Network is the solution of choice for customers who require seamless in-building wireless connectivity, the flexibility to support a broad range of wireless technologies, and the investment protection of a future-proof platform that adapts and scales to meet evolving wireless requirements," said Cathy Zatloukal, president and CEO of MobileAccess. "MobileAccess is honored to play a vital role in supporting the thousands of CDC employees tasked with protecting health and safety."

http://www.mobileaccess.com

QUALCOMM Reinforces Commitment to WLAN Market, Joins Wi-Fi Alliance

QUALCOMM Incorporated, (quote - news - alert) a leading developer and innovator of Code Division Multiple Access (CDMA) and other advanced wireless technologies, announced that the Company has joined the Wi-Fi Alliance. The Wi-Fi Alliance is a global, non-profit industry trade association with more than 200 member companies devoted to promoting the growth of wireless local area networks (WLAN). By joining the Wi-Fi Alliance, QUALCOMM will be directly involved with ensuring the compatibility of Wi-Fi technology with the Company's Mobile Station Modem™ (MSM™) chipsets.

"QUALCOMM is pleased to take the important step of joining the Wi-Fi Alliance to help ensure the interoperability of WLAN technologies with CDMA2000, WCDMA, and other wireless standards supported by our solutions," said Ed Tiedemann, senior vice president of engineering for QUALCOMM. "We look forward to working closely together on the interoperability of 802.11 a, b, and g standards — as well as the expected 802.11n standard — with cellular networks, helping ensure seamless compatibility across our product portfolio and driving the convergence of mobile capabilities."

"In becoming a member of the Wi-Fi Alliance, QUALCOMM joins more than 200 companies from around the globe who are committed to delivering interoperable Wi-Fi solutions and ensuring a positive user experience through our Wi-Fi CERTIFIED programs," said Frank Hanzlik, Wi-Fi Alliance managing director. "We are pleased to have them become a part of the organization."

With the aim of enhancing the user experience for mobile wireless devices, the Wi-Fi Alliance's testing and certification programs ensure the interoperability of WLAN products based on the IEEE 802.11 specification. Since the introduction of the Wi-Fi Alliance's certification program in March 2000, more than 2,000 products have been designated as Wi-Fi CERTIFIED™, encouraging the expanded use of Wi-Fi products and services across the consumer and enterprise markets.

http://www.qualcomm.com

Mario Kart DS Launches with Wi-Fi Gaming Service

That sound you hear is the thunder of video gamers worldwide revving their engines for the launch of *Mario Kart*® DS and Nintendo® Wi-Fi Connection. For the first time ever, the online gaming service for Nintendo DS™ lets users connect wirelessly to race. From the very start, users can log on easily to shoot shells and burn rubber against opponents from all around the world.

"Nintendo (news - alert) Wi-Fi Connection removes the barriers that have prevented many mainstream players from going online," says Reggie Fils-Aime, Nintendo of America's executive vice president of sales & marketing. "We have made the service incredibly simple to use and free at select hotspot locations — but this is just the beginning."

Whether users are at home or out and about, they have a variety of uncomplicated ways to connect to Nintendo Wi-Fi Connection.

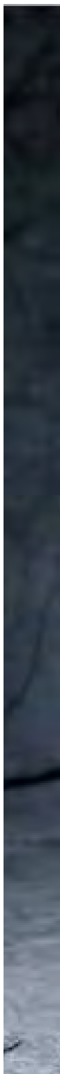
If the user already has a Wi-Fi home network, the Nintendo DS unit will connect directly with minimal setup procedures. Those who have a home high-speed Internet connection such as cable or DSL, but no Wi-Fi home network, can either purchase a compatible wireless router or the Nintendo Wi-Fi USB Connector, which plugs into the USB port of a PC running Windows XP to create a wireless access point to connect a DS to Nintendo Wi-Fi Connection. The Nintendo Wi-Fi USB Connector will be sold exclusively on Nintendo.com at an MSRP of $34.99.

Outside the home, when a user brings a Nintendo DS unit and a Wi-Fi-enabled game into a Wayport-enabled McDonald's restaurant locations, the user simply launches the game in Nintendo Wi-Fi Connection mode. No setup is required.

Nintendo DS owners who want to play games at Wi-Fi locations outside McDonald's restaurant locations or their homes can use a laptop computer and the Nintendo Wi-Fi USB Connector to access Nintendo Wi-Fi Connection. The Nintendo Wi-Fi USB Connector enables the Nintendo DS to share an Internet connection established by a PC, providing a conduit to play games via Nintendo Wi-Fi Connection.

Nintendo DS owners may also connect at a number of other free hotspot locations with no additional hardware and minimal setup.

http://www.nintendowifi.com



WiFi's Dirty Little Secret — You're Not Safe!

Protect Your Identity and Your Data When Using a WiFi 'Hotspot' — StealthSurfer II Makes Wireless Surfing Safe and Worry-Free

Stealth Ideas Inc., (news - alert) a portable privacy software company, announced that its recently unveiled StealthSurfer II device will now allow for 100 percent secure and encrypted wireless Web browsing. StealthSurfer II is a thumb-sized flash storage drive which allows consumers to surf the Web with anonymity from any computer — even over a wireless connection. StealthSurfer II's integrated privacy features keep surfers safe from identity theft as well as from phishing and pharming spam attacks. Ideal for frequent coffee shop surfers, airport travelers, Internet cafe-goers or anyone who uses their laptop wirelessly in a public hotspot, the StealthSurfer is the most robust and complete privacy and identity protection tool on the market.

Wireless networks in public areas or hotspots (like Internet cafes) often do not provide adequate security. A recent polling revealed that most public and private WiFi networks used no encryption at all. Moreover, even when the host does have the proper security settings enabled, consumers still leave themselves at risk to knowledgeable hackers or wardrivers (those who use easy-to-obtain 'sniffing' software to find unprotected networks). Once penetrated, an experienced attacker could implant malicious programs, including spyware, adware or Trojan horse applications, directly onto a computer.

With StealthSurfer II, wireless Internet access is totally safe and secure. By using the device's customized FireFox browser and integrated IP masking, an encrypted tunnel is created between the surfer's laptop and the receiving Web site. This encrypted data stream cannot be read by anyone but the host computer. Any data packets intercepted would appear as illegible data garbage rendering the attack useless. All user data (such as confidential credit card and social security numbers) remains protected and secure.

Tiny enough to carry on a keychain, the USB 2.0 flash drive plugs into the USB port of a computer and allows users to surf the Web with total privacy. The device is available in memory configurations of 128 megabytes to one gigabyte at pricing starting at $89. Small, sleek and versatile, StealthSurfer II offers the very best in portability, privacy and identity theft protection.

StealthSurfer II is loaded with a host of privacy-protection tools that are seamlessly integrated into one tiny and portable keychain device. With all these tools "under one hood," users can not only protect their laptop or personal computer, but can now take this virtual "armor" with them wherever they go — even when traveling or using public computers.

http://www.stealthsurfer.com

Vapps Integrates Web Conferencing into CB1000 Creating First IP Based Audio and Web Conferencing Service Platform

Vapps, (news - alert) a leading global provider of audio conferencing systems for IP and TDM networks, announced a strategic partnership with WebDialogs (news - alert) to incorporate Web conferencing technology into its flagship CB1000 audio conferencing platform. The enhancement will allow carriers and service providers to offer the first collaborative voice and data conferencing service simultaneously over legacy PSTN and next-generation IP networks.

The central component of the CB1000 enhancement is WebDialogs' WebInterpoint™ architecture, enabling end users to share documents, presentations, Web content, and applications online and in real-time. With its flexible design, and open API's, WebInterpoint™ easily integrates into the Vapps platform, giving service provider customers an easy to use interface for hosting or joining audio/web conferences over any type of network utilizing any type of access device including: traditional handsets, cellular phone or IP devices.

"It's important in today's evolving telecom industry to offer high quality solutions that support both legacy and newer IP infrastructures," said Ben Lilienthal, CEO and co-founder of Vapps. "WebDialogs has given us the ability to offer the industry's first collective voice and data conferencing solution that works across all communication platforms, an invaluable competitive advantage for us.

CB1000 is a SIP-enabled, carrier-grade conferencing platform that is VoIP native and delivers reliable, seamless reservation-less conference calls on both legacy and IP-based telecom systems. Features include: web-based call control, standard and advanced call management (lists, mute, volume control, conference locking and recording), operator support and call record details among others. The product supports up to 18,000 total conference participants in multiple simultaneous conferences, with the ability to easily scale on a card-by-card basis.

"We've built our business by providing our customers premier web conferencing services that are easy to use," stated Lou Guercia, CEO of Wedialogs. "Our partnership with Vapps furthers our initiative by leveraging the CB1000's hybrid architecture in order to enter into new arenas such as the exploding VoIP marketplace."

http://www.vapps.com
http://www.webdialogs.com

Acme Packet Joins Level 3's (3)VoIP Technology Alliance Program

Acme Packet, (news - alert) the leader in session border control, announced that it has been selected by Level 3 (news - alert) for participation in its (3)VoIP Technology Alliance Program (TAP). The (3)VoIP TAP program formalizes the relationships between Level 3 and a select group of best-of-breed technology vendors to help service providers quickly deploy services interconnecting with the Level 3 network.

Acme Packet's Net-Net family of session border controllers (SBCs) satisfies critical security, service assurance and regulatory requirements for voice, video and multimedia sessions crossing IP network borders. Acme Packet's NetSafe™ architecture sets the bar for session border controller security by protecting the SBC itself against DoS/DDoS attacks and overloads; the service provider's service infrastructure (e.g. SIP servers, softswitches, application servers, media servers or media gateways); and subscriber, enterprise and service provider security including confidentiality and privacy. Now Application Service Providers, Broadband Service Providers, or anyone wishing to offer advanced multimedia services across IP network borders can connect to Level 3's selection of wholesale transport services.

"As a market leader in session border controllers, Acme Packet is a natural fit for the (3)VoIP TAP program," said Craig Schlagbaum, vice president of Channel Development for Level 3. "Acme Packet is certified in our interoperability labs and we share many joint customers that interconnect with the Level 3 network. We look forward to working together as VoIP, video and other multimedia services continue to gain momentum in the market."

(3)VoIP TAP provides valuable interoperability testing between Level 3 and numerous IP equipment vendors and network providers. The program speeds time to market by increasing confidence in vendors solutions that have already been proven with Level 3's wholesale services network including (3)Voice, (3)VoIP Enhanced(SM) Local, (3)VoIP Toll Free(SM), and (3)VoIP(SM) Local Inbound.

"Acme Packet is honored to be one of the few technology vendors to participate in the (3)VoIP TAP," remarked Seamus Hourihan, vice president product management and marketing, Acme Packet. "This program is one of the first of its kind in the industry dedicated to working with vendors to enable service provider connections to Level 3's network in a quick and efficient manner."

http://www.acmepacket.com
http://www.level3.com

Lanck Telecom Using MERA VoIP Transit Softswitches

By Patrick Barnard, TMCnet Associate Editor

Russia-based VoIP wholesaler Lanck Telecom (news - alert) announced that it will be using MERA Systems' MERA (news - alert) VoIP Transit Softswitch (MVTS) to control both signaling and media flows across IP network borders — a move that will help establish Lanck as an international exchange carrier (IEC).

Along with regular session control functions, including NAT/firewall traversal, network topology hiding and call admission management, the MVTS performs intelligent call routing and brings together incompatible networks and equipment from a diversity of vendors. MERA says the MVTS works equally well with both H.323 and SIP signaling standards.

With points of presence (PoPs) in all the major cities of Russia, as well as international cities including Frankfurt, Hong-Kong, and London, Lanck is a fast-growing carrier whose core business is selling VoIP minutes to partner carriers across the globe. The deployment of MVTS provides Lanck with an end-to-end solution for intelligent management of call flows in its distributed VoIP network, and also allows interoperability with peering partners to ensure they can quickly and cost-effectively interconnect with new partners.

To serve Lanck's needs, MERA has offered a distributed, cluster solution that can handle up to 10,000 concurrent calls. In addition, MERA's solution delivers a customer-tailored configuration that seamlessly integrates into Lanck's network and takes into account all the peculiarities of the network topology and the carrier's business processes. MERA says the distributed solution can scale quickly and easily to accommodate Lanck's aggressive growth.

"We have designed this three-layer cluster specifically for Lanck Telecom," said Konstantin Nikashov, CEO of MERA Systems, in a news release issued today. "This project demonstrates the level of flexibility that our softswitching technology can provide."

Sergey Alekseev, CIO of Lanck, said the new cluster has increased network reliability "from 99.9% up to 99.999%."

"Earlier, our network downtime would reach up to several hours a year, which caused us significant financial losses," Alekseev said. "Now, the system is completely fault-proof, as all the components including the database are fully redundant."

http://www.lancktelecom.ru
http://www.mera-systems.com

Artesyn Communication Products and Surf Communication Solutions Partner to Offer Triple-Play Media Gateway and Server Blades

Artesyn Communication Products, (news - alert) a subsidiary of Artesyn Technologies, announced that it has partnered with Surf Communications Solutions ("Surf") (news - alert) to develop open architecture, Triple-Play voice, video, and data (fax/modem) media processing solutions. Artesyn will combine its PICMG 2.16 and AdvancedTCA blades with Surf's PTMC and AdvancedMC DSP farms to create commercial, off-the-shelf media gateway and media server subsystems that deliver voice, video, fax and modem over IP, mobile, wireline, and wireless networks.

"Surf's extensive DSP hardware, software and media processing expertise makes them an ideal partner for creating open architecture, Triple-Play media processing solutions," said Todd Wynia, vice president of marketing at Artesyn. "Our PICMG 2.16 and AdvancedTCA blades, together with Surf's AdvancedMC and PTMC media processing DSP farms, will give telecom OEMs an out-of-the-box solution for building a broad range of modular, open architecture media processing gateways and servers that offer Triple-Play capability."

"Artesyn's open architecture blades provide an excellent platform for our AdvancedMC and PTMC media processing DSP farms, said Amir Zmora, vice president of marketing at Surf. "The resulting integrated solution provides a comprehensive 'hardware and software', high-availability, high-density infrastructure subsystem for telecom equipment manufacturers that greatly reduces their time to market."

http://www.artesyn.com
http://www.surf-com.com

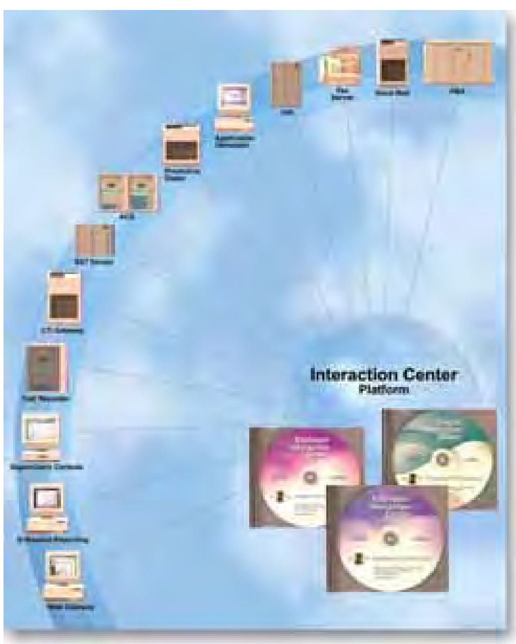
Interactive Intelligence Announces New Version of IP Contact Center Software

Major upgrade delivers increased scalability, enhanced agent optimization, and superior enterprise messaging capabilities

Interactive Intelligence Inc., (news - alert) a global developer of business communications software, announced availability of a major upgrade to its flagship IP contact center software, Customer Interaction Center® (CIC).

The new version of CIC provides three important new capabilities. First, the release introduces a new session initiation protocol-based Interaction Media Server™, which provides a dramatic increase in IP system scalability and reliability by transferring the majority of all media processing to the dedicated media server. Second, CIC delivers additional agent optimization enhancements including screen recording, whisper coaching and agent alerts. The final addition to the release targets the product's enterprise telephony users with enhancements to the voice messaging component of the offering; including a speech recognition interface for message management and the addition of customer-configurable voicemail menus.

"This release is a significant step forward, with clearly demonstrable cost savings and functionality benefits for our customers. With these additions, CIC becomes a viable option for nearly any size contact center," said Dr. Donald Brown, Interactive Intelligence president and chief executive officer. "We have continued to respond to customers who want the rich functionality of CIC in larger size deployments. With the addition of the SIP Interaction Media Server, we are delivering improved scalability and reliability at a reduced cost for larger deployments. The other product enhancements associated with this release continue to broaden our appeal as an innovative enterprise-wide alternative to competitor's proprietary, hardware-centric offerings."



The addition of the new SIP Interaction Media Server increases scalability by moving audio recording and audio processing from the CIC server to the dedicated media server. In CIC deployments that include voice board hardware, scalability has more than doubled to 1,000 supported agents per system. In an all-software deployment of CIC, scalability has increased about 12 times, from 25 agents to approximately 300.

By eliminating a single point of failure, Interaction Media Server also gives organizations the added benefit of increased reliability. The architecture of the Interaction Media Server means that in the event of a failure of the CIC server, calls remain connected without user disruption.

Additionally, because the media server also reduces the reliance on third-party software and hardware in many configurations, contact centers can leverage much larger VoIP deployments at cost savings of up to 40 percent.

Affordable, integrated agent optimization functionality becomes a reality with the latest version of CIC. Additions include screen recording — synchronizing screen recordings with audio recording files. The addition of multimedia screen recording extends management's ability to mentor and improve agent effectiveness for all interaction types.

CIC 2.4 also introduces support for real-time and alert-based agent messaging, agent real-time queue status, and whisper coaching. All of these features are designed to increase agent productivity and improve customer service levels.

http://www.inin.com

Signate Announces PBX Management Software for Telephone Service Providers

Signate, (news - alert) the leading provider of VoIP telephony solutions that combine high performance hardware and open source software, announced SigMAN SP, its Hosted PBX (Private Branch eXchange) software for telephone service providers with up to 3,000 customers. Hosted PBXs using SigMAN SP can support remote businesses, remote offices and remote employees with VoIP extensions.

A superset of the functionality Signate introduced in September with SigMAN, its PBX management software, SigMAN SP turns a Linux-based computer into a full featured, partitioned PBX that allows providers the ability to offer their customers advanced telephone capabilities without the capital investment of a dedicated system. Each customer partition can function like a dedicated telephone system with its own auto attendant(s), music on hold, operator(s), call queues, audio conference rooms, hunt groups, voice mail and call detail reporting. Providers may give customers as much, or as little, access as their businesses require, so customers can manage their own facilities.

SigUSERtm, the included user panel, gives PBX users the ability to manage their own voicemail, call forwarding and follow me settings from a web browser without going through a PBX administrator.

"Every Signate product has a foundation of industry standard hardware and open source software so it can deliver roughly twice the performance of proprietary products at half the cost," said William Boehlke, Signate's Chief Executive Officer. "SigMAN SP delivers advanced functionality to service provider customers for an initial investment of around $10 per user," he said.

SigMAN SP can be configured with a hot failover server, so if a server fails its function can be automatically assumed by another system. Signate further backs SigMAN SP with support around the clock and around the world, from offices in the United States, England and New Zealand.

SigMAN SP requires an Intel x86 server capable of running Red Hat ES Linux, or Signate's high capacity Telephony Server 5000. A PCI board or external gateway is necessary to connect the PBX to the public switched telephone network. A high speed internet connection is required to link the PBX to VoIP telephone service providers. Telephone handsets connect over a standard computer network.

http://www.signate.com

Tatara Systems First to Deliver Converged Mobile VoIP; Allows Mobile Operators to Significantly Enhance Laptop Mobile Office Service

Tatara Systems, (news - alert) the market leader in mobile services convergence, announced availability of its Converged Mobile VoIP solution to enable nomadic VoIP services using a subscriber's mobile phone number across any IP network, with the first application as an integrated laptop solution for mobile operators. Tatara has entered market trials with major carrier customers marking an industry first where an end-user subscriber is able to maintain their mobile phone number identity to both make and receive VoIP calls across a variety of access networks using converged mobile devices such as laptops.

This offering gives mobile operators, both traditional and MVNOs, the opportunity to participate in the growing VoIP market by providing a significantly better, integrated end-user solution than you can get today via a standalone softphone service. "One-number" VoIP yields a quick return on investment for carriers looking to deliver new service revenues leveraging their existing networks, while building the foundation for a graceful transition to more advanced IMS applications.

"Convergence is the most important trend in communications — and not just wireless — today," said Craig Mathias, a Principal with the wireless and mobile advisory firm Farpoint Group. "One-Number VoIP will be very popular with end users, and offers mobile operators a unique opportunity to retain customer ownership while gaining from the proliferation of VoIP service offerings entering the market."

With Converged Mobile VoIP solutions, mobile operators are now able to extend 2G/3G voice services for their subscriber base across a variety of private and roamed networks. This provides a valuable service extension for customers traveling outside the 2G/3G coverage, such as in-building or internationally. In addition to delivering a new revenue generating service, Converged Mobile VoIP also enables mobile operators to retain customers considering competitive VoIP offerings by delivering a far superior product offering. Having already delivered Converged Mobile Messaging, Tatara continues to be the industry leader by being the first to deliver Converged Mobile VoIP services enabling customers to have a significant time to market advantage leveraging key capabilities of the Tatara Mobile Services Convergence Platform.

"Mobile operators are now seeing significant subscriber adoption of their laptop-based 3G/Wi-Fi mobile office service by their most valuable customer segments," said Steve Nicolle, CEO and President of Tatara. "Our solution allows them to utilize the advantage they have over standalone VoIP providers by delivering a consistent and converged set of voice and data services under one brand across the end-user's two most significant devices - their handset and their laptop."

http://www.tatarasystems.com

Brix Networks Announces Enhanced IP Video Quality Algorithm To Ensure Success In IPTV Rollouts

Brix Networks, (news - alert) the trusted provider of the most widely deployed service assurance solutions for interactive applications, today announced an enhanced IP video quality algorithm — the Brix Video Quality Index (Brix VQI) — that ensures providers' success with their IPTV rollouts.

"In order to effectively deliver profitable triple-play services, cultivate customer loyalty, and grow market share, it is essential that network operators provide their subscribers with high-quality offerings," said Robert Travis, director of product marketing, Brix Networks. "Our VQI technology is a critical component of our comprehensive service assurance platform, and helps determine the overall quality of a variety of existing and emerging IP-based video applications, such as IPTV, video-on-demand, and interactive video."

The Brix VQI is an objective measurement scale for video over IP networks and applications providing accurate, easy-to-understand metrics for highly distributed, large-scale carrier and enterprise networks. The Brix VQI incorporates the impact of network transmission quality — the most important factor contributing to video quality — and quantifies the effect latency, packet loss, jitter, packet discards, buffering, and re-buffering events have on a video service.

With today's announcement, the Brix VQI has been enhanced to be "application aware," meaning it can distinguish, analyze, and correlate the key performance indicators (KPIs) related to data impairments that, in turn, contribute to picture jerkiness, blurriness, or no picture in video-based services.

"Developed with ongoing standards initiatives in mind, Brix VQI is an innovative video quality algorithm that yields useable metrics to generate an IP video quality score — per video stream — that accurately represents the user's quality of experience," said Brix Networks' chief technology officer, Kaynam Hedayat. "VQI builds on our extensive industry-leading experience in VoIP networks, and quantifies the impact that latency, initial wait time, network jitter, packet loss, and other application-level impairments have on the overall performance of an IP video service."

http://www.brixnet.com

Go To Table of Contents | Go To Ad Index

Pingtel Transcends Traditional PBXs by Enabling True, Standards-Based Real-Time Communications

Pingtel Corp., (news - alert) the leading provider of open source, commercial-grade enterprise communications solutions, announced the availability of the SIPxchange Enterprise Communications Server (ECS), a fundamentally new approach to addressing the convergence of voice and data within the enterprise market. Based on the 3.0 release of its award winning SIPxchange PBX, SIPxchange ECS takes enterprise communications to the next level with a whole new set of capabilities, including an integrated presence server, automatic call distribution (ACD) services based on presence, support for additional SIP phones, and a new, highly extensible system management platform.

"Enterprise customers are increasingly less inclined to accept technologies into their network that do not properly account for future needs. By delivering the first 100% standard SIP solution with integrated presence that is capable of unifying all types of enterprise communications, Pingtel is delivering on enterprise expectations," said William J. Rich, president and CEO of Pingtel. "Pingtel's breakthrough SIPxchange ECS is the next logical and future proof step in enterprise communications that enables the enterprise to fully leverage a robust suite of IP-based communications services through one integrated, standards-based, real time platform."

Based on the open source sipXpbx project from SIPfoundry, SIPxchange ECS includes the features of traditional PBXs, (define - news - alert) but is the first solution that recognizes that SIP and presence will form the core of enterprise communications and delivers both major functions — SIP and presence infrastructure — in one highly extensible, highly scalable platform.

The SIPxchange ECS's open source heritage provides customers with an ideal combination of functionality and value. In addition to PBX features, the SIPxchange ECS includes core business applications, including an automatic call distributor (ACD), and a built-in presence server for true next-generation applications. Coupling the power of standards based SIP with open source development enables Pingtel to introduce the SIPxchange ECS at a price point up to 50 percent less than traditional vendors.

http://www.pingtel.com

Brekeke Software Announces Coming Release of OnDO SIP Server v1.5 with Enhanced Dial Plan Functionality

Brekeke Software, Inc. (news - alert) announced they will release version 1.5 of their OnDO SIP Server, a software proxy and registrar server, used to register and authenticate VoIP calls, then route calls between user agents. This software release includes improved dial plan features that increase compatibility between SIP compliant products and provide added capability for creating complex and powerful dial plans.

With the growing popularity of SIP (define - news - alert) technology, and the variety of SIP devices to choose from, compatibility between SIP servers and SIP devices is fast becoming a concern for many users. Brekeke's OnDO SIP Server can help solve these issues by providing the flexibility to rewrite, insert, or delete SIP headers according to rules defined in the web-based administration tool. These rules allow the OnDO SIP Server to easily accept any SIP compliant products used throughout the system. For example, a rule can be defined to edit a SIP header depending on the incoming call and the SIP device used by the caller. These dial plan settings are easily changed through the web-based administration tool.

At $200 per license, the OnDO SIP Server provides an affordable software alternative for those not wanting to purchase costly hardware products. For educational institutions and individual users, it's free to use, if you qualify under licensing requirements. A trial version is available to all users for a 30-day evaluation period from http://www.brekeke.com.

http://www.brekeke.com

Moscow City Telephone Network Applies FrontRange IPCC to New Automated Systems

Russian Communications News service ComNews.ru reported that Moscow City Telephone Network (MGTS) has launched a new service for 09 directory assistance. This service, which increases bandwidth via automation and voice recognition, was developed for MGTS by FrontRange Solutions, known in Russia for its CRM solutions. The system includes an integrated speech recognition module from technology leader Nuance Communications.

With the new call processing service, the Automatic 09 Operator software/hardware system, subscribers can quickly obtain any of 2,000 essential telephone numbers. The free service offers the option of using voice commands instead of touch-tone buttons, to navigate the menu; a speech-recognition system processes the commands.

The system divides directory assistance inquiries into several categories: Emergency Assistance, City Government, MGTS Information, Transportation, Repair Services, and Medical. MGTS plans continuous updates to this list wherein the automatic operator suggests menu items to subscribers such as: "Welcome. For emergency assistance, say: 'Assistance.' To switch to the Transportation menu, say: 'Transportation.'"

The automatic operator currently has 5,000 words in its active dictionary. The system's speech recognition accuracy is 95% and its software can process calls over 210 telephone channels simultaneously.

Dmitry Stoliar, FrontRange Solutions (news - alert) Sales Director in Russia, explained that the setup of the system started in March 2005 and launched test runs of the service in July. At the same time, its capacity was built up so that in September it reached the target level of seven E1 circuits, or 210 channels, each of them with speech recognition technology in operation. The system is easily scalable and allows unlimited expansion of the directory database.

http://www.frontrange.com

The Benefits Of Blended Agents

By Tracey E. Schelmetic, Editorial Director, Customer Inter@ction Solutions

In an era when some call centers report annual turnover rates among agents of 100 percent or more, companies are having to think more carefully about retention. The aggregate costs of replacing an agent by scouting, testing, hiring and training a replacement are staggering — the estimated numbers run from $6,000 at the low end to $10,000 and beyond at the high end. If your organization has 500 call center agents and your turnover is 100 percent per year, you are easily spending a half million dollars or more each year on turnover alone. For a large call center, shaving only a few percentage points off turnover rates can add up to considerable savings.

In the days when call center jobs were commodities, agents would leave for the company across the street for an additional 25 cents per hour. The trick is to make sure agents perceive that working for your call center is not merely a necessary evil that generates a much-needed paycheck. Most call center organizations are catching on, adding perks such as child care, attractive break rooms, regular appreciation lunches, exercise facilities, quality cafeterias, flexible scheduling, periodic visits by massage therapists and a host of other appealing services.

It's no secret that turnover among outbound agents soars above that of inbound agents. Outbound is hard work. It takes persistence, an optimistic attitude and a very thick skin to be a good outbound agent. Some agents thrive in the position. Others burn out very quickly, taking hang-ups and rudeness personally. But let's face it: someone has to do outbound work. It's still big business in the U.S.

Technologies that allow the blending of outbound and inbound within the same agent group can cure a lot of what's behind the turnover in the call center. Though many companies have claimed for years that they blend outbound and inbound within the same agent pools, the process was cumbersome and limiting. Data silos, differing inbound and outbound applications and a patchwork of phone hardware made it more of a nightmare than coping with agent turnover.

Using IP telephony, this "disconnect" between inbound and outbound can be solved. Inbound agents can be switched to outbound during slow times, and outbound agents can be drafted to inbound work during call spikes. All customer information stays in the same place. Convenience and better customer service aside, a blended schedule generally makes for a happier agent. Conventional wisdom says that part of the cause of high turnover is boredom. Variances in agent duties during the day keep agents more stimulated: not only performing both inbound and outbound, but handling chat and e-mail, as well. Agents are more likely to perceive their jobs as careers rather than paychecks, and view themselves as professionals rather than "butts in chairs."

http://www.tmcnet.com

Go To Table of Contents | Go To Ad Index

Overnight, Texas Expands its 2-1-1 VoIP Call Solution to Help Katrina Evacuees

In response to the needs of Hurricane Katrina evacuees, eLoyalty Corporation's (news - alert) clients, the State of Texas Department of Information Resources (DIR) and the Texas Health and Human Services Commission (HHSC), were able to — in a twelve-hour period — expand the capacity of Texas' 2-1-1 hotline to accommodate an unprecedented level of calls to the hotline. Similar expansions typically require a full month to complete. This 2-1-1 hotline assists the evacuees in getting information on shelter locations, hospitals, Red Cross locations, supplies, and humanitarian needs.

Recognizing the magnitude of the destruction caused by Hurricane Katrina, the first priority of HHSC and DIR was to do the right thing for the evacuees. Expanding the state's 2-1-1 call capacity was critical to helping to direct evacuees to shelters and other assistance. The State of Texas reports that the number of calls handled daily by the 2-1-1 operation increased from about 2,500 to more than 10,000 in the first days after the disaster, as hurricane evacuees sought assistance. Since September 1st, over 225,000 calls have been made to Texas 2-1-1.

The increase in call capacity and new call management solution began with DIR's actions to move excess capacity from the Texas Capitol complex telephone system to 2-1-1's VoIP network. Then, DIR, eLoyalty, and the telecommunications partner's technicians working overnight were able to re- route overflow 2-1-1 calls in the Houston area to other information centers around the state.

In Austin, HHSC and eLoyalty set up a new 50-telephone center to take the calls. In a matter of hours starting the afternoon of September 2nd, eLoyalty and HHSC dsigned a network with the state's partners that significantly canged capacity and call routing algorithms, ceated new IVR applications, built intelligent call routing rules, and created new operations reporting.

All of this was done on the Cisco-based VoIP enterprise IPCC solution and all of these critical tasks were accomplished by 5 a.m., September 3rd. Due to the uncertainty of cell phone access from providers outside of Texas, HHSC implemented a toll-free number for disaster information that also utilizes the 2-1-1 network.

"One of the strengths of the state's 2-1-1 system is the capability to adjust capacity, but making that adjustment overnight is nothing short of incredible. We were able to make this happen because of the exceptional communication and coordination we have with HHSC and our vendor partners," said DIR Telecommunications Director Brian Kelly.

http://www.eloyalty.com

Focus on Customer Service Is Reducing Churn in Wireless

By David Sims

Wanna know how to reduce churn? Have your customer service people be nicer when serving your customers. There you go. Frame that, we know it's the first time you've heard it.

Wireless service providers are having some success in reducing customer service churn and complaint rates, but they still have significant challenges ahead, according to a new report from In-Stat: (news - alert) About one in eight wireless customers considered their wireless carrier's customer service "excellent," although the results for individual U.S. carriers ranged from a low of 7% to a high near 21%.

The research firm found several customer service topics that received praise and complaint for each wireless carrier regarding — stop me if you've heard this before — hold times, service policies, first-call problem resolution, and customer service representative attitude.

The In-Stat report, "Wireless Customer Service: Not Over the Hump Yet," surveyed U.S. wireless subscribers and conducted what the firm says were "extensive interviews with key wireless industry insiders."

"Customer service efforts appear to be contributing both to churn rate reduction, a key factor in company financial performance, and in the reduction of the relative complaint rate," said David Chamberlain, In-Stat analyst. "Nevertheless, a substantial number of subscribers are, or could soon become, defectors."

The report found that in most cases, satisfaction with customer service nearly mirrors subscribers' overall attitude toward the carrier. In other words, efficient handling of billing problems is now considered one of the best parts of the customer service experience, and lengthy hold times and one-call resolution are the worst.

It also found customers are indifferent about Web-based or voice-response self-care initiatives, and what industry insiders have long known, that both the newest and longest-term customers are the least likely to churn.

However, among those in the middle — customers who have been with the carrier from one to two years — more than 25% indicate they probably or definitely will switch to another carrier.

http://www.in-stat.com

GN Netcom and Westcon Group Sign Distribution Agreement

Westcon Group, Inc. (news - alert) has signed an agreement to distribute GN Netcom corded and wireless headsets throughout Europe. GN Netcom is the world's largest manufacturer of headsets and provides a variety of solutions for VoIP-enabled SMB and enterprise networks.

By including the award-winning line of GN Netcom (news - alert) headsets into an overall value proposition, resellers gain the ability to add a relatively simple incremental revenue opportunity to a convergence solution. GN Netcom headsets are fully compatible with telephone handsets from Westcon's telephony systems vendor partners, including 3Com, Avaya, Nortel, Mitel, and Siemens.

In addition to the added convenience and productivity benefits headsets provide, recent research by national government agencies across Europe suggest that jobs necessitating high telephone usage remain areas of health concern in an increasingly telephony-based, service-oriented workforce.

Guy Koster, Director of Product Management, Westcon Europe, commented: "The addition of GN Netcom to the Westcon convergence portfolio represents an exciting opportunity for our reseller and solution provider partners. With headset penetration in call center environments generally accepted to be at saturation point, now is a great time to be looking at the general office environment as a new market opportunity for these products."

"By partnering with Westcon, we can play an important addition to its VoicePoint program, which has proven to be a very significant resource to reseller partners," added Hans Henrik Lund, President, GN Netcom. "Westcon clearly recognizes the significance of adding headsets to an overall VoIP solution and we are thrilled to have our products sold alongside Westcon's core vendor offerings."

http://www.gnnetcom.com
http://www.westcongroup.com

Cohere Picks BroadSoft to Offer VoIP Services

By Johanne Torres, TMCnet VoIP Minute Watch Columnist

VoIP application software provider BroadSoft Inc. (news - alert) was tapped by IP and Internet systems provider Cohere Communications (news - alert) of New York City to launch Broad Exchange Services (BxS), a VoIP service for the multi-site small and medium enterprise market. Cohere chose the BroadWorks VoIP application platform to launch the new VoIP offering. Cohere also expanded its BxS service offerings beyond the northeastern region of the United States to deliver services nationwide.

"The VoIP services Cohere delivers meet the needs of small and medium enterprises, especially those with multiple office locations," said Steven Francesco, CEO, Cohere Communications. "The BroadSoft VoIP application software lets us deliver the services our SME customers want, while allowing us to expand our portfolio to also offer wholesale services. When it was time to bring a platform in-house, the BroadWorks platform was the obvious choice for all our VoIP service offerings." Cohere Communications and partner, Globalive, plan to migrate all VoIP customers to the BroadWorks platform from a previously deployed platform.

BroadSoft's IMS-compliant BroadWorks VoIP application platform provides a set of applications, including hosted PBX, IP Centrex, mobile PBX, business trunking and residential broadband services integrated into a single VoIP application platform.

"Our customers' success is the number one priority for us," said BroadSoft's chief executive officer Michael Tessler. "Cohere Communications has grown carefully into a standalone VoIP service provider, increasing both the depth and breadth of services they provide to SME and wholesale customers across the United States. Their focus on support and security, coupled with the processes they've put into place, allows Cohere Communications to deliver the highest quality VoIP services to their customer base."

http://www.coherecomm.com
http://www.broadsoft.com

Greenwoods to Sell ShoreTel VoIP in the UK

Greenwoods Communications, (news - alert) the UK's leading independent telecoms services company, has been appointed channel partner for ShoreTel's (news - alert) award-winning VoIP product suite. Greenwoods Communications will be aiming the ShoreTel VoIP telephone system primarily at enterprise customers in the UK where the company can offer a complete supply, installation and support package.

"Greenwoods Communications sees the ShoreTel appointment as a valuable foothold in the SME market where our natural and well-proven strengths in the telecoms field can be used to enhance our product offerings and to provide a springboard for our operations and maintenance services," comments Peter Skinner, Commercial Director, Greenwoods Communications. "We will also be offering the solution to European multi-nationals where the proven benefits of VoIP can significantly enhance productivity and reduce communications costs."

ShoreTel's channel program provides extensive modular and technical training, support and rewards to ensure the success of the channel partnership. ShoreTel will also assist with the development of Greenwoods Communications' sales and marketing strategies to meet agreed, predetermined revenue and customer satisfaction forecasts.

"Greenwoods Communications is a large and credible organization that is well known and has a substantial presence in the market. The company has a strong desire to strengthen its position by offering advanced VoIP solutions for the enterprise market," says John Jarvis, MD, ShoreTel EMEA. "The agreement enables Greenwoods Communications to sell our solution nationally, but importantly for ShoreTel, the agreement gives us a strong presence in the Midlands region.

http://www.greenwoodscomms.com
http://www.shoretel.com

Envox Worldwide Broadens Global Distribution For Its Product Portfolio

Envox Worldwide, (news - alert) a leading global provider of voice solutions, announced that is has entered into new sales and support agreements with leading value-added distributors (VADs) throughout the world. The agreements give these VADs the ability to resell the entire Envox Worldwide product portfolio, which includes: the Envox 6 Communications Development Platform, Envox 6 VoiceXML Studio, Envox CT Connect™, and Envox CT ADE™. Under the terms of the agreements, each of these VADs will also have access to Envox Worldwide's value-added services, including professional services, training, and technical support.

"We are pleased to be partnering with this distinguished group of VADs," said Mark D. Flanagan, president and CEO of Envox Worldwide. "These distribution partnerships are a key element of our global growth strategy. We're working closely with each of these partners to build strong customer relationships and broaden our presence in the rapidly expanding voice solutions market."

This expanded distribution network will enable Envox Worldwide to more efficiently address the growing demand for its products and related services, better serve its customers at a local level, and expand its presence as the fast-growing voice solutions market rapidly shifts to open, standards-based solutions. Improved global access to the Company's products and services will enable more developers to create the wide range of voice solutions that their customers need with unmatched flexibility and broad support for new standards and emerging technologies, such as speech recognition and VoiceXML, VoIP and HMP and Web services.

Each of these VADs will now be representing the entire Envox Worldwide product portfolio. Envox Worldwide has signed sales and support agreements 25 VADs Worldwide — 5 in North America, 11 in the EMEA region, and 9 in the APAC region.

http://www.envox.com

ZTE Exclusively Awarded IPTV Project by China Telecom, Jiangsu Branch

ZTE Corporation, (news - alert) the fastest growing global provider of telecommunications equipment and network solutions, has won a 100,000 subscriber IPTV contract from the Jiangsu Branch of China Telecom, China's largest fixed-line telecommunications operator.

The IPTV project, following ZTE's IPTV contract win from the Shannxi Branch of the same operator in September this year, is also China's largest IPTV network.

China is experiencing surging subscriber demand for new communications and entertainment services and IPTV is seen as the cut-in point and foundation of the future digital home. It is a pivotal service for enabling telecommunications operators to provision subscribers with triple play services — voice, data, and video. Global fixed-line operators are now focusing their attention on deploying IPTV services and providing customized and content-rich services to subscribers.

Jiangsu is one of the most developed provinces in China and one of the most important strategic telecoms markets in the country. The IPTV project with China Telecom's Jiangsu Branch is a key part of China Telecom's IPTV trials, and is only the second tender result formally announced by the company after the Shannxi contract.

ZTE won the IPTV project in competition with most of the major international IPTV system providers. The win includes an agreement that ZTE will be the sole provider of IPTV systems and equipment to China Telecom, Jiangsu Branch.

With its advanced telecommunications equipment technologies, in-depth insight into multimedia telecommunications services as well as strong R&D capabilities, ZTE's NGN-based IPTV system delivers distinct advantages in terms of network planning and service operation. It can help fixed-line operators realize retained profit by constantly and effectively providing abundant customer-made services.

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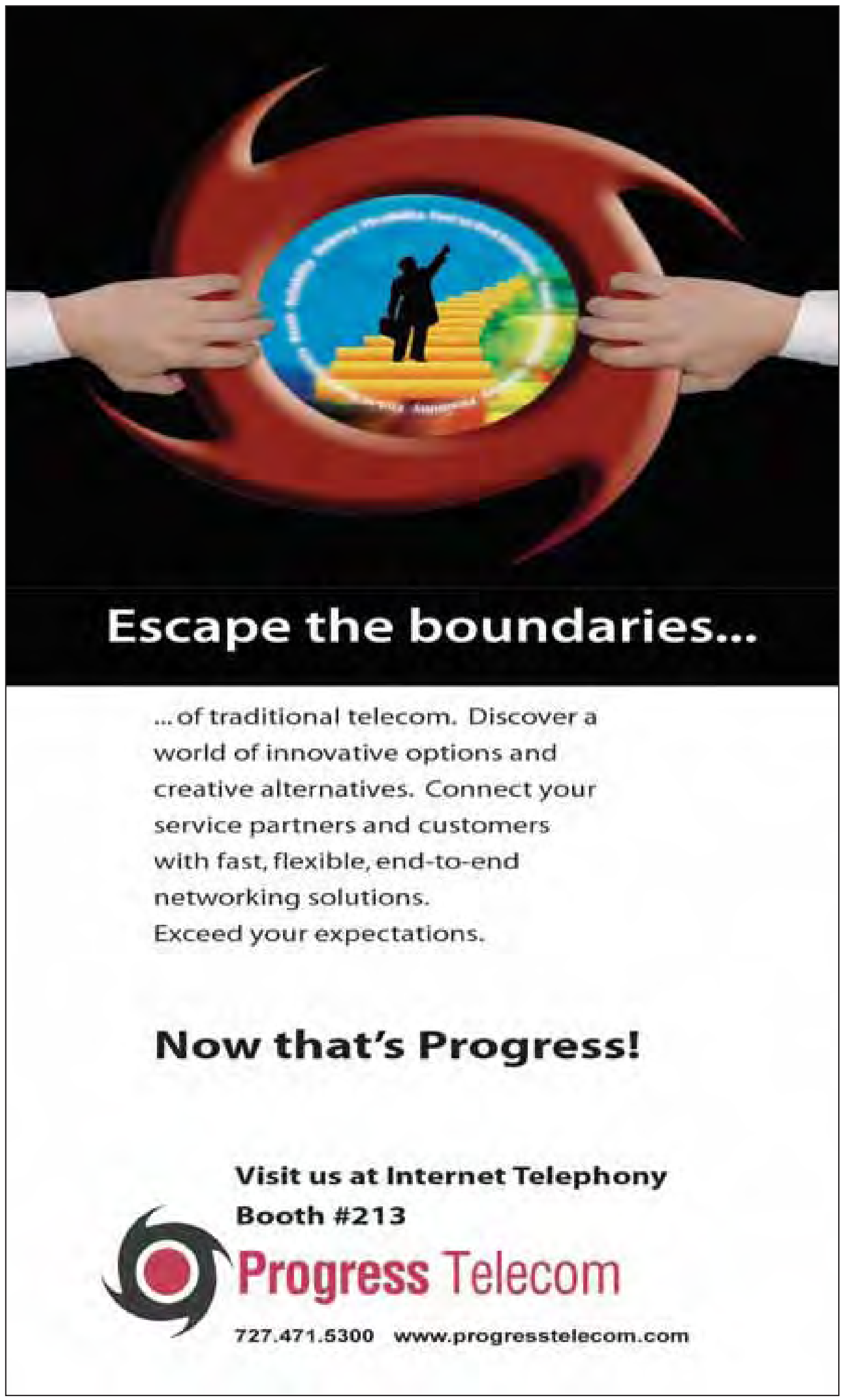

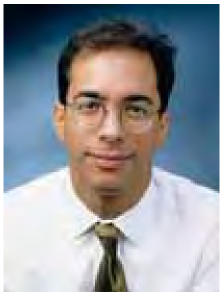

By Marc Robins

All Together Now: The Embedding of Real-Time Communications

One of the most exciting trends today in the IP communications industry is the spread of real-time communications capabilities into every nook and cranny of the global IP-network infrastructure, including the host of software applications that have access to these networks. This trend promises to create tantalizing new opportunities for not only application and software providers but also for VoIP service providers offering vital termination and other important transport, security, billing, and QoS-related services.

In essence, the "separateness" of voice/video communications is coming to an end. Traditionally stand-alone "applications" separated and walled-off from all other data network and computing resources, voice and video communications are now becoming woven into the very fabric of our IP-enabled world. In my previous column, *Welcome to the Golden Age of Web Telephony* (November 2005 issue), I discussed some of the current incarnations of this trend — including the ability to click-to-talk on phone numbers that appear on Web pages and the growing popularity of voice and video as part and parcel of the IM experience. Today, most if not all IM clients support this real-time communications capability.

With the aid of specialized browser toolbars, various application "hooks," and other programming wizardry, voice (and in many cases video) communications is fast becoming just another feature available to users of applications running on broadband-connected PCs, laptops, and other myriad IP-enabled devices. Signs of this integration are also evident in many other industry sectors, including the mobile communications, consumer electronics, and entertainment industries. Anyone who has seen their teenage son become absorbed by one of the new, multiplayer Xbox online games knows that real-time VoIP (define - news - alert) is a key ingredient of that full-blown multimedia experience.

The next phase, I believe, will involve the ability to initiate a voice or video session directly within a host of popular applications, much the same way you can invoke an e-mail communication from within Microsoft Word. Whatever it is you're doing, whether writing a letter or e-mail, piecing together a presentation, developing a spreadsheet, running a database, or most any other computer-based task, you'll have the ability to "reach out and touch someone" with the point and click of your mouse, your stylus, or your voice command. Such integration with everyday software programs promises to create a whole new level of worker productivity and spawn new, feature-rich collaboration capabilities that will help shorten decision making timeframes and boost the overall quality of the workgroup experience.

Anyone who has seen their teenage son become absorbed by one of the new, multiplayer Xbox online games knows that real-time VoIP is a key ingredient of that full-blown multimedia experience.

One area that stands to benefit tremendously is technical and product support. With embedded real-time communications capabilities, software providers (or corporate tech support operations) will be able to field queries directly from users who need help with various features, and provide real-time, step-by-step assistance.

The potential will exist to create new types of customer support services. For example, software vendors could offer specialized Web telephony-based training services, where, for additional fees, users can be guided through a variety of applications tours, or support specialists can be "on call" to help deal with hair-pulling software issues during late-night marathon work sessions.

The continuing, rapid growth of e-commerce is also clearly a big driver of Web telephony adoption. While current Web telephony models either require callers to be paying subscribers to make Web calls (such as with Teleo) or be part of a network of subscribers in order to make free calls to other members (such as with Skype or Vonage), opportunities still exist for providers to fill service voids. One such void is the Web telephony equivalent of the "toll free customer service line." While some pioneering e-commerce companies have text and voice chat set up to help online surfers find the right items, hundreds of thousands (millions?) of other Web sites that sell things don't yet offer online shoppers the ability to connect to a live person in real time. I think the time is ripe for such an offering. IT

Marc is Chief Evangelism Officer of RCG (Robins Consulting Group), a marketing intelligence and communications company dedicated to the needs of the IP communications industry. Marc has been involved in the telecommunications industry as a reporter and analyst, trade show producer and publisher, and marketing executive and consultant for more than 25 years. Contact RCG at 718-548-7245 or e-mail info@robinsconsult.com for more information.

Go To Table of Contents | Go To Ad Index

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FAST FACT

35 Years of Focused Commitment in Business Communications.

By Tony Rybczynski and Phil Edholm

Architectural Discontinuities That Will Transform Your Enterprise

Budgets are flat, while overall complexity, traffic, applications, and security threats keep growing. Fortunately, there are a number of fundamental technology-enabled transformations that will lead to IT simplification and elimination of time, space, and quality of experience barriers to lower TCO, stronger customer engagement, and increased employee productivity — as did digital communications in the ’70s, IP/Ethernet networking in the ’80s, and browsers and ubiquitous wireless in the ’90s.

Converged networking forms the foundation for these transformations and is evolving to autonomic networks. Autonomic networks are real-time secure multimedia networks that can dynamically adjust to changing conditions to optimize network performance and user quality of experience. They allow people to specify what they want in application and business terms, rather than in terms of technology-defined parameters. These networks proactively discover, configure, monitor, and act to bring convergence with simplicity to end users and operators. They heal, secure, tune, and optimize the networking environment, which is continuously adjusted with a closed-loop feedback control system. A wide range of technologies will make this happen, including discovery/auto-provisioning, admission controls, flow-based intelligence, and closed-loop policy-driven layered defense.

For IT, autonomous networking reduces TCO and finger problems that impact security and reliability. For end users, autonomic networking brings plug-and-play to secure application-aware networks, in contrast with traditional best effort, non-secured plug-and-play. For business, it increases agility in moving towards an “everything-networked” world with all its benefits and in leveraging the three transformational discontinuities discussed below.

Work is something you do, not somewhere you go — customer service should know no boundaries.

The Hyper-Interactive World

Hyper-interactivity leverages real-time converged communications to fundamentally transform how users collaborate and how enterprises engage their customers. Unified communications and SIP will do for real-time interhuman multimedia collaboration what the Web and HTML did for information access.

SIP is a control protocol that initiates, modifies, and terminates communication sessions with one or more participants across a broad range of media. SIP (define - news - alert) technologies form the basis of the industry’s IP-centric Unified Communications architecture, which brings together telephony, video, conferencing, application sharing, presence, instant messaging, and personal agents. The latter allows you to be known by name rather than a bank of phone numbers and addresses. It also allows you to control who can reach you, and how and when. The network does the rest.

Virtualization and Elimination of Distance

Virtualization disassociates control and signaling, as well as users and the devices they use. It also eliminates physical barriers to delivering applications that support business processes. Virtualization impacts many aspects of the enterprise by eliminating geographic boundaries across the employee, customer, and supplier base.

For example, mobility allows users to communicate wherever they are, however they are connected, and with whatever device they may be using. After all, work is something you do, not somewhere you go — customer service should know no boundaries. Wireless technologies will deliver ubiquitous broadband access through wireless mesh networks, mobile WiMax, and multi-Mbps public services. Mobile SIP VPN clients, dual mode devices (WiFi & public wireless), and location-based services are key enablers of virtualization.

Role-based identity attaches users’ rights to the roles they have within organizations, while federations address the rising need to dynamically establish trusted relationships with partners, suppliers, and customers. Together they enhance business operation and enable new opportunities. For example, business processes are often linked to phone numbers. As carriers move from device-level identity (phone number) to federated user-level (name) identity, a trusted and unique subscriber name will be delivered to transform this process to be customer-centric.

Webification and Application Convergence

Webification is the communication enablement of business applications (e.g., ERP, document handling, work flow) through Web services, open APIs and SIP. This will result in applications that take real-time collaboration to new heights, bringing users and applications closer together into a new inter-human web. For example, a supply chain management application that detects a change to a critical supply metric could initiate a collaborative session and deliver relevant data to stakeholders, hastening decision making. When combined with sensor networking (including location and RFID), the result will be new environmentally-aware networking and applications.

Already, agent-assisted and self-serve multimedia contact centers are on the verge of dramatically changing the customer experience. These will deliver engaged applications that are media adaptive, time critical and anticipatory, and will take customer service to the next level.

Agent-assisted and self-serve multimedia contact centers will deliver engaged applications that are media adaptive, time critical and anticipatory, and will take customer service to the next level.

Managing IT Does Matter

These architectural discontinuities are highly inter-related. They also provide a tremendous opportunity for enterprises to take advantage for business advantage. The end game is the real-time virtual enterprise, lowering the time to market, time to decision, time to revenue, time to beat the competition, and ultimately time saved for the customer.

Tony Rybczynski is Director of Strategic Enterprise Technologies at Nortel. (quote - news - alert)*He has over 30 years experience in the application of packet network technology. Phil Edholm is the Nortel CTO & VP Network Architecture - Enterprise and is responsible for vision and architectural directions.*

IT

By Hunter Newby

Event Horizon

From Summit to Summit, VoIP Peering has peaked — interest in the technology, that is. VoIP Peering still has a long way to go, but it has certainly begun to sink in with many in the industry. After being the keynote speaker for the State of VoIP Peering during the VoIP Peering Summit at the Internet Telephony Conference & EXPO in Miami back in February, it became clear to me that many looked on with a passive sense of acknowledgment, but didn't feel threatened in any way as that was the first time they had heard anything about such a concept. Besides that, most believed that VoIP Peering was only a technical benefit, improving provisioning methods. They either didn't hear, or they misunderstood the part about ENUM. Those that did hear and clearly understood, namely the VoBB and smaller cable companies with flat-rate VoIP offerings, jumped right on it and have been pursuing and implementing it ever since.

By the time we got to Los Angeles, ENUM VoIP peering was a half-billion minutes strong and everywhere was talk of implementations! The VoIP (define - news - alert) Peering summit at the LA Internet Telephony Conference & EXPO in October was a big step forward. Now we had some real debate from the attendees in the form of nay-sayer questions. There were even a few new service offerings announced at the conference. It was clear that the potential economic impact of multi-lateral VoIP Peering was being understood. The questions were good, challenges overcome, and objections noted, but overall it was a testament to how far the concept had come in less than a year. Any time there are people serious enough to try and discount or discredit reality, you know you're on to something. More providers trying to get in to the space is also a good indication.

As all types of business models around the world change those of us in telecom need to realize that we are not immune. Actually, we're far from it. In the past there have been radical shifts and new businesses born from them. Ultimately, all of those changes are what led us to where we are today as a global society. To us it's all good, as we don't remember first-hand the pain and suffering the now infamous buggy-whip makers went through while their business model (and entire industry) was wiped off the face of the earth. There's no need for many of us to try and imagine what it must have felt like as anyone in the circuit-switched minutes business is feeling the same way right about now. If you're not one of those people, you're probably only one degree of separation away from someone who is. TDM switches are being replaced with softswitches, or at least upgraded to IP front ends until the eventual phase-out. There is a new skill set to acquire, a lot of work to do, all for lower margin services. It's a lot of effort and many people in the path of this evolutionary change are questioning the potential returns.

Any time there are people serious enough to try and discount or discredit reality, you know you're on to something.

Just as the event horizon of the black hole is met and, from that point on, all matter is crushed in to singularity, margins on minutes and the "minutes" themselves have now met the VoIP Peering Event Horizon. If nothing else, bi-laterals become easier to provision and the whole market moves faster, with shorter contracts and more price compression. That's the bright side for the existing model. The multi-lateral VoIP Peering that is occurring today in ever-growing daily volumes is whisking PSTN minutes away in to the ether, never to return again. The minutes universe is getting smaller by the second. If you haven't done so already, it's time to evaluate your voice service business model and form an escape plan.

There is a sunny side of the street. With the help of the visionary, Rich Tehrani, VoIP Peering is gearing up for a big year in 2006. Rich has independently declared 2006 as the Year of VoIP Peering. This declaration is, no doubt, based on its successes in 2005, but it is probably much more than just a guess. Rich, President of TMC, has been in this business of VoIP and Internet telephony for many years. He's seen all that has come before, what worked, what didn't, and why. VoIP peering is a better provisioning methodology as well as the key to a definable ROI and bottom line margin improvement for carriers and enterprises alike. VoIP Peering makes common sense and a lot of sense to Rich. Thanks for the vote of confidence! See you at the next VoIP Peering Summit in 2006! IT

Hunter Newby is chief strategy officer at telx. For more information, please visit www.telx.com. (news - alert)

By Max Schroeder

Communications Continuity Planning

Late in September and in the wake of the recent hurricane disasters on the United States Gulf Coast, and the well-documented communications failures in New York City on September 11, 2001, I contacted Rich Tehrani, to discuss how leading industry practitioners could contribute in an effort to avoid such problems in future disasters. It was decided that TMC and the Enterprise Communications Association (ECA) would host a press conference and planning meeting at the Los Angeles INTERNET TELEPHONY Conference & Expo Fall 2005 to determine industry interest in creating a forum to discuss solutions for ensuring communications in the wake of a disaster.

In a pre-show announcement, Rich Tehrani, a foremost expert on VoIP, commented, "I have mentioned before that I am proud to be involved in the VoIP industry as we are giving back to the world. We are helping the underprivileged and we are allowing those people who could not communicate before to now speak with one another."

Certainly the activity at the show validated Rich's observation. As soon as the conference opened, it soon became obvious that the vendors were not only interested in supporting this initiative, but many were already engaged. Despite the last minute announcement, some of the exhibitors arrived equipped with products (hardware, software, mobile units) specifically designed to address the disaster preparedness marketplace including Sphere Comunications, Inveneo and Telephony@Work.

Sphere Communications

(news - alert) Sphericall VoIP technology offers a scalable, portable communication infrastructure with multiple access gateways to the PSTN (define - news - alert) via VPN, T1/E1 and satellite. The combination of terrestrial and space-based interfaces provide for both traditional communications continuity using backup sites and mobile continuity with mobile satellite units.

Inveneo

Inveneo (news - alert) is a non-profit entity that has developed and deployed a Linux VoIP solution powered by a combination of solar and bicycle power. The original concept was to employ the units to connect remote villages using VoIP and wireless technology. A perfect fit for third-world countries. However, when Katrina hit the Gulf Coast, virtually the entire company packed up and moved to areas ravaged by Katrina. Working in combination with the Army, who provided the company with generators (no pedaling required), access to a T3 plus climbing gear enabling them to climb poles to mount their wireless devices, Inveneo helped set up communications for emergency workers and other organizations. Their solution, as exhibited in L.A., was compact and utilized recycled materials to keep the cost low enough for deployment in the third world.

Telephony@Work

Certainly contact centers are critical to many organizations during a crisis. Products from another exhibitor, Telephony@Work, (news - alert) specifically address this market. I visited one of their client sites about two years back to attend a seminar on Disaster Preparedness. Their client was the Municipal Credit Union (MCU) of New York and the location was 22 Cortlandt Street New York, NY. The building also fronted Church Street directly across from the World Trade Center site. Looking down on the WTC site from the eighth floor of the MCU offices definitely put disaster preparedness in perspective. It was very evident why the MCU had established a mirrored site in an adjacent state. The second site ensures that their contact center design meets the 3 key stages of a major disaster plan — preparedness, continuity of operations, and recovery.

We are helping the underprivileged and we are allowing those people who could not communicate before to now speak with one another.

Disaster Planning Communications Forum

Recent disasters in the USA have focused attention on the issue of government preparedness and almost totally ignored

Go To Table of Contents | Go To Ad Index

the enterprise. Yet, keeping the enterprise running is a critical factor in bringing a community or geographic area back to normalcy. The Disaster Planning Communications Forum (DPCF) is in the early stages of development and is looking for members to contribute time to launching and defining the organizational structure. One key suggestion presented thus far is that the organization becomes a valued resource for enterprises and government agencies when seeking information on how to prepare for a disaster. A second suggestion is that the forum should develop a resource guide outlining how to construct a disaster plan. The guide would include case studies and a listing of companies and products that address this critical market.

Keeping the enterprise running is a critical factor in bringing a community or geographic area back to normalcy.

The above examples were selected because they cover a broad spectrum of disaster planning. Combined, they address the needs of everyone, from the third world to the largest and most sophisticated cities in the world. Many more exhibitors displayed solutions that can be incorporated in disaster planning. Perhaps your company's products and experience also fit into that equation? If so, you should participate in the Disaster Planning Communications Forum (http://www.tmcnet.com/DPCF). Take the time to visit the site now. Send your commitment to participate or any questions to: maxschroeder@tmcnet.com. IT

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

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By Jack Jachner & Chris Vuillaume

User-Centric Seamless Mobility

Mobility is a killer app in enterprise communications — enabling both voice and data communications for conducting business when away from the office. Mobile phones provide anytime anywhere telephony for mobile workers. IP provides low-cost voice access (VoIP) (define - news - alert) from branch offices and for teleworkers. VPNs provide secure remote data access.

To date, these are separate solutions, with dedicated terminals and applications: cell phones, IP PBXs, and laptops with VPN (define - news - alert) software and tokens. There is no continuity of service between mobile and office telephony, or for data access. Nor is there a continuity of user experience, which is fragmented among different devices and applications. We are now on the verge of a transition toward User-Centric mobility that provides a seamless service and seamless user experience wherever the user is located, or transitions during the working day: home, office, airport, convention center, and so on.

Rethinking Mobile Communication

The driving force for the new Mobility is the ubiquity of IP service. IP is already used to connect branch offices and remote workers with headquarters, initially for simple toll bypass and then to allow access to HQ applications (including IP Telephony) in a uniform way across all locations of an enterprise. IP is now going wireless, with WiFi and IP enabled cellular. The new Mobility will extend enterprise communication service beyond of the enterprise borders (home, airport, etc…). Employees are always connected and with uniform enterprise communication services: calling your colleague by name, by speech recognition or by four- or five-digit dialing, accessing corporate unified mail, or conferencing. Ultimately all the enterprise IT applications will extend outside of the enterprise, to fully support employee mobility. The new IP mobility supports continuity of user experience, services, and applications, and its personalization to person to person communication regardless of their location.

How can you start your conference call at home using your PC's softphone and then continue the same conference on your mobile device?

Emergence of Mobile Convergence

Enterprise-class Voice over Wireless LAN (VoWLAN) solutions are emerging, as evidenced by the rapid growth and maturity of WiFi infrastructure solutions, by the emergence of dual-mode (WiFi and cellular) handsets combined with the development in softphones based on the Session Initiation Protocol (SIP). While Voice over WiFi solutions have been in use in some industry verticals (such as warehousing), the technology has not reached readiness for the general enterprise market. Mobile and WiFi markets are recently converging and evolving to provide a seamless wireless network for voice and data communication.

This mobile convergence is interesting as it can bring reduced call costs, improved reachability, and increased productivity for the enterprise user with the associated reduced churn and increased ARPU for the Service Providers.

A number of requirements must be met for mobile convergence to be successful in the market. The solution should be open enough to support a wide range of wireless access, user terminals, and network connectivity options, and may leverage SIP as the enabling protocol. The user experience must be simple and seamless for user adoption. It must exploit the benefits of wireline access (such as lower communication cost) and deliver service level mobility demanded by business users. For faster adoption, the service must be capable of coexisting and collaborating with other popular business applications like VPN, XML, http, IM, conferencing. The solution should have a low startup cost and "pay-as-you-grow" pricing model for growth. This solution has to be remotely manageable (software updates, remote troubleshooting, remote license keying…). Finally, the solution must be able to support new network-hosted applications provided by MVNO and customer premise-based systems and applications.

Quest for Smooth Handovers

The typical mobile worker's day is interesting, and so are the requirements for his mobile access. He may be in the office with its VoWLAN phone, where he has to be able to hop from access point to access point maintaining quality and communication services. The problem is more complex when the user leaves the office and would like to keep the communication services. This is called *Communication Handover*.

Mobile users typically use many devices during the day: deskphone, PC, Laptop, home phone, mobile phone. The multiplicity of devices poses challenges. How can you start your conference call at home using your PC's softphone and then continue the same conference on your mobile device? This is called *Device Handover*.

In real life, mobile users cross multiple networks, including

Go To Table of Contents | Go To Ad Index

WiFi hotspots, and emerging cellular networks using IMS, GSM/GPRS, hopping from service provider to service provider. How do you maintain the communications services roaming from one network to another? This is called *Network Roaming*.

Promising Solutions

To satisfy all the requirements, the new mobility solution will necessitate evolution in both the network infrastructure and in communication devices. The mobility services can be delivered from the network with the existing desktop terminals, but will benefit from new converged wireless terminals.

There is a voice-data convergence in wireless terminals, with personal digital assistants (PDAs) becoming telephony enabled and smart mobile phones able to access data. Both are becoming open to run Enterprise-grade telephony applications, benefiting from a colored graphical user interface. Internet access, Office applications, personal information managers, directory accesses, unified communications, and mobility are no longer dedicated to laptops and desktops.

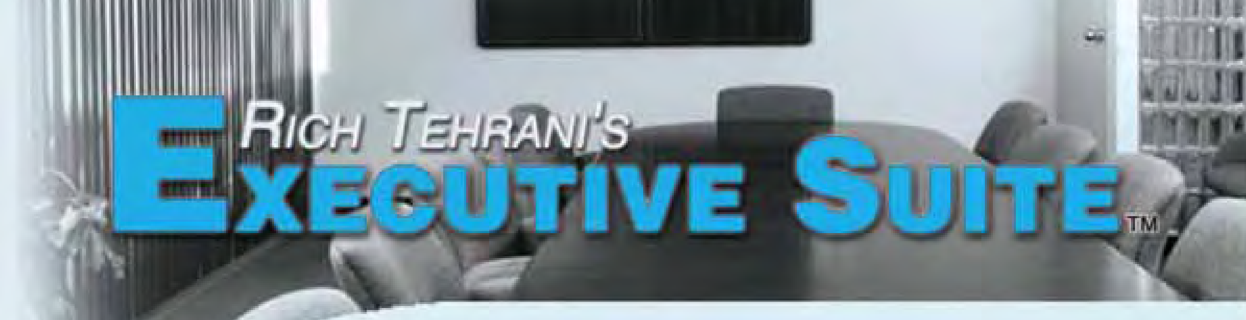
Using WiFi or data-enabled mobile networks, these new devices may provide the same level of features as deskphones (call server independent, incoming and outgoing calls, call transfer, unified messaging, mute button, virtual keyboard for call-by-name feature, message service, unified messaging, personal assistant, collaboration and presence, access to local address book, and access to the call log).

By using WiFi Networks (in the Enterprise LAN or VPN in WiFi hotspot) to connect to company's infrastructure, these devices also save money for the enterprise by leveraging the enterprise communication system.

For the benefit of the user, the same device is becoming "your single device" with single number, single unified communication, while keeping the same level of enterprise communication services (as you had in the enterprise) and providing seamless transitions between the private WLAN domain and the public mobile domain.

The new emerging mobility solution is lowering telecoms cost with a higher predictability, simplifying deployment, and management of the mobile worker toolbox, increasing end user productivity and satisfaction, and allowing flexible and scalable (up and down) infrastructure deployment. For the user, the benefit is even more obvious: a consistent interface and continuous service wherever business communication needs to be done. IT

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



Toshiba's Larry Meyer

In this month's edition of his Executive Suite, TMC's (news - alert) Group Publisher and Editor-in-Chief interviewed Toshiba's Vice President of Sales & Marketing for the Digital Solutions Division of Toshiba American Information Systems (news - alert) Larry Meyer.

Toshiba is a leading provider of electronic and communications equipment to both enterprises as well as consumers. Toshiba's IP telephony solutions for small and medium-size businesses converges PBX and VoIP systems, making it possible to use a private IP and Internet connection for both voice and data communications. Today, Toshiba offers a comprehensive feature-rich suite of communication solutions for the Small and Medium Enterprise market.

Larry took the time to speak to Rich about the state of the VoIP (define - news - alert) industry today, its general acceptance among key corporate decision makers, and how he expects its deployment to evolve in the years ahead.

Where do you think we are as an industry in terms of VoIP acceptance?

In our experience, the users are still finding their way. As an industry, we've done a good job of convincing them that they need VoIP, but the vast majority of our customers (SMBs, government and national accounts) tell us they aren't ready to go to an all-IP telephone system yet, but they do want either a converged system or, at the very least, one that is VoIP-ready.

Who do you think are the decision makers for IP telephony?

We find there are two key types of decision makers: IT managers and executives who want VoIP because they understand IP and look at voice as just another application; and management executives and company owners, who, in most cases, are not sure of all the benefits, but they know they want a system that is VoIP-ready. Of course, there are some in both groups who do understand VoIP, but even then, most are choosing a converged system. Using a converged system allows them to use VoIP only where it really makes sense. In a recent survey we did of our dealers, they told us that, on average, 36 percent of the systems they've sold so far in 2005 are converged, versus only 7 percent IP-only. The remaining 57 percent are TDM, but they most are VoIP-ready systems.

What do you see as the key trends in IP market acceptance among SMB enterprises and regional offices/retailers?

Most SMBs and small regional offices and retailers are still keeping their toes in TDM technology by choosing converged systems. Driving IP market acceptance are the need for a new telephone system, or upgrade from existing system; the need to connect remote workers without installing multiple systems; and, to a lesser extent, the desire to move to the VoIP platform. This market segment does not like to make a big change unless there is a clear motivator — such as cost savings, productivity improvements, or the ability to do something they could not do before. However, they are reading about VoIP and hearing about it from others, so it does have a presence in their minds as they do make communications decisions going forward. One thing is for sure, when they are buying new communications systems, nobody wants to get left behind. Nearly every system we sell today is either converged or IP-ready.

How about government users? Are they keeping pace with enterprises?

Yes, many government agencies have already implemented VoIP, announced plans to implement, or are currently researching, especially at the local and state levels. Additionally, many federal agencies have plans to implement in the coming year. As in enterprise, we expect that it will continue to increase over the next three- to five-year period.

How important is it to offer a converged solution?

As I mentioned, our dealers tell us, on average, 36 percent of the systems they've sold so far in 2005 are converged. They do expect that in three to

Go To Table of Contents | Go To Ad Index

five years, they will be selling more and more all-IP systems. But for now, offering a converged system is very important. There are just too many customers out there who simply don't care what the delivery method is, as long as their telephones work, but they do want to be IP-ready so they don't miss out on the future.

What do you feel are the major drivers for IP telephony today?

Remote workers and mobility are clearly the key drivers for IP telephony. VoIP gives remote workers and offices a cost-effective, productivity-enhancing way for people to stay connected to the corporate communications system. They can plug in their IP telephone (or IP softphone) virtually anywhere and have access to their calls, voice mail, extension dialing, and other features, as if they were at the main location. Additionally, VoIP gives them integration with more applications, for example IP softphones on PDAs and the ability to browse or access the Web from a variety of end-points. The integration with applications enables the end user to customize communications and enhance productivity.

How about cost savings? Is that an issue among Toshiba's customers?

Conversely, cost savings does not seem to be the big driver that it was expected to be, given the low cost of long distance. Today, our customers are looking at improvements in productivity and being able to do things they could not do before — such as taking their desk phone with them and plugging it into virtually any Internet access point. They want be able to do more; that's the real motivator, not cost savings.

What are drivers that may surprise some of us?

From Toshiba's perspective, given our large installed base, migration is a huge driver, where migration is possible. Toshiba has had a commitment to "never leave any customers behind" in its migration path, so even with our newest Strata CIX IP business communications system, our users can migrate from the previous platform to create a converged or all-IP communication system. They protect much of their initial investment by being able to keep much of their original equipment, in particular if they choose a converged system. In this case, cost savings related to migration can be a driver because users can move to the technology of the future with less initial cash outlay.

There are a lot of vendors who disagree and feel that a forklift to VOIP is the solution. How important is migration in the transition from TDM to VoIP?

For our customer base, migration is essential. For nearly 30 years, we've promised a migration path, and we've delivered on it. We have a "customer for life" mentality, and it shows in that a good chunk of our customers have stayed with us from the old days of analog, to digital systems, and now onto IP-based systems. They trust us to help them preserve their investment and move them to the next technology platform as smoothly as possible, and at a pace of their choosing.

What does Toshiba do specifically to help its dealer network transition to VoIP?

Toshiba has made every effort to help its dealers move to IP telephony. For example, we have given them a one-stop shop to deliver everything the customer needs to buy and install an IP telephony system (converged or all-IP), including in-house support and partnerships with SMC for networking capabilities, SonicWall for security solutions, and Peak Technology for network assessments. The partners' products have been tested and proven and, in fact, are being used at Toshiba. The partners also have been trained on Toshiba systems. Of course, dealers can use whichever vendors they'd like or, in many cases, they can deliver some of those services themselves. We also have one of the most extensive training programs in the industry, including online and interactive classes, sales and technical training classes and certifications to teach IP implementation.

Where do you see market acceptance going over the next three to five years?

We expect converged systems to be the systems of choice for the majority of our customers in the short term, but we believe that the shift to VoIP is accelerating and will dominate the market in a few years.

What do you think is important to accommodating the telecom users today as they transition to VoIP? What can we do as an industry?

Education is the key — in magazines, Web sites, white papers, conferences like IT EXPO, and through one-on-one contact, be it through salespeople or others. From there, the channel partners need to help their customers understand that voice is not just another application on the network and the network has to be primed for bandwidth and QoS capabilities before voice can really work. There has to be a lot more hands-on delivery of the system and long-term relationships between the dealer and customer. Additionally, the VoIP technology needs to be transparent, so end users do not experience a negative impact to their business. Ease of use and Quality of Service must be inherent in the VoIP system.

At Toshiba, customers can count on a smooth migration path, as they have for nearly 30 years. IT

Go To Table of Contents | Go To Ad Index

IP Phone Roundup

With all the choices and options for VoIP users, there is one thing all Internet telephony users must have: an endpoint. As with any other piece of hardware, there is a surfeit of IP phone options — some are desktop units, others are wireless options, and still others are conference phones, not to mention USB phones and softphones. Different user groups, indeed, different individual users, have varying needs and, therefore, require different endpoints.

In this month's product round-up, we have selected a sample of the many firms currently offering desktop IP phones (though many also offer alternative options as well). What we have *not* done is list all the individual features of each phone — that would have taken a year's worth of pages. It would also have been overly repetitive, since many of the features from manufacturer to manufacturer and model to model are the same standard offerings — things like muting, speed-dialing, intercom, conferencing, call appearances, line appearances, group status, user-programmable buttons, and speakerphone.

Instead, we have made a basic assumption that the vast majority of these firms offer a phone or phone line, if you will, that, at the very least, offers most of the basic features and functionality. Within these pages, we have opted to relate what each manufacturer believes sets their IP phones apart.

Naturally, because we are physically constrained within the pages of the magazine, we are aware that there are manufacturers and providers that did not make it onto these pages. Once this listing makes in to TMCnet.com, we will add a listing of those firms.

3Com
www.3com.com

3Com's (quote - news - alert) broad range of IP phones extend productivity-enhancing benefits to users wherever and whenever they require IP telephony connectivity, and give organizations implementation options that can align with their business needs and budget. Advanced features and high-fidelity in 3Com IP phones help enterprises function more productively and meet customer needs more competitively.

3Com® NBX® and VCX™ IP Telephony modules support a portfolio of 3Com business phones and attendant stations, all designed to lower costs, increase user productivity, and strengthen customer interactions.

3Com has a phone to cost-effectively fit virtually every business application — from the 3Com 3100 Entry Phone, that is particularly practical for common areas, to the 3Com 3103 Manager Phone with its large LCD display and Gigabit-speed network connectivity, to the 3Com 3106 and 3107 Cordless Phones that so effectively support worker mobility.

For connection flexibility, 3Com IP phones support two Ethernet ports. One can be used to connect the IP phone to the network and the other to enable an innovative pass-through capability that lets a PC connect to the network through the IP phone. This functionality can reduce total port count requirements and, if the user has a laptop, form part of a company's power recovery strategy when integrated with an industry standard Power over Ethernet LAN switch.

3Com offers five desktop IP phones, an attendant console, and two wireless models.

Aastra Technologies
www.aastra.com

The Model 480i is an advanced, fully featured, IP screen Telephone that provides a flexible IP solution designed with interoperability and ease of use in mindEnhanced Call Management — Large storage for personal directory, callers log, and redial list

The 480i features a large display with and eight-line backlit screen with six context sensitive softkeys to provide more information and flexibility in call handling. It also offers enhanced call management features, tight integration, multiple line/call appearance lights to support up to 9 simultaneous calls —

Go To Table of Contents | Go To Ad Index

a user can juggle multiple calls easily with a single press of a button. Shared calls allow you to place a call on hold at one set and pick it up easily at another set. The incorporated speakerphone provides excellent voice clarity and delivery.

With the 480i, your investment is protected — Firmware upgrades can be downloaded and installed in the field as standards develop and protocols evolve. There is also less wiring involved, as built-in dual 10/100 switched Ethernet ports let you share a connection with your computer. Inline power support eliminates power adapters.

Aastra (news - alert) offers three additional desktop models, as well as a cordless handset and a conference bridge.

ADTech

http://www.adtech.be

ADTech (news - alert) has developed an IP phone with a built-in smart card reader to exploit the full power of VoIP installations, allowing users to be more efficient and productive. ADTech SI 160 is a powerful platform designed to provide a large panel of added value services in addition to IP telephony.

Storing user specific settings on a smart card (phone extension, preferred tones, access policy, etc.) makes a user totally mobile, with the phone being reprogrammed at every card insertion, and the calls being automatically rerouted.

Alcatel

www.alcatel.com

As part of the Alcatel (news - alert) professional line, these state-of-the-art IP phones are full-featured with integrated IP connectivity and telephony, bringing you the converged power of data and voice over IP.

Besides their optimized design, these terminals offer high-resolution, adjustable color or gray screens, wide band audio, superior quality ring tones, the freedom and connectivity of Bluetooth® wireless technology, plus the capability to support any web-based business application.

Alcatel IP touch provides the very best in sound quality thanks to a large range of new advancements, and an XML interface enables you to customize your communications infrastructure to the unique demands of your business. The new terminals offer the full range of telephony services found in Alcatel's acclaimed OmniPCX servers unsurpassed in terms of functionality, features, reliability, and quality of service

Alcatel also offers its entire series of Reflexes IP telephones.

Allworx

www.allworx.com

Allworx (news - alert) VoIP phones save you money by sending calls over the Internet, save you time with easy-to-use design, and travel effortlessly between work, home, and clients.

The 9102 is ideal for those who want the power and cost efficiencies of VoIP but don't have a high call volume. It has a sleek, compact design, supports two lines, and includes two programmable function keys.

The 9112 is perfect for executives, office managers, administrative assistants, remote users, and employees who make and received a high volume of calls. It is designed for people who make and receive high volumes of calls. It provides 12 programmable feature keys, which can accommodate any combination of PBX call appearance, Key system line appearance, speed dialing, and direct station dialing.

Both models feature elegant design and adapt to any business environment — from engineering consulting firms to retail locations. Both also incorporate all the commonly used features needed for day-to-day business, which are accessible directly from the phone, such as voicemail messages, transfer, conference calling, intercom, mute, hold, do not disturb, call waiting, and full-duplex speakerphone.

AltiGen

www.altigen.com

The IP 710 is a fully featured IP telephone designed to empower the user. Bringing stylish form and functionality to the desktop, the IP 710 makes sophisticated features simple and intuitive. Users have single-button access to voicemail, activity/presence selection, voicemail greeting selections, call recording, call conferencing, call transferring, and even placing calls to employees in other countries.

With the 4 line, backlit liquid crystal display the IP 710 is capable of displaying time, Caller ID name and number, real-time call center workgroup statistics, do-not-disturb, and call forwarding status.

User's can personalize the IP 710 with 15 backlit user-defined keys. These programmable keys can be set up for any combination of configurable features like, but not limited to, speed-dialing, extension busy/ringing appearances, call appearances, line appearances, and workgroup activity status.

AltiGen (news - alert)also provides you with 14 combined traditional and melodic ring tones which can also be assigned to line and extension appearances on the programmable keys.

AltiTouch 510 telephones make executive-level features and capabilities affordable enough for every user. The large, easy-to-use display shows caller ID numbers and names, date and time, do-not-disturb status, call forward status and has pre-programmed soft key function keys that provide effortless access to the rich AltiServ system functionality.

The AltiTouch 510 is equipped with a high quality speaker, a large (7.4 square inch) backlit LCD display with adjustable viewing angle, and an LED voice mail waiting indicator. One-touch buttons provide access to features and call control including: volumes, mute, voice mail, voice mail greeting management, do not disturb, intercom, headset answering, conference, hold, transfer, redial, call release, speed dial, call directory, extension management, line pickup, account codes and much more.

Avaya
www.avaya.com

Both standards-based Avaya (quote - news - alert) IP phone series — the 4600 Series and the 5600 series — bring the rich features and functions of Avaya Communication Manager directly to the desktop, while also supporting desktop applications above and beyond telephony. Several models are available, ranging from entry-level IP telephones to those built specifically for demanding contact center environments, to sophisticated color-display screenphones. The phones all are easily configured to support both the H.323 protocol and SIP.

The 4600 and 5600 series both feature models with a built-in high quality, full duplex speakerphone with echo cancellation and a tuned speaker cavity to provide a best in audio quality. Both also provide the same advanced calling features and functions as traditional Avaya telephones. These stylish phones incorporate a number of convenient features, such as a message waiting indicator, easily readable displays, hearing aid compatibility, NetMeeting compatibility, multiple programmable feature keys and display navigation keys.

Avaya's IP phones deliver an extensive set of software features, high audio quality, and attractive streamlined design. Advanced Web-enabled graphical displays on several versions support browser-based desktop applications such as online order entry and inventory lookup in addition to more traditional voice applications.

Avaya's VoIP solutions include more than a dozen desktop units, as well as wireless and softphone options.

Cisco
www.cisco.com

Cisco (quote - news - alert) IP phones provide solid, inviting, simple-to-use, functional, and fully featured next-generation communications devices. Cisco IP phones give customers an exciting new user interface that offers display-based access to features, productivity-enhancing applications, value-added services, and the industry's first Gigabit Ethernet IP Phone. From the company lobby to the desk of the busiest of managers, from the manufacturing floor to the executive suite, at home, on the road or from a branch location, there is a Cisco IP Phone designed to meet every need.

Cisco provides a complete range of next-generation communications devices that take full advantage of the power of your data network, while providing the convenience and ease of use you've come to expect from your business phones. Cisco IP phones enhance productivity and address the needs and capabilities your organization.

High-quality, hands-free speakerphone capability and built-in headset connectivity are included. The large pixel-based display supplies important communications information and ease of feature usage, as well as access to many exciting productivity enhancing applications via XML capabilities. XML-based services can be customized to provide users with access to a diverse array of information such as stock quotes, employee extension numbers, or any Web-

Go To Table of Contents | Go To Ad Index

based content.

Cisco's 7900 series consistes of more than a dozen desktops, wireless, conference, and softphone products.

Grandstream Networks

www.grandstream.com

Grandstream (news - alert) GXP-2000 is a next-generation enterprise IP telephone based on open industry standards. Built on innovative technologies, GXP-2000 features market-leading superb audio quality, rich functionality, and excellent manageability at affordable prices.

GXP-2000 features dual 10/100M switched Ethernet ports, headset jack, adjustable large 130x64 graphic LCD (with support for multiple languages), 7 programmable speed dial keys, 11 line indicators (4 dedicated and 7 shared with speed dial keys), with further expandability to support several dozen additional programmable keys via an add-on keypad box, and much more.

Innovaphone

www.innovaphone.de

Innovaphone's (news - alert) IP 200 is a user-friendly telephone designed for the use in offices. It is the ideal system telephone component for the innovaphone PBX. The ergonomically designed device offers intuitive operation for a great number of added features. This soon gains it favorite status with all those who use the telephone widely

Innovaphone's IP 200 is perfectly suited for home offices. It can be connected directly to a DSL modem. Its implemented PPoE protocol enables it to sign on to the Internet provider. As soon as it is logged on, the IP 200 creates a VPN connection with the company's network, and signs on at the company's telephone system via the established tunnel.

Six function keys allow the user to further optimize his work procedures. Each key supports up to four different definitions, which will be shown in the corresponding display line. A selection of 23 available key functions guarantees truly customized configuration possibilities. The functions range from activating call diversion or bell signal muting to voice announcements or a partner key with busy lamp field for secretarial functions.

Innovaphone IP telephones include the IP10, IP100, IP200, IP202 models, as well as a softphone.

Inter-Tel

www.inter-tel.com

Whether you need a sensible office phone, a menu-driven display phone, or a multimedia touch-screen phone, Inter-Tel's suite of endpoints delivers the flexibility your business needs for on-site employees and remote users to perform their business functions with continuity.

To suit customers' dynamic needs, most Inter-Tel (news - alert) IP endpoints are available in two modes: Inter-Tel Protocol and SIP (Session Initiation Protocol) mode. Inter-Tel Protocol provides an IP endpoint with access to the features and functionality of a traditional Inter-Tel endpoint. Implementing SIP mode allows your enterprise to access the Shared Extension feature, which allows multiple endpoints to use the same extension number on an Inter-Tel advanced communications system. Incoming calls to a shared extension are sent to SIP endpoints simultaneously; offering you dynamic mobility when you're away from your main desk. Additionally, SIP mode enables IP endpoints to interoperate with third-party SIP solutions.

Regardless of your business' needs, Intrer-Tel extends award-winning application performance to its wide range of flexible, intelligent, mobile, and easy-to-use communications endpoints, all of which are designed to suit your unique requirements.

INTER-Tel also offers three wireless IP devices, as well as a softphone.

Linksys

www.linksys.com

The Linksys (news - alert) One PHM1200 IP telephone looks and works like a key system phone, but it will also connect to the Internet and will be very easy and cost-effective to deploy. Linksys One technology is automatic and self-configuring — plug in a Linksys One phone to the Linksys One Services Router switch port and you'll be ready to make and receive calls using business quality VoIP. You'll

also get advanced features and ease of use.

A planned applications programming interface will eventually allow applications to be hosted on the phones either local to the business site or delivered as a component of a hosted service. But there's no need to wait — today's Linksys One IP phones have all useful features of most popular key sets and can save you money, starting now.

The Linksys One PHM1200 IP telephone delivers both traditional telephony services to the small business user such as call forwarding, transfer, busy lamp field and shared line appearances on a next-generation IP phone platform. This platform contains a high-resolution (320 X 240 X 16 bit) color, backlit display; 24 programmable buttons with integrated LEDs, 4 softkeys, 3 fixed function keys and a 5-position navigation key. A full duplex speakerphone, handset and headset port offer superlative audio quality, volume control and muting control. An integrated voice mail application provides cost-effective,high-quality voice messaging service with both local and remote access to voice mail.

Mitel

www.mitel.com

Designed with ergonomics and modern office aesthetics in mind, Mitel's desktop devices give users easy, intuitive access to the feature-rich telephony and advanced desktop applications enabled by Mitel Integrated Communications Platforms.

Mitel (news - alert)continues to offer a wide range of IP phones from affordable entry level to sophisticated IP phones and devices including Wireless Handsets, Conference Units and PC-based Attendant Consoles.

The true promise of VoIP is at the user's desktop, where combining voice, data, and video enables advanced new IP-based multimedia applications. The 5200 IP Desktop portfolio gives users unprecedented levels of personal control over their business communications. The phones and devices feature a context-sensitive softkey/display combination — technology that was pioneered by Mitel almost a decade ago. This combination reduces user uncertainty while supporting efficient call processing, application access and preferences programming by displaying only those prompts appropriate to the function in progress. Users can easily program their personal phone preferences via their Web browser. In fact, one 5200 IP Desktop device — the Mitel 5230 IP Phone — even combines the power of IP Telephony and PDA technology to provide users with unparalleled levels of access, flexibility, and control.

Nortel

www.nortel.com

Nortel (quote - news - alert) IP Phones are capable of supporting the comprehensive telephony features and applications available with Meridian 1, Business Communications Manager, Communication Server 1000, Multimedia Communication Server 5100, as well as Nortel service provider platforms.

Whether you choose a full-featured desktop Internet telephone or prefer the comfort and convenience offered by our software-based Internet telephone, Nortel Networks has the right solution for you. The Internet tele-

phones operate seamlessly across our entire range of IP-enabled platforms, offering a complete, full featured VoIP solution unmatched by any other vendor.

The Nortel Networks Internet telephones provide support for a wide range of today's high-value eBusiness applications, including CallPilot Unified Messaging and Symposium Call Center services. This rich, future-proof feature set will evolve to support advanced services such as voice-activated dialing and corporate and personal directory services.

Nortel's IP Phone 2007 incorporates a 5.7" (12.7cm x 17.8cm) color touchscreen, bringing multimedia presentation support to the desktop IP Phone. Virtual Network Computing technology optimizes content pushed from external application servers in advanced text or graphical format. A USB port supports standard USB mouse and keyboards for simple "point-n-click" access and navigation.

Nortel's line of IP phones includes five desktop units, a conference phone, wireless phones, and a conference phone.

Polycom soundpoint IP family

www.polycom.com

The Polycom VoIP product portfolio includes

the SoundPoint IP family of desktop phones, an attendant console based on the SoundPoint IP 601 and Expansion Module, and the SoundStation IP 4000 conference phone. With the greatest breadth and depth of integrated video, voice, and Web solutions, only Polycom delivers the ultimate communications experience.

Polycom VoIP phones leverage the legendary Acoustic Clarity Technology — validated by 13 years of market leadership in voice conferencing — to deliver exceptional sound quality and full-duplex interactive conversations that are as natural as being there.

Polycom's cutting-edge software enables enhanced call handling, security, and advanced applications on Polycom VoIP phones. Multiple call and flexible line appearances, HTTPS secure provisioning, presence, instant messaging, three-way local conferencing, and custom ring tones are just a few of the many software features developed to augment the customer experience.

Polycom (quote - news - alert) VoIP phones are your future-proof choice for business IP communications. Their standards-based phones can be upgraded to new software and firmware in the field. Polycom VoIP phones will meet your evolving needs, whether you switch to a new call server solution or merely prefer to keep your organization on the cutting edge of developments in standards-based protocols.

ShoreTel

www.shoretel.com

ShoreTel (news - alert) offers a wide range of telephones to provide the right solution for any. ShoreTel phones provide end users the features and quality they demand and are pre-configured to match the ShoreTel system to eliminate configuration issues during installation

ShorePhone IP telephones are designed to please the eye as well as the ear. The concave sweep of the face places the keys on a horizontal plane while keeping the display vertically aligned for easy viewing. A bright, backlit display (IP 560 / 530) is easy to read, and the message waiting light is visible from a full 360° viewing angle. Color-lit line buttons (IP 560) provide immediate, at-a-glance information about incoming calls and messages. Designed to conform to the human body, the IP telephone encourages interaction. Tactile keys are comfortable to the touch and reduce strain on the fingers and wrists. The precision balanced, contoured handset includes a cushioned grip and finger notch that makes it comfortable to hold or rest on the shoulder.

ShorePhone IP telephones offer a wideband audio codec that supports seven full octaves of human sound — far superior to the three-octave capability of other vendors' phones. High fidelity, full duplex speakerphones (IP560 / 530 / 210) deliver audio with astonish ing clarity — adding immediacy and depth to hands-free conversation.

All ShorePhone IP phones support plug-and-play installation — reducing time-to-deploy and easing administrator workload. As you connect new telephones to the network, ShoreWare™ Director automatically discovers the new devices and adds them to the ShoreTel system, where you can manage them as nodes on your enterprise network.

Siemens

www.communications.usa.siemens.com

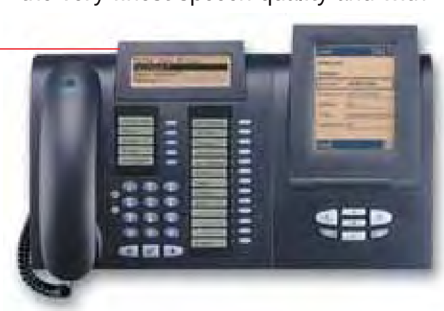
You need the right phone to make full use of IP telephony, and Siemens (news - alert) offers a particularly broad range of products: in addition to the optiPoint 410 and optiPoint 420 families, the convergent optiPoint 600 — the new flagship model — offers you the highest levels of flexibility. With optiPoint SIP telephones, Session Initiation Protocol (SIP) is used for connection control and for the transmission of speech in IP networks.

Many optiPoint IP models have a dedicated mini-switch that ensures data and speech can be exchanged on one line. Adaptation to individual needs is as simple as downloading the appropriate software. Service and software upgrades are administrated remotely. Relocation is also easy with optiPoint IP phones: connect the phone to the LAN, enter the phone number and password, and you're ready to call.

OptiPoint 410 takes IP telephony to a whole new performance level. It's the ideal solution for enterprises that want to introduce or consolidate modern communication structures and at the same time reduce costs. The optiPoint 410 telephone range is perfect for connection to real-time IP systems.

Choose the optiPoint 420 family to gain the scope needed for more mobility and flexibility within the company. Thanks to the new technology for the automatic transfer of key assignments, your own key arrangement is always available to you after logging on — regardless of which optiPoint 420 you are currently phoning from.

Convergent networks demand a new breed of telephone. The optiPoint 600 office supports connection to TDM voice networks as well as IP data networks, giving you maximum flexibility. The large display with graphic capability doubles as a touchscreen. All this in the very finest speech quality and with



all the performance features your communication infrastructure provides.

Snom

www.snom.com

With the VoIP telephones snom 360, and snom 320, you are choosing the optimum level of functionality and comfort in terms of quality, security, and equipment.

In addition to the common standards, the Linux-based SIP telephones support the most modern technologies.

Due to the two VoIP security standards SIPS (RFC2246) and SRTP (RFC3711), from now on potential eavesdroppers of conversations between your snom telephones and other compatible terminals don't stand a chance.

You will also have decisive benefits through high interoperability. The combinability with numerous devices of other established manufacturers such as Cisco, Siemens, or Microsoft guarantees you the greatest possible independence.

Ideal for general office and knowledge-worker environments, the snom 320 is an affordable, yet powerful SIP business telephone with built-in, full duplex speakerphone and three-party conference bridging.

The snom (news - alert)360 was designed for maximum productivity and efficiency in the everyday business environment. Dedicated keys provide you with direct access to the functions for audio and call control, and context-sensitive menus offer you the additional functionality that you may need at any given moment. The graphical display can be tilted for optimum reading angle.

SOYO Communications

www.soyousa.com

Soyo IP phone is an advanced IP device, allowing its users to experience superb voice communications over IP network. It works as a standalone device (no PC is needed). Providing conveniences and functionalities of traditional phone, it perfectly converges the best of analog and IP phone features.

Using audio compression technology (CODEC G.711, G.723, G.729, etc.), it transmits its packets using minimal bandwidth without interfering with other data signals. Furthermore, to provide consistent superior voice quality over varying IP network conditions, it is equipped with advanced Quality of Service (QoS) technologies such as CODEC negotiation, enhanced jittering technology, lost packet reconstruction, echo cancellation, and packet delay recovery. Combining these features, it provides superb and reliable voice communications over congested public IP networks.

Tadiran

www.tadiranamerica.com

Available in several models with a wide range of features and options, Coral FlexSets are true information terminals offering versatility and flexibility. FlexSets scalability is evident in the basic design. As many as 148 user-defined buttons give users virtually unlimited power to define individual preferences. The optional LCD presents caller and calling-party information, and when using a FlexSet with soft-keys, the display changes to show applicable feature options for one-touch activation. The hot-dial keypad and handsfree speaker permit fast on-hook dialing, and the highly visible message waiting indicator announces new messages.

Install optional base-mounted modules to take advantage of Computer Telephony Integration (CTI) and other advanced applications that enable users to communicate and work more efficiently.

The advanced Coral FlexSet-IP phone enables remote and home offices to access all the Coral features and functionality at corporate headquarters. A simple RJ-45 connection to any LAN or WAN means no more complex changes when moving premises or moving employees; just unplug the Coral FlexSet-IP and reconnect it in its new location.

Talkswitch

www.talkswitch.com

With its rich, innovative features and simple configurability, TalkSwitch (news - alert) can transform the way you communicate — and improve productivity, reduce communication costs, increase collaboration, and stay connected everywhere

TalkSwitch is a complete PBX with auto attendants, voicemail, VoIP capabilities, advanced call management and

much more. TalkSwitch also includes the features that other companies charge extra for. Unlike phone company services like voicemail or call forwarding, you don't have to pay every month. Call forward, find-me/follow-me, ring groups, music on hold, mode scheduling and call screening are just a few of the built-in TalkSwitch features.

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

Need to integrate distant teleworkers or small branch offices with your TalkSwitch system? TalkSwitch VoIP gateways extend the reach of your TalkSwitch, delivering the benefits of the 48-CVA to even your smallest locations.

Telrad Connegy
www.telradconnegy.com

From a high-end executive phone to a simple, cost-effective solution for your part-time work force, Telrad Connegy (news - alert) has an Avanti phone extension to meet your every business need. But some users just need the best IP phone there is. Some users want to interact with a world of applications, view video feeds, access corporate databases or browse a website on their office phones. Some users need Telrad Connegy's i.Picasso

All Avanti phones work with AdvanceIP systems of any size, assuring your investment is protected as you grow. Moreover, as part of our Evergreen policy, the Avanti family was designed with backward compatibility so that our existing customers can easily upgrade their legacy system to work with Avanti phones.

All Avanti models come with advanced capabilities such as a new message indicator, hot-key dialing, voicemail and ACD integration, soft keys and programmable keys, and an option for wall mounting and for connecting a headset. All Avanti phones can be extended to support more programmable keys by attaching a DSS module. Individual models have more features.

Telrad Connegy's i.Picasso sets the standard for advanced IP telephones. Developed with a true vision of what

VoIP technology brings when it is stretched to its limits, i.Picasso offers the advanced capabilities of no other IP telephone. With its large, color rich, liquid crystal display, which also doubles as a touch screen, i.Picasso is truly one of a kind.

Whether in the main office or in a remote or home office, i.Picasso functions over any IP network. Connected to AdvanceIP, i.Picasso delivers all the features you expect from a state of the art high-end business telephone, such as Unified Messaging, Advanced Speed Dialing, extension usage indicators, call center connectivity and more. But this is just the tip of the iceberg: with its integrated HTML capabilities, you can easily configure it to interact with any HTML application.

Toshiba

www.tais.com

Toshiba (news - alert) IP telephones extend full telephone system functionality to any location in the world via your private intranet or the Internet. Remote users even enjoy the same productivity features as locally connected extensions. Now that's power and versatility — a true Toshiba advantage.

Are you using the handset or the speakerphone? Thanks to state-of-the-art Toshiba technology, your callers will have a hard time telling. You'll appreciate ergonomic design for exceptional comfort, easy-to-read feature buttons, and programmable buttons that condense operational sequences into single-button ease.

Toshiba Digital Display Speakerphones make using your business communication solutions simple. Large easy-read LCD screens display feature-prompting information, helping you manage your telephone easier. One-touch programmable buttons save time and give you faster access to advanced calling functions. It's all designed to enable employees to work smarter, minimize training time, and take productivity to new levels.

Select from the 20-button or 10-button 2-line display model, or the 14-button 8-line display model, each available in your choice of two colors.

Zultys Technologies

www.zultys.com

Zultys (news - alert) offers a line of fully functional phones that provide businesses with choices of feature set and expenditure. Each phone provides a high degree of functionality, all in a well engineered business device. From the simple business phone to the highly integrated remote office phone, all operating on open standards, making them a great complement to any IP PBX deployment.

The return on investment in Zultys products can be easily observed through many aspects of an enterprise. Streamline deployment and maintenance of these products provides little need for third party services. Intuitive interfaces means less time learning tools. Highly integrated features increase the productivity levels for both administrators and users. Scalability of systems provides room for growth regardless of a company's size. Finally, Zultys' adoption of open standards ensures interoperability of its products with other equipment, which gives a business choice in purchasing devices and services.

The ZIP 4x4 integrates a business phone with a line-rate Ethernet switch. In addition to four call appearances and four Ethernet ports, the device is loaded with functionality, including encryption, calculator mode, single button functions, adjustable LCD, full duplex speakerphone, headset jack, Power over Ethernet, message waiting indicator, and QoS support. Using an automated rollout procedure supported by the ZIP 4x4, IT managers are discovering that with the ZIP 4x4 they can minimize their effort in installation and deployment of a VoIP system. Because it's 100% based on open standards, the ZIP 4x4 can be used within any SIP-compatible network.

Zultys offers seven additional models, plus a softphone, and a wireless phone. IT

Speaking With… …Earl Comstock

Earl Comstock
President and CEO
COMPTEL

2005 was certainly a year in which regulatory topics remained at the forefront of many of the discussions involving our industry. Issues such as 911, lawful intercept (CALEA), taxation, and more were at the core offerings of many conferences and keynote speeches throughout last year. And yet, as 2006 dawns, many questions remain unanswered.

I had the chance to ask Earl Comstock several questions regarding his thoughts on the state of VoIP regulation. Earl Comstock is President and CEO of COMPTEL, an association formed in March 2005 by the merger of COMPTEL/ASCENT and the Association for Local Telecommunications Services (ALTS). With more than 350 members, COMPTEL is a leading industry association representing competitive facilities-based telecommunications service providers, emerging VoIP providers, integrated communications companies, and their supplier partners. The association is based in Washington, D.C.

– Greg Galitzine

GG: What is the state of VoIP Regulation today? Where does our industry stand in relation to one year ago?

EC: This year has been a busy one for VoIP (define - news - alert) providers and regulation. The FCC preempted State regulation late last year and the Senate took affirmative steps this fall to provide access to 911 facilities and to provide VoIP providers with the same liability coverage as is available to other voice service providers. At the same time, the FCC has required VoIP providers to offer 911 service in a much shorter time frame than the agency has required of any other industry segment.

Over the past year, VoIP has increasingly come into the mainstream, with an ever growing number of providers offering it to businesses and consumers. COMPTEL members have been at the forefront of those offering VoIP, and VoIP issues will continue to be a key component of COMPTEL's legislative and regulatory efforts. We are continuing to work with interested Senators on passage of S. 1063, the Senate VoIP bill, and will work to get the House to pass the bill early next year.

GG: What does it say to the FCC that the U.S. Senate Commerce Committee has taken the stance that it did?

EC: COMPTEL (news - alert) applauds the bipartisan fashion in which the Senate Commerce Committee acted on S. 1063. The amendments in committee improved the bill, and the strong vote coming out of the Committee will help ensure the bill's passage. The bill addresses a number of issues that were outside of the FCC's legal authority to address, so the actions of the Committee complement nicely the action taken earlier by the FCC. It is COMPTEL's hope that the FCC will embrace the approach taken by the Senate Commerce Committee.

GG: Please explain and expand upon your recent comments, that "The House Commerce and Energy Committee's revised draft legislation shortchanges consumers by paving the way for the creation of gatekeepers to the Internet."

EC: Both staff drafts released by the House Energy and Commerce Committee would create gatekeepers on the Internet because both versions would abandon common carriage as the underlying framework for communications law in the United States. Instead, both versions of the staff draft would permit the companies that control the transmission facilities used to reach business and residential consumers to deny access by competitors to those facilities. Unlike the situation under common carrier rules today, these network operators would have no obligation to provide service to anyone, no obligation to extend their facilities to anyone, and no obligation to permit anyone to obtain transmission services at just and reasonable rates. The only obligation imposed on packet-switched transmission facility operators under both versions of the bill would be to directly or indirectly interconnect their transmission networks with other transmission network operators on whatever terms the parties arrive at through commercial negotiations.

Both drafts assume that there will be multiple network operators that have facilities that reach all consumers — notwithstanding the fact that, today,

Go To Table of Contents | Go To Ad Index

there is only one network operator with ubiquitous wireline facilities that reach all residential and business customers (the ILEC) and one other operator that reaches the vast majority of residential customers (the cable operator), but few, if any, businesses. While it is true that there are wireless and satellite operators who could potentially reach the vast majority of residential or business users, the reality is that their service offerings are more expensive complements to, rather than substitutes for, the wireline offerings. This truth is illustrated by the fact that less than 10 percent of wireless subscribers have elected to discontinue their wireline service and few, if any, businesses are served entirely by wireless today.

It is the gatekeeper aspect of the proposed drafts that is driving much of the debate on Net neutrality, along with the statements by the CEOs of several ILECs. Unfortunately, the Net neutrality provisions in both drafts have exceptions that basically gut the rule by explicitly allowing the network operator to favor its own services. And even if the Net neutrality provisions were fixed, net neutrality alone is not sufficient to prevent the network operators from becoming gatekeepers — if there is no obligation to provide non-discriminatory service and to interconnect with other network operators at just and reasonable rates, then the dominant network operators can and will advance their own financial interests simply by denying competitors access to the network in the first place. The Net neutrality provisions only apply to subscribers whom the network operator agrees to serve — if you can't get on the network then the Net neutrality rules can't protect you.

GG: What do you make of the recent comments attributed to SBC Chairman Ed Whitacre, where he takes an aggressive stance towards VoIP providers? Here's the Whitacre quote: "Now what they would like to do is use my pipes free, but I ain't going to let them do that because we have spent this capital and we have to have a return on it."

EC: I applaud Ed Whitacre for his forthright statement regarding SBC's business interests. Like any good CEO, he is working to maximize the value of his company for his shareholders. What his blunt statement reveals is that, in his assessment, the way to do that is through capturing the additional revenue that comes from the applications that ride over SBC's and other network operators' transmission networks. In Mr. Whitacre's assessment (an assessment apparently shared by other Bell company CEOs), it is not enough that their companies get paid for the service they actually provide — namely, transmission. Rather, they believe that they should get some or all of the value of the content that is transmitted over their networks. That is what Congress should not allow to happen — transmission providers, whether incumbents or competitors, should not be able to leverage their ownership or control of transmission networks to dominate, or extract a surcharge from, the complementary markets for goods and services that use those transmission networks to reach business and residential consumers.

Further, the reality is that VoIP subscribers have already paid SBC for the use of its transmission networks through the purchase of DSL service, and the common carrier that delivers the VoIP traffic to SBC (quote - news - alert) has likewise paid SBC for termination of the VoIP traffic on its network under the FCC's current access charge rules, so no one is using SBC's "pipes" for free. Mr. Whitacre's problem is that the FCC has failed, in the nearly 10 years since the passage of the Telecommunications Act of 1996, to address access charge and intercarrier compensation reform. As a result, some data transmissions, including VoIP transmissions, may pay SBC less for using the network than the circuit-switched voice services that VoIP is replacing. Most COMPTEL members would argue the lower rate paid by data transmissions is, in fact, the correct one that all transmissions should pay; most incumbent LECs would argue the higher rate should apply. The truth may lie somewhere in between. What COMPTEL supports is fair compensation for use of transmission networks, regardless of the application that is using the transmission network. There is no reason for video bits to pay more or less than voice or data bits for the use of the same transmission network; all bits that require the same quality of service should pay the same cost-based rate, including a reasonable profit, for use of the networks. Whenever the FCC gets around to establishing a rational intercarrier compensation regime that also addresses access charges, much of the current debate over the "free" use of any provider's network will disappear.

The Net neutrality provisions only apply to subscribers — if you can't get on the network then the Net neutrality rules can't protect you.

GG: What do you think of the job that Chairman Martin of the FCC is doing?

EC: While COMPTEL does not agree with all of the positions Chairman Martin has adopted or supported in his tenure at the Commission, the Chairman is to be commended for his open door policies and his considerable skill at getting his fellow commissioners to join him in adopting unanimous orders on difficult issues. He is a diligent public servant who is doing what he believes to be in the best interests of the Nation. To the extent that the assumptions made by the Commission in adopting any of its orders do not turn out to be correct, it is COMPTEL's hope that the Chairman will exercise his leadership to promptly adopt appropriate modifications to those orders.

GG: Please describe the net effect of the recent FCC order to expand the Wiretap Act (CALEA), which would impose upon carriers the need to

Go To Table of Contents | Go To Ad Index

revamp their networks to make it easier for the Federal government to monitor online communications.

Although COMPTEL certainly supports the purpose of CALEA the FCC has bent the CALEA statute beyond the breaking point.

EC: Although COMPTEL certainly supports the purpose of the Communications Assistance for Law Enforcement Act (CALEA) — ensuring that law enforcement has lawful access to telecommunications networks — the FCC has bent the CALEA statute beyond the breaking point. Because the FCC chose, in its Wireline Broadband Order, to completely deregulate broadband transmission services by reclassifying them as information services, it then had to quickly make up for the fact that it was cutting off law enforcement access to such services. As a result, the Commission was forced to conclude that broadband transmission services that it had just found to be information services were, nevertheless, covered by CALEA, notwithstanding the explicit exemption for information services that Congress wrote into CALEA.

Further, the FCC immediately set a deadline for "full compliance" for the information services providers it had swept into CALEA, but deferred to later decisions both the question of what compliance means and the resolution of possible exemptions from coverage. For example, the FCC appears to have swept private networks, including universities and corporations, into the purview of CALEA, as well as other traditional information service providers, such as email providers, despite Congress' clear ban on such inclusion. The FCC's CALEA ruling has put many entities in the impossible position of having to begin compliance efforts without knowing what is required and whether they are even covered. If the FCC had not bent over backwards to give the Bell and cable companies the freedom to discriminate in the provision of broadband transmission services, these contortions would not have been necessary.

GG: What is COMPTEL and what role is the organization playing in influencing regulation that affects VoIP and the future of telecommunications?

EC: COMPTEL is the nation's largest trade association, representing the competitive communications industry. We represent more than three hundred companies from across all communications sectors, including voice, video, and data service providers, equipment manufacturers, and content companies. Our member companies provide consumers with competitive alternatives to the entrenched incumbents. We advocate open networks that allow consumers to access the providers and content of their choice, rather than being forced to accept the limited choice of services and features the incumbent providers offer. We advocate policies at the FCC, in Congress, and in the courts that protect consumers against Internet gatekeepers.

COMPTEL's goal is to ensure that all policymakers understand that the widespread benefits of the Internet are available to Americans because of the pro-competitive policies found in title II of the Communications Act. Those policies ensure open networks and efficient use of communications networks built over public rights of way or using public spectrum. The Internet exists today because of those policies, not in spite of those policies, as advocates for closed networks would have the public believe. COMPTEL members are the entrepreneurial innovators who are bringing new services and content at competitive prices to consumers who want them. We developed and deployed DSL broadband, VoIP, and other new technologies and services before the incumbent providers. COMPTEL is the association of choice for entrepreneurial companies; we will continue to work on successful enactment of legislation enabling VoIP providers to offer 911 service and on pro-competitive reforms to the Nation's communications laws. IT

Healthcare leader in four-state region chooses a converged voice and data network to advance patient care

Imagine a world where nurses quickly order medications or enter a patient's status directly into an electronic chart from the patient's bedside using a Mobile Medical Workstation; or where doctors and hospital staff can log into the core network from any remote location over a secure virtual private network connection and have full access to the information they need to deliver first-rate patient care; or where healthcare workers from one branch location can plug into the hospital's network while at another site and access all of the same voice and data features.

That is the world emerging at Erlanger Health System, a non-profit, academic teaching center affiliated with the University of Tennessee College of Medicine in Chattanooga, serving southeast Tennessee, north Georgia, north Alabama, and western North Carolina.

With 10 locations and some 900 physicians, delivering health care that makes a difference in patient quality of service is a life or death matter for Erlanger. Such responsibility is one of many reasons Erlanger selected a Nortel Networks converged voice and data network to expand the reliability, manageability, and information access of its communications network.

Network aches and pains

Information technology at Erlanger is essential to efficiently delivering patient care. According to network director John Haltom, "It's important that necessary information is made available when and where it's needed to facilitate clinical and business decisions. This means providing a variety of communications services among our several campuses, departments, disciplines, and care teams as well as to our patients and their families."

With a communications network playing such a critical role in meeting the needs of patients and staff, it is not difficult to understand why Haltom and Erlanger's 16-member IT team found themselves concerned and frustrated with their existing network infrastructure. Erlanger's separate voice and data networks covered a 125-mile radius and some 30 sites, making the infrastructure costly and cumbersome to manage.

"What's more," said Haltom, "we found ourselves in the very precarious position of having an unsupportable network. Due to churn, including vendor acquisitions and discontinued service support for our existing ATM Local Area Network (LAN), our data network was in serious jeopardy. Our voice network included PBXs that were no longer under warranty and were difficult to maintain. It was absolutely necessary to replace our network end-to-end or be without vendor support."

"Our goal," Haltom continued, "for about five years has been to put in place a common infrastructure for all voice, data, and video services provided and then take advantage of converging technologies as industry standards became established and adopted."

The Vendor Search

Erlanger launched a comprehensive, two-year vendor selection process to determine the best means of meeting its long-term network objectives. Haltom and his team clearly delineated the evaluation criteria and approached Nortel Networks, as well as Cisco, Marconi, Alcatel, and Extreme.

The criteria for vendor selection were as tough as it gets. Erlanger looked closely at the scope of functionality and services the new network would support, as well as other important factors,

Go To Table of Contents | Go To Ad Index

such as network management, cost and time requirements for training the IT team and hospital staff, the extent of vendor support in the form of technical manpower, and Web-based support services, maintenance, investment protection, and especially the total cost of ownership of the network.

"We gave each vendor a set of criteria on which to base a network scenario for the bid," said Haltom. "This provided a visual reference for us to use in evaluating the strengths and weaknesses of the offerings. Because of our dire situation with our previous vendor, we also verified the financial stability of the bidding companies looked closely at their ability to meet our future needs."

"In the final analysis, Nortel Networks not only delivered more options than we had asked for, but the company did so with a single-switch configuration, full end-to-end manageability, and a very simple-to-implement network plan," Haltom explained.

"Where other vendors had trouble meeting all of our requirements, Nortel (quote - news - alert) Networks excelled in providing an elegant, complete solution."

Furthermore, even though there were less expensive offerings, in terms of initial capital outlay, the total package solution from Nortel would cost far less and provide more functionality over the course of three years.

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Anatomy of a converged VoIP network

The network overhaul for Erlanger Health System involves a three- to five-year phased-in approach to prevent disrupting the delivery of patient care. The data network, once at risk due to product discontinuations and a lack of vendor support, now is supported solely by Nortel.

• Nortel Networks Passport 8600 routing switches and Business Policy Switch (BPS) 2000 create the data backbone for a future-proof end-to-end converged VoIP solution.

• A Synchronous Optical Network

A Pervasive Wireless Vision

The Miller School of Medicine at the University of Miami facility includes 770 faculty physicians and some 6,000 employees and, in addition to the 76-acre complex in Miami, it operates a dozen other facilities throughout Southern Florida. Chris Bogue, the institution's IT Director at, has a vision: To enable appropriate access to any network or information resource for anyone, anywhere, at any time.

To execute his ambitious plan, Chris has become one of the medical IT community's leading proponents of wireless technology, and he has been among the first IT executives in North America to address the challenges of truly pervasive wireless access that supports a full range of applications across a large and geographically dispersed facility.

Today, Meru Networks is helping him realize his vision. With millions of square feet of classroom, hospital, clinic, laboratory, and administrative space to cover, it was clear to Bogue from the beginning that wireless LAN technology would be the only feasible approach.

While his vision of pervasive connectivity includes general wireless access to e-mail and Internet, it is primarily driven by the convergence of biomedical technology and IT technology in applications, like wireless patient charting systems, EKG machines that travel with patients, beds that monitor patient vital statistics and relay them to nurse stations, and wireless video transmissions that educate patients about their health issues.

"As the lines between biomedical and IT disciplines begin to blur," observes Bogue, "a lot of biomedical devices will eventually connect to wireless infrastructures in hospitals. We want to make UM School of Medicine one of the leading organizations for leveraging wireless technology in healthcare."

Bogue's team began executing his vision in 2002 by deploying wireless access points for student use in the cellular biology classrooms and in hospital operating rooms at the University's Bascom Palmer Eye Institute. Since then, Bogue's team has continued to expand wireless coverage and applications with a series of initiatives, including:

- Deployment of mobile wireless carts in other hospital areas to support registration, medical records access, patient scheduling, and clinical information applications;
- Additional classroom coverage throughout the medical school;
- IP voice communication badges that link staff and physicians between clinics in Naples, West Palm Beach, Deerfield Beach, and the main campus via a secure private WAN to enable immediate communications and eliminate long distance telephone charges;
- A facilities work order management system that allows employees to use wireless barcode readers to look up preventive maintenance schedules or

Go To Table of Contents | Go To Ad Index

maintenance histories or to order parts for heating, cooling, and other systems;
- A “Community Cloud” providing 1.5 square miles of outdoor wireless coverage.

However, with the expansion of its wireless LAN deployment, the IT team began to encounter rogue access points set up by students or others who wanted to jump on the wireless bandwagon more quickly than specified in the IT department’s rollout plans. A far more difficult problem, however, was the need for rapid scalability. “This is a pretty large organization, and there are a lot of departmental moves and changes from one facility to the next,” says Bogue.

Meru’s WLAN system was the answer for Bogue:
- It coordinates all traffic on the network and eliminates co-channel interference by placing all APs on a single channel;
- Meru’s Virtual Cell Technology eliminates handoff delays and creates seamless access;
- Over-the-air Quality of Service (QoS) for both downstream and upstream traffic ensures high quality voice and data service to all Wi-Fi clients;
- It supplements existing APs with the Meru WLAN System while preserving existing investments in other products, such as the outdoor WLAN product;
- It allows for easy scalability to meet the dynamic requirements of the medical center by automatically optimizing coverage and compensating for shifting user density and application loads
- The institution derives significant ROI by eliminating costs associated with site surveys or RF planning and by leveraging a single infrastructure to deliver voice and data applications. To date, UM has spent approximately $800,000 on its wireless implementation, with an expected return on investment in less than 24 months.

The return on investment comes predominantly from the efficiencies gained in mobilizing the workforce while being able to tools to get the right information to the right people at the right time. In the last four months, there has been a 26% increase in the number of wireless devices using the network and it continues to rise. Within the next two years, it is estimated that the network will sustain between 700 and 2200 concurrent wireless connections. Given the millions of square feet for this deployment, the ability to provide seamless pervasive coverage would have been virtually impossible with any other product. IT

(SONET) smart ring provided by BellSouth uses Nortel Networks high-speed OC-3 links to interconnect the core at Erlanger’s downtown campus with the company’s three main remote campuses.

- To provide the infrastructure and QoS to support advanced voice communications, Nortel simplified Erlanger’s network with an Ethernet solution versus the previously costly and hard-to-manage ATM configuration.
- Nortel’s Succession Communication Server for Enterprise (CSE) 1000 — a fully distributed IP PBX — supports a wide spectrum of leading applications and telephony features. With its Release 2.0 software, the Succession CSE 1000 will allow Erlanger to have seamless network integration, simplified network management, greater flexibility in network deployment, and reduced costs for supporting its users across multiple sites and those needing remote access.
- Six Succession CSE 1000 Branch Office Gateways and two Nortel Networks Business Communications Manager (BCM) systems will be deployed throughout Erlanger’s metropolitan region.
- The Contivity Virtual Private Network (VPN) client software will allow users to securely access the network from remote locations
- The Optivity Telephony Manager 2.0 (OTM) brings converged voice and data network management to Erlanger’s IT department.
- Physicians, nurses, and staff will gain access to full featured desktop solutions with a broad range of features using Nortel Networks i2002 Internet Telephone, i2004 Internet Telephone, i2050 Software Phone, and CallPilot 2.0 Unified Messaging.

Abundant network benefits

During Erlanger’s two-year vendor selection process, many benefits of the Nortel Networks solution emerged — Haltom says Erlanger continues to discover new advantages.

“Our biggest benefit was the immediate feeling and realization that we — our IT staff — are now in control of the network, not the other way around," said Haltom. "The network was easy to implement and still is. We have so many application tools and troubleshooting techniques that were non-existent in the past. The combination of Nortel Networks' latest suite of network management solutions makes configuring, installing, managing, administering, and recovering a breeze."

"The management of a single converged voice and data network is freeing so much of our time. We used to fight fires and do crisis management. Now we're actually able to be proactive, to take on projects we couldn't do before, to provide QoS, and guarantee our users that their mission-critical applications will be available."

Haltom cites other important benefits:

- Integrated infrastructure — The Succession CSE 1000 will let Erlanger operate a converged voice and data network and manage it as one. It will make possible the delivery of integrated services, and positions Erlanger to fully leverage next-generation applications such as unified messaging, mobility, centralized directories, and other emerging services.
- Costs savings — In addition to substantial savings in training time and costs over the original ATM LAN equipment, Haltom estimates an overall cost savings of $40 every time Erlanger deploys an i2002 or i2004 Internet Telephone compared to a traditional digital set — the organization expects to replace some 1,000 existing digital sets.
- Ease of administration — The Optivity network management solutions make monitoring, troubleshooting, managing VLAN (define - news - alert) configurations, and implementing business-level policies across the network "a true networking lifesaver." Also, the Internet Telephones will automatically register users with their same services at their new location without

Health Care Provider Advocates IP

Advocate Healthcare is Illinois' largest full-service health care company and the state's second largest employer. Advocate operates eight acute care facilities, two children's hospitals, several state-of-the-art medical facilities and a physician's network of 4,600 doctors at over 200 sites, all of which necessitates that each associate have the ability to communicate quickly and efficiently. Standing up to this challenge, Advocate has implemented an IP network that ensures seamless communications across the entire company.

The challenge

Advocate Health Care has grown through mergers, the addition of new facilities, and the creation of a physicians' network. The mixed nature of Advocate's communications network created a difficult situation on many levels. For instance, the cost of operating a separate voice only network was far too high. Additionally, it was difficult to integrate Advocate's business systems, including its various call centers and automated customer service functions, because of the differences between hardware at each site. In fact, there were as many as 29 different management platforms in place. A single distributed communications network would make communication easier and faster, not only for patient care delivery, but also for overall operations.

It was important to deploy a single telephony standard throughout the entire Advocate infrastructure without completely disrupting its business operations. A dedicated voice network would enable remote management and automatic updating of all nodes in the network, and the deployment of new services across the network. A single telephony network would also provide a comprehensive call center that could handle the over 16,000 calls per day through their central appointment system.

The solution

Advocate's new network implementation was a challenging undertaking; it was vital that any upgrade not disrupt business operations on a large scale, so it needed to be installed in pieces over time. The Alcatel OmniPCX Enterprise was the perfect solution, due to its flexibility and ability to interoperate with many existing systems. Using a combination of IP, digital, wireless, and analog technologies, Alcatel partnered with Advocate to devise a plan to migrate Advocate from its many disparate systems to a single network-wide communications system.

After reviewing the requirements at each location, the company decided to utilize its existing data infrastructure to support the voice network — this would also result in a significant cost savings. The redesigned IP communications platform provides Advocate with a next generation communications infrastructure that doesn't sacrifice legacy systems and services.

Go To Table of Contents | Go To Ad Index

Where possible, IP phones were installed parallel to the existing telecommunications systems. The corporate support center also was converted entirely to IP. Once the communications solution was operational, the old system was removed with no disruption. The flexibility and scalability of the communications server enabled each site on the network to be upgraded while communicating easily with the old infrastructure. This enabled a gradual migration to a new communications infrastructure.

Where IP was not needed, digital, analog, or wireless handsets were installed. Because the new communications solution is standards-based, it was possible to interface it with existing systems that provide additional services, thereby utilizing embedded equipment and easing the cost of migration.

The deployment of new functions, like mobile wireless handsets, Interactive Voice Response system (IVR), dial-by-name phone book, text messaging, and network-wide voicemail further enhanced the ability of Advocate's associates to perform regular duties, while leaving room for potential future services. Today, doctors who have more than one office across different Advocate locations can have the same phone number and a single voice mail box, regardless of which office they are using at the moment. Ultimately, these features enable Advocate to serve its patients better and to operate more efficiently.

Because collaboration is essential to not only efficient operation, but also to any future communications enhancements, Advocate is conducting ongoing planning and trials to ensure that connectivity and effectiveness will not be sacrificed in the future. This includes investigating the use of Unified Communications; developing a strategy to use XML to create new services; and building a high capacity network to link two fully redundant data centers. This new high speed network will connect all of our Acute Care Centers and provide a hierarchical QoS to ensure speedy delivery of time sensitive data.

At one hospital, Advocate is using testing a tool to integrate its bed-tracking system to wireless handsets, which allows nurses to automatically page the responsible maintenance person to a specific room. The health care system also plans to upgrade its data network to incorporate 802.11, to provide physicians with handheld devices, and to integrate Alcatel's OmniTouch Unified Communications software into the handheld devices. IT

administrative involvement, allowing network staff to focus on more mission-critical responsibilities.

• More robust connections — Once limited to a few 100 Mb connections to the hub and servers, Haltom said his department is now using multiple Gigabit Ethernet interfaces to support imaging and application servers throughout the network.

• Secure remote access — The Nortel Networks Contivity VPN solution and the i2050 Software Phone combine to provide traveling staff with a means of staying in touch by logging into Erlanger's network. With a simple Internet connection via dial-up modem or DSL service, users get secure, high-quality voice with four-digit dialing wherever they live or travel. In addition, users have transparent access to the same advanced telephony features available on-site at the hospital.

• Effective, satisfied users — "I have people lined up and waiting for some of the applications our new network makes possible," said Haltom. Nurses participating in the pilot of the Mobil Medical Workstations (see "Traveling Nurse's Station," page 84) "can't get enough of the new technology; others are knocking at my door wanting the branch-to-branch VPN and remote access using the i2050 Software Phone."

Investment protection

Erlanger envisions many opportunities for network expansion and new healthcare applications as it completes its current installation and looks to the future. The healthcare environment necessitates taking the time for a smooth installation, conducting beta tests and network trials for the various components, replacing equipment, and rolling out added functionality carefully to ensure high-quality patient care throughout the network's implementation.

That said, Erlanger feels confident that its Nortel Networks converged VoIP network offers investment protection for years to come and creates a

"Where other vendors had trouble meeting all of our requirements, Nortel Networks excelled in providing an elegant, complete solution."

solid foundation for the future. Erlanger anticipates continued network expansion and support of new applications, such as imaging over Digital Subscriber Line (DSL) technology to a physician's home or to any number of Personal Digital Assistant (PDA) devices used by roaming or remote personnel.

"We didn't enter into this agreement without a long-term commitment," reminded John Haltom. "We fully intend on keeping this partnership going, and we'll work together through every beta test to continue putting intelligence in our network." When asked to describe the greatest benefit of his company's new converged voice and data network, Haltom returns to what's really important at Erlanger: "Our greatest benefit," he said, "is having a strategic partnership for advancing patient care. What Nortel Networks has allowed us to do is to put tools in the hands of our staff to improve patient care. That's our job." IT

Traveling nurse's station brings better care to the bedside

One of John Haltom's favorite applications riding on Erlanger's new network is the Mobile Medical Workstation used by nurses. These high-tech rolling carts use 802.11a wireless technology to stay in touch wherever they go, thereby providing the caregiver with roving access to the Hospital Information Network. With the i2050 Software Phone, nurses can have unrestricted access to a telephone wherever they need it, allowing them to order, from the patient's bedside, needed tests, treatments, medications, x-rays or lab work, and to enter new data on the spot. The nurses are so fond of their mobile pals that many have given them pet names, like "Petey."

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Planning for VoIP? Know You're Ready

As the new year approaches and IT budgets replenish, many companies will have VoIP pilots and deployments at the top of their 2006 "To Do" lists. If you are considering VoIP or expanding the use of VoIP applications, you are probably concerned whether an existing network can support high-quality voice transmissions. This article outlines what you should consider when assessing for VoIP readiness.

To ensure your network is ready to handle the increased network load and ensure proper configuration so voice traffic will run smoothly across your network, you must consider four key requirements before installing the first IP phone on your network— Building a Network Inventory, Utilization Assessment, Bandwidth Modeling and VoIP Quality Assessment. Your end users have high expectations for uptime and call quality due to the highly stable PBX systems. The uptime and call quality end users expect is obtainable through proper assessment and reporting of your network.

Network Inventory

When building a network inventory, you must account for the entire infrastructure necessary to support a VoIP deployment. Routers, switches and links are configured to support your data network. VoIP traffic is extremely time sensitive so voice packets cannot be queued and must have priority over data information. If your network is not configured correctly, end-users' calls will sound jittery or even incoherent.

Discovering all network assets will also help you assess whether any hardware upgrades or additional hardware is necessary to support the converged voice and data network. When testing each component of your converged network, you essentially guarantee support of voice and data and reduce the probability of jitter, packet loss, low latency and clogged bandwidth.

Utilization Assessment

Since VoIP traffic requires different configuration and requirements, you must understand the current utilization of your network devices and links. Assessing utilization involves gathering data on routers, switches and WAN links to determine their capacity for carrying VoIP traffic—looking for specific information on queuing and drops, bandwidth, CPU and memory.

These metrics will help you determine your network's ability to handle VoIP traffic. The information might also establish a configuration baseline for your network devices. While collecting the required data from these devices can be time consuming, NetIQ solutions can not only automate the process, but also provide extensive reporting on the results.

Bandwidth Modeling

You must also assess your network for bandwidth increases. Bandwidth modeling looks at existing telephony usage and network data utilization to determine if your network can support future VoIP traffic. Modeling is the "what if" analysis in VoIP readiness.

Four key parameters that provide the data necessary for a successful model are bandwidth capacity, current bandwidth utilization, number of calls and type of codec used. When measuring bandwidth capacity and utilization, you must obtain metrics from various times in the day. You should also measure the number of calls as the total number of calls to support at any given moment.

Determining the correct codec is crucial to call quality and bandwidth used. Codecs that require 64 Kbps provide a higher call quality but also require more bandwidth. One that requires a lower Kbps may allow for more calls, but could lower your call quality.

The simplest type of modeling uses the projected call volumes, codec selections and bandwidth requirements as input. Calculations from these projections determine the amount of bandwidth needed to support VoIP. You must perform these calculations on a regular basis with different variables. You should consider different software solutions that provide this information, such as NetIQ.

VoIP Quality Assessment

VoIP quality assessment is the fourth key component to a successful VoIP readiness test. Voice traffic places a new set of requirements on data networks with the goal being to provide excellent call quality which is measured on the average mean opinion score (MOS) assigned to VoIP traffic. MOS takes into account metrics, such as codec, delay, lost data and jitter buffer loss that can affect call quality. Higher MOS means higher call quality.

Simulating voice traffic on your network before you install your first IP phone is the best way to measure your network's call quality. You should measure call quality at every location or call group that will be in your deployment. To determine the MOS, an assessment solution should place simulated voice traffic over your network. You should also test at various times and locations to establish a comprehensive assessment.

Conclusion

An accurate VoIP readiness assessment along with proper reporting will verify your network is ready to deploy VoIP. With the right solution, the process of deploying VoIP successfully on a data network can become a clear-cut process. Knowing your network assets are ready —and knowing your optimum network utilization and bandwidth capacity—will help you reach your goal of ensuring perfect call quality. With NetIQ's solutions, you know your network is ready for VoIP.

Randy Rosenbaum is Product Marketing Manager at NetIQ. For more information, please visit the company online at www.netiq.com

NGT Treats Level 3 Business VoIP Resellers and End Users to Quick Switch

The year 2005 started off with a shock to the business VoIP world: In January, Level 3 Communications announced it was discontinuing its (3)Tone hosted IP PBX (define - news - alert) business service. Dozens of Level 3 resellers were suddenly faced with a tight deadline for transitioning hundreds of businesses and thousands of (3)Tone business VoIP end users to a new platform and new wholesale provider. New Global Telecom (NGT) moved quickly to sign resellers and start transitioning end users.

By leveraging its close relationship with Level 3 and coordinating with newly signed (3)Tone service provider resellers, NGT (http://www.ngt.com) began a migration project that would successfully and seamlessly transition more than 60 percent of Level 3's entire (3)Tone customer base by the June 15, 2005 deadline.

Situation:

New Global Telecom (news - alert) knew it faced a tough assignment: migrate 5,500 end users from 245 different companies to its 6DegreesIP hosted IP PBX service in five and a half months — all with minimal downtime, while remaining hidden from service providers' end user customers.

"While we were pleased to get the lion's share of Level 3's (3)Tone customers," said Mike Coar, NGT director of client services, "we knew it would be an involved process with tight deadlines, requiring end-to-end coordination and cooperation to effect a seamless, on-time migration."

NGT's new service provider resellers included TMC Communications, (news - alert) a California-based provider of long distance, local, conferencing, data, and Internet access services (http://www.tmccom.com); Access Point, Inc., (news - alert) a North Carolina-based provider of integrated business communications services (http://www.accesspointinc.com); and CentricVoice, (news - alert) a Texas-based voice-enabled ISP (http://www.centricvoice.com).

Jeff Rothell, President and CEO of CentricVoice stated, "We understand what it takes to be successful delivering VoIP services to small- and medium-size enterprises. So we placed our confidence in New Global Telecom to successfully migrate our customers and support our business."

Solutions:

To ensure a successful migration, NGT conducted testing and established a sound migration process. Testing was conducted 10 times during a six-week period in February and March. Because end users were moved from the Sylantro platform at Level 3 to the Broadsoft platform that NGT uses, NGT conducted customer premises equipment (CPE) testing in its product lab, moving demo accounts from the (3)Tone service to 6DegreesIP and working with various types of customer hardware and architectures.

"We tested, developed, and perfected processes," said Caitlin Clark-Zigmond, NGT director of product management. "We provided a detailed 12-page migration plan for the service providers and then performed their internal migrations before moving onto end users."

Clark-Zigmond said that migrating service providers at the beginning allowed NGT to identify potential problems in advance and bring the service provider up to speed.

At the end of March, NGT hosted a group training session for service providers. The session lasted four days and trained 24 staff members from seven service providers.

"This advanced preparation gave service providers a comfort level with the process before a single end user was migrated," stated Clark-Zigmond. "The testing and documentation impacted the project by speeding up migrations and reducing downtime."

“We had to be more than just adequately prepared, and having everyone in one place at one time before the actual migration proved more effective than individual on-site training,” said Coar. “Service provider staff benefited by hearing about challenges other SPs were facing and by working through solutions to those challenges in advance.”

The training and preparation paid off, with end-to-end migrations averaging 30 minutes per site (the average site had 20 seats). The average downtime per site was about two to three minutes, with NGT being completely hidden from the end user, who was unaware of the multiple vendors involved in the VoIP service provider transition.

NGT remained flexible early in the conversion process. For example, Level 3’s daily service activation time (when Level 3 sets its routing tables to determine how calls are handled) occurred just as many East Coast end users were arriving at work. That prompted NGT to work with Level 3 engineers to construct a two-phased implementation. The work-around allowed for a temporary routing change to occur after East Coast business hours, with the final routing change taking place in the morning.

“We looked at the two-phased approach as training wheels for our service providers,” said NGT Director of Network Operations Glenn Pearston. “Once they were comfortable with the process and ‘got their balance,’ they found it easier to conduct the migrations in one fell swoop, with the first phase becoming more of a hindrance that was no longer needed.”

Hardware compatibility also presented some initial hurdles. There were so many different products on the market, that each and every potential configuration could not be completely recreated during testing. This resulted in some additional compatibility testing during the migration process.

“The good news is, we actually discovered just how compatible our 6DegreesIP service on the Broadsoft platform is with the majority of hardware out there,” said Clark-Zigmond.

Through preparation, continuous process improvement, and documentation, NGT was migrating nearly 20 companies per week by mid-May. During the 10-week live transition period, NGT successfully migrated an average of 35 sites and 550 end users per week with an end-to-end migration time of less than 30 minutes per site. Average downtime was less than 2 1/2 minutes per site.

The Benefits:

New Global Telecom’s testing, training, and operational experience in hosted IP PBX led to a smooth, seamless transition for service providers and their end users migrating to NGT’s 6DegreesIP wholesale hosted VoIP service. Service providers, end users, and NGT realized numerous benefits from the successful migration.

Service Providers

Service providers achieved a seamless, cross-platform transition in a critical situation on a tight deadline. With access to NGT’s experience, extensive training, Web portal, and compatibility testing, service providers were able to switch to the robust 6DegreesIP product suite with minimal downtime. NGT’s robust product suite includes hosted IP PBX and Class 5 features, end customer support, network and facilities management, and back office functionality.

End Users

End users also benefited from the minimal downtime, so they could continue their work with little or no interruption. The seamless transition also kept NGT behind the scenes and gave end users what amounted to a significant upgrade that included an integrated toolbar for click-to-dial functionality from Microsoft Outlook and reception console for front desk reception management.

NGT

For New Global Telecom, the rapid and successful completion of the migration has reduced trouble tickets overall and increased customer volumes. NGT has expanded its base of knowledge for dealing with hardware and software compatibility and migration issues, and is leading the industry in bring new features and functionally to market through its hosted IP PBX product suite. IT

2005 Internet Telephony Products Of The Year

It's time yet again for INTERNET TELEPHONY®'s Product of the Year awards. Each year, the editors of INTERNET TELEPHONY® sit down and pore over hundreds of applications submitted over the course of the preceding several months. And each year, it becomes more difficult to select companies for the honor.

The list, at least in my view, has never been about selecting a single product that stands out above the rest. Frankly, I think it's impossible to choose one product across multiple categories and multiple target audiences that should be considered "the best." For example how does one compare an IP telephony headset to a softswitch, to an operator's console, to a video headend, to a development toolkit…?

For me, the list has always been about providing our readership with a grouping of products that merits their consideration. Pure and simple. If one of our readers is in the market for an enterprise IP telephony, then there will appear among this list of vendors several providers of such a solution. The purpose of this list is to provide a starting point for individuals seeking solutions that will help them achieve their goals, be it to save money, grow their business, or embrace the hottest telecommunications technology out there.

In our February issue,we will expand upon the simple list you see here, with brief write-ups detailing each company's award-winning solution, making it easier for readers to drill down and find the solutions they're looking for. A more complete listing will be made available online at that time as well. For now, rest assured that each of the companies presented on these pages brought a product or solution to market in 2005 that is certainly deserving of your attention.

COMPANY	WEB	PRODUCT
3Com	www.3com.com/voip	3Com V6000 Integrated Branch Communications
AccessLine Communications	www.accessline.com	AccessLine's SmartVoice Service for Business
ACE*COMM	www.acecomm.com	NetPlus for VoIP
ACULAB	www.aculab.com	Prosody X
ADC	www.adc.com	TrueNet Midspan Power over Ethernet Controller
Adomo Inc.	www.adomo.com	Adomo Voice Messaging
ADTRAN, Inc.	www.adtran.com	NetVanta 1224STR PoE
AltiGen Communications, Inc.	www.altigen.com	IP710
Amdocs	www.amdocs.com	Amdocs IP Convergence Solution for IPTV
Aspect Software	www.aspect.com	Aspect Uniphi Suite v6.1
AT&T	www.att.com	AT&T Voice DNA
ATCOM technology co. Ltd.	www.atcom.cn	AU-600
Atreus Systems, Inc.	www.atreus-systems.com	Atreus Multi-Service Provisioning Software
AudioCodes	www.audiocodes.com	Mediant 1000
Avaya	www.avaya.com	Avaya IP Office with Microsoft CRM
BayPackets	www.baypackets.com	Agility Multimedia Ringback Tone
BEA Systems, Inc.	www.bea.com	BEA WebLogic SIP Server
BlueNote Networks	www.bluenotenetworks.com	SessionSuite
Brekeke Software, Inc.	www.brekeke.com	OnDO PBX
Brix Networks	www.brixnet.com	BrixCare Self-Service
Brooktrout Technology	www.brooktrout.com	SnowShore IP Media Server - Software only
Centillium Communications	www.centillium.com	Atlanta Family
Check Point Software Technologies	www.checkpoint.com	Check Point VPN-1 Pro NGX
Cicero Networks	www.ciceronetworks.com	CiceroPhone
Citrix Systems	www.citrix.com	Smart Agent for Citrix Application Gateway
Coaxsys	www.coaxsys.com	TVnet/C
Cognio	www.cognio.com	ISMS Mobile
CommuniGate Systems	www.communigate.com	CommuniGate Pro version 5.0
Convedia	www.convedia.com	Convedia Media Servers
Converged Access, Inc.	www.convergedaccess.com	Converged Access Point (CAP)
Convergin	www.convergin.com	Accolade Platform
CopperCom	www.coppercom.com	The CopperCommander Management System
CosmoCom	www.cosmocom.com	CosmoCall Universe 4.5
CounterPath Solutions, Inc.	www.counterpath.com	X-Lite for Linux
Covad Communications	www.covad.com	Line-Powered Voice Access
CTI²	www.cti2.com	InTouch
Data Connection Limited (DCL)	www.dataconnection.com	DC-SBC
DecisionOne Corporation	www.DesktoptoDialtone.com	Desktop to Dialtone
deltathree	www.deltathree.com	iConnectHere
Ditech Communications	www.ditechcom.com	PACKET VOICE PROCESSOR
Dovado	www.dovado.com	WRG - Wireless Residential Gateway
Embedded Communications Computing, Motorola	www.motorola.com/computers	VoIP Open Application-Enabling Platforms from Moto
Empirix Inc.	www.empirix.com	Hammer DEX
Envox Worldwide	www.envox.com	Envox 6 Communications Development Platform
Esnatech	www.esnatech.com	Telephony Office-LinX version 6.5
Excel Switching Corp.	www.excelswitching.com	Integrated Media Gateway (IMG) 1010
FacetCorp	www.facetcorp.com/facetphone	FacetPhone
Flarion Technologies	www.flarion.com	NETGEAR Mobile Broadband Router 814 (MBR814)
Flextronics Software Systems	www.flextronicssoftware.com	SIP Network Server (SIP NS)
Fonality	www.fonality.com	PBXtra
GigaBeam Corporation	www.gigabeam.com	WiFiber(TM) Wireless Fiber

COMPANY	WEB	PRODUCT
Go2Call.com, Inc.	www.go2call.com	Go2Call Softphone
Grandstream Networks, Inc.	www.grandstream.com	GXP-2000
I.S. Associates, Inc.	www.isassoc.com	TeleCount Billing
IBASE Technology, Inc.	www.technoland.com	PICMG 1.3 Intel Pentium M SHB Express---IB868
iBasis	wwwibasis.com	DirectVoIP Broadband
Inalp Networks AG	www.inalp.com	SmartNode SN4630
Incognito Software Inc.	www.incognito.com	Broadband Command Center Appliance
InfoVista	www.infovista.com	VistaInsight for IPT Telephony
Ingate Systems	www.ingate.com	Ingate Remote SIP Connectivity w SIParator 45/45+
InSciTek Microsystems Inc.	www.allworx.com	Allworx 10x
Intel	www.intel.com/go/iptoday	Intel NetStructure Host Media Processing Software
InterEdge Technologies, LLC	www.inter-edge.com	Dial-Up VoIP Intelligent Telephone Adapter ITA-100
Interoute Communications Ltd	www.interoute.com	ARENA
Interstar Technologies	www.faxserver.com	Interstar XMediusFAX Service Provider (SP) Edition
Inter-Tel, Inc.	www.inter-tel.com	Inter-Tel 5000 Network Communications Solutions
Interwise Inc	www.interwise.com	ECP Connect 6.0
Iotum Corporation	www.iotum.com	Iotum Relevance Engine
Iwatsu Voice Networks	www.iwatsu.com	Enterprise-CS
Juniper Networks	www.juniper.net	E320 Broadband Service Router
Kayote Networks, Inc	www.kayote.com	FronTier - Advanced VoIP Traffic Management System
Kentrox	www.kentrox.com	Q1300 QoS Appliance
Level 3 Communications	www.level3.com	Level(3) E-911 Direct
Lucent Technologies	www.lucent.com	Lucent IMS Service Enhancement Layer
LumenVox	www.lumenvox.com	Speech Driven Assistant
MCI	www.mci.com	Next-Generation VoIP Portfolio for Business
Mediatrix Telecom Inc	www.mediatrix.com	Dial IPCS
Meru Neworks	www.merunetworks.com	Radio Switch family
MetaSwitch	www.metaswitch.com	UC9000 Unified Communication Systems
Minacom Labs, Inc.	www.minacom.com	PowerProbe 30 Service Level Responder & PocketDQ
MIND CTI Ltd.	www.mindcti.com	iPhonEX
Mitel	www.mitel.com	Mitel Navigator
NEC Unified Solutions	www.necunified.com	SV7000 Multiple Purpose System (MPS)
Nero Inc.	www.sippstar.com	Sipps Connect
Net2Phone Inc	www.net2phone.com	VoiceDirector
Netcentrex, Inc	www.netcentrex.net	MyCall v3
Newport Networks	www.newport-networks.com	Newport Networks 1460 Session Border Controller
NewStep Networks	www.newstep.com	Mobile Call Handoff
Nortel	www.nortel.com	Business Communications Manager 50
Occam Networks	www.occamnetworks.com	BLC 6312
Orative	www.orative.com	Orative Enterprise Software
Pactolus Communications Software	www.Pactolus.com	RapidFLEX Call Complete
Pandora Networks	www.pandoranetworks.com	Worksmart
Pangean Technologies	www.pangeantech.com	insta-REACT!
Pannaway Technologies, Inc.	www.pannaway.com	Broadband Access Manager (BAM)
PARADIAL	www.paradial.com	RealTunnel
pbxnsip Inc	www.pbxnsip.com	pbxnsip pbx
Performance Technologies	www.pt.com	NexusWare Linux-based Software Suite
Pingtel Corporation	www.pingtel.com	SIP IP PBX Appliance
Polycom, Inc.	www.polycom.com	Polycom ReadiConvene
PowerDsine	www.powerdsine.com	48-port PoE Midspan
Pronexus	www.pronexus.com	VBSALT 1.2
Psytechnics	www.psytechnics.com	PSI

Go To Table of Contents | Go To Ad Index

COMPANY	WEB	PRODUCT
Pulse Voice Inc.	www.pulsevoice.com	pulsescp
RADVISION	www.radvision.com	Click to Meet for Microsoft Office LCS
RingCentral	www.ringcentral.com	RingCentral
Sansay	www.sansay.com	Sansay SPX
Seawolf Technologies Inc.	www.seawolftech.com	XRainbow Softswitch
SecureLogix	www.securelogix.com	ETM (Enterprise Telephony Management) System
ShoreTel	www.shoretel.com	ShoreTel6
Shunra Software	www.shunra.com	Shunra Virtual Enterprise 3.5
SIPquest	www.sipquest.com	SIPquest Mobile Console
SJ Labs, Inc.	www.sjlabs.com	SJphone
snom technology AG	www.snom.com	snom 360 VoIP Business Telephone
softroute corporation	www.vbuzzer.com	vbuzzer
Spanlink Communications	www.spanlink.com	CentralControl
Sphere Communications Inc.	www.spherecom.com	Sphericall IP PBX
SPIRIT DSP	www.spiritdsp.com/voip.html	TeamSpirit
Sprint North Supply	www.sprintnorthsupply.com	Connection Central
Strix Systems	www.strixsystems.com	Access/One Network Outdoor Wireless System (OWS)
Surf Communication Solutions	www.surf-com.com	SurfRider-812
Switchvox	www.switchvox.com	Switchvox SOHO
SyChip	www.sychip.com	SyVoice VoWLAN 7100
Symmetricom, Inc.	www.ntp-systems.com	SyncServer 250
SyncVoice Communications	www.voicemanagement.com	VXTracker
SysMaster Corporation	www.sysmaster.com	Norfa Lite
Tadiran Telecom	www.tadiranamerica.com	Coral IPx Office
Tekelec	www.tekelec.com	Tekelec Fixed Mobile Convergence (FMC)
Telchemy Incorporated	www.telchemy.com	VQmon/SA-VM IPTV Performance Monitor
Telco Systems	www.telco.com	T-Marc
TeleVoce Inc.	www.televoce.com	TeleVoce-Connected VoIP Technology Platform
TelTel	www.teltel.com	TelTel SVNO Program
Texas Instruments	www.ti.com	TMS320DM642 DSP-based digital media processor
Toshiba America Information Systems, Digital Solutions Division	www.telecom.toshiba.com	Toshiba Video Communication System
Touchstone Technologies, Inc.	www.touchstone-inc.com	WinEyeQ
Transera Communications, Inc.	www.transerainc.com	Seratel
triacore solutions	www.call-plans.com	telextreme
Trinity Convergence	www.trinityconvergence.com	VeriCall Edge 2.0
TriVium Systems, Inc	www.triviumsys.com	FloristCRM
Ubiquity Software	www.ubiquitysoftware.com	Ubiquity Voice Plus
VegaStream	www.vegastream.com	Vega 5048
Veraz Networks, Inc.	www.veraznet.com	I-Gate 40000 EDGE
Visual Networks, Inc.	www.visualnetworks.com	Select VoIP
Vodavi Communications Systems, Inc.	www.vodavi.com	Vodavi XTS-IP and XTSc-IP
VoIP, Inc.	www.voipinc.com	v911
VoIPshield Systems	www.voipshield.com	VoIPaudit
Vonexus Inc.	www.vonexus.com	Enterprise Interaction Center (EIC)
Voxeo Corporation	www.voxeo.com	Voxeo VoipCenter SIP Platform
Whaleback Systems	www.whalebacksystems.com	Whaleback Systems SMB 1500
WildPackets, Inc.	www.wildpackets.com	Omni Distributed Network Analysis Platform
Witness Systems	www.witness.com	Witness Systems Impact 360
XConnect Global Networks Ltd.	www.xconnect.net	XConnect
Xelor Software	www.xelorsoftware.com	XelorRate Service Quality Manager Software
Zultys Technologies	www.zultys.com	WIP 2

insta-REACT!

Pangean Technologies
59 Heritage Drive
Pleasantville, NY 10570
Tel.: 877-472-6432
Web: www.pangeantech.com

Price: Call for pricing.

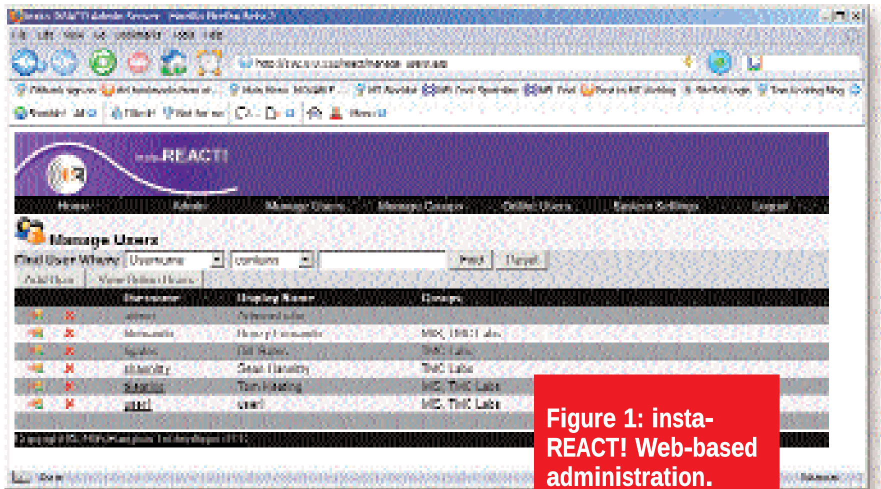

Figure 1: insta-REACT! Web-based administration.

Pangean Technologies' (news - alert) insta-REACT! (Real-Time Enablement for Advanced Communications Technologies) Voice Communications Platform is a VoIP- and software-based communications solution used for instant message and instant voice communications within an enterprise. Unlike most consumer-based IM clients, the insta-REACT! Voice Communications Platform provides secure, standards-based multimedia communications using the SIP protocol over an IP-multicast enabled network. Its major features include presence, voice recording, global intercom, broadcast and missed broadcast, hoot and holler, instant messaging (IM), push to talk, one-to-one page sessions, and dynamic conferencing. Insta-REACT is a great solution for when you need to instantly and spontaneously communicate with several co-workers in your enterprise, especially larger organizations that often have employees in several disparate buildings. Using an IP-based solution such as insta-React can save of the costs of a PSTN-based audio-conferencing bridge.

The insta-REACT Voice Communications Platform is made up of three components, including insta-REACT! Admin Server, insta-REACT! Server, and insta-REACT! Client. The insta-REACT Admin Server is a Web-based management server (Figure 1) that allows an administrator to set up and modify groups and user configurations. The insta-REACT! Server runs on a Windows 2000/2003 server (with SQL Server) and provides media control and end-user management, enabling insta-REACT! Clients to communicate with each other. Finally, the insta-REACT! Client (Figure 2) is a Windows-based SIP endpoint that integrates voice and IM and is the main interface for the users to communicate with each other. The application uses a "Buddy lists" style GUI with user-defined groups and provides presence and availability information. Using the client you can perform such features as "Push to Talk" (individual user or group), Paging, Global Intercom, Instant Messaging and Multicast based group conferencing.

Interestingly, the insta-REACT! solution uses IP multicast technology, which lends itself to much greater scalability than other IP conferencing solutions that require each client to make a connection to each of the other clients or that require a centralized MCU (Multi-point Control Unit) to perform audio transcoding/mixing and then transmit the audio to each client — all of which adds latency to the audio. In fact, the total bandwidth required for multiparty conferences increases as the square of the number of parties involved, leading to scalability problems. Conversely, Pangean's multicast solution simply uses multicast IP technology (address range is from 224.0.0.0 to 239.255.255.255) to transmit the audio across the network, which results in bandwidth and latency savings.

Insta-REACT also uses SIP and SIMPLE to provide instant voice communications on a one-to-one, one-to-many, and many-to-many basis. This enables group voice broadcast, hoot and holler, point to point, and Push-to-Talk utilizing the client's existing IP network. The application also provides for compliance with Quality of Service (QoS) policies enforced in the communication networks.

With its multicast support, insta-REACT! supports an unlimited number of users within a group to transmit and receive audio with no centralized mixer (MCU) required. Pangean's solution utilizes Voice Activity Detection and uses standard UDP/RTP protocols for voice transport and standard codecs G.711 u-Law, G.711-aLaw and GSM 6.10, and they claim that it provides for low latency of less than 40ms.

RATINGS (0–5)
Installation: 5
Documentation: 4.5
Features: 4.75
GUI: 4.75
Overall: A

Conferencing

Pangean claims that their architecture "allows a complete scalable solution that is bandwidth efficient even with a 10,000

Go To Table of Contents | Go To Ad Index

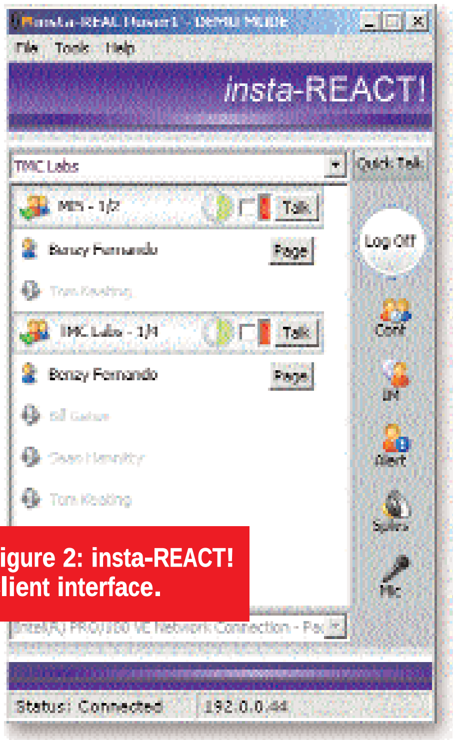

Figure 2: insta-REACT! Client interface.

users per group conference." Their ad hoc conferencing uses a decentralized multicast conference, so any endpoint coming into the dynamic conference must have been invited (INVITE) by another user and can also send an INVITE to other potential participants. The user receives the "INVITE" stating that another user has invited them to a conference. Those alerted endpoints can accept or decline the invite within the specified timeout period all without the need for a centralized MCU.

Hoot and Holler

Users can tune into an unlimited number of Hoots simultaneously, each of which can be configured through the configuration server. Hoot volume controls are available on the hoot buttons to reset the volume of any particular hoot. The user can also send audio onto the hoots that they have talk-back access to (i.e., "enabling" the hoot) by depressing the microphone on the right side of the hoot button. The ability to "listen only" or "talk and listen" is enabled by the Web-based administration screen and is configurable on a per-group/hoot basis.

Instant Messaging

Instant Messaging within insta-REACT supports SIMPLE and has an open, interoperable architecture that allows insta-REACT to adapt other protocols like XMPP and others that may emerge in the future. It also allows users to exchange IM with other users using an instant message client that supports SIMPLE.

We made some test calls. First we made a single user-to-user page by clicking the page (Talk) button, speaking into the microphone and we could then hear the audio almost immediately come out of the remote PC sitting behind us in the lab. This test passed with flying colors so we proceeded to try a group page. We highlighted a group we created with four members. We clicked the Talk button and we could instantly talk to all members within the group. We should mention that you have to press and hold the talk button to speak unless you switch over to the hands-free-mode. In the non-hands-free mode, this allows you to send quick audio messages and the microphone is switched off automatically when you let go of the talk button. For conferencing you can click the Conference button and invite individuals from any group into the dynamic conference. Importantly, the software supports drag-and-drop for adding users to a dynamic conference. We were able to hold a conference with four people with very good sound quality. One thing we noticed when doing a group conference is that it doesn't indicate on the screen who is currently speaking. We would like to see the icon next to the person talking change color or animate in some fashion. All in all, our tests passed with no problems and the audio was very clear with minimal latency.

Two other important features include the ability to see missed pages in the Alerts screen, and the ability to use the product across WANs. You have a couple options for working across a WAN. You can use Microsoft VPN, which will encapsulate the multicast packets. Cisco VPN doesn't allow for encapsulating multicast packets and, according to Cisco, they have no plans to support multicast over VPN since they view it as a security risk. The second option is Pangean's own home-grown solution. Pangean told TMC Labs that, since Cisco VPN (define - news - alert) is very popular, they've decided to devise a product of their own to get around the multicast over WAN links issue. On their roadmap is a "reflector" that can capture multicast packets, encapsulate them into unicast packets, and send to all the other reflectors. This would enable multicast packets to go across WAN links.

Room For Improvement

Although Pangean supports SIMPLE and any IM client that supports SIMPLE, we'd like to see Pangean actively and natively support the major IM networks, such as MSN Messenger, Yahoo! Messenger, and AOL's AIM, at least for IM capabilities (VoIP interoperability would be icing on the cake). This way users can have one less IM client that they need to have running and taking up system resources. We'd also like to see some sort of conference scheduling integration with clients such as Outlook.

We noticed that you cannot private "talk" to one participant from when you are in the Dynamic Conference window. There is a workaround — you can however set up a separate "page" from the main GUI to that user for a private voice talk while still in the conference. We'd also like to suggest a "whisper" feature while on a conference that let's you highlight a participant, press a button, and say something private to just the one conference participant. On a related note, we'd like the ability for the creator of the conference to have moderator privileges that enable the moderator to mute an unruly participant.

Finally, we mentioned to Pangean we'd like to see video; they do have video on their roadmap.

Conclusion

Most businesses use paging/intercom systems using paging groups defined on the phone system, which are, in turn, usually configured by department. The problem with this is that they must be defined by the telecom manager. Often users don't know how to page a specific group and, worst of all, it doesn't lend itself to personalized paging groups. Pangean's IP-based software solution does lend itself to user personalization and is much easier to use than remembering a cryptic sequence of touch-tones on the phone to initiate a group-specific page. The ability to send quick audio messages to specific users or groups, the product's chat capabilities, its excellent scalability due to multicast support, and its easy-to-use interface make this a superb solution for inter-office communications. IT

Go To Table of Contents | Go To Ad Index

IP Communications Ushers in a New Age of Collaboration

For the past quarter-century, companies have acquired and deployed information and telecommunications technology in a never-ending search to improve collaboration between stakeholders while cutting costs, boosting productivity, and improving the user experience. Progress has been painfully slow and difficult. IP communications promises to change that by introducing better, faster, and cheaper ways for employees, suppliers, partners, and customers to work together.

Collaboration tools are nothing new. Teleconferencing, e-mail, workgroup solutions, video, and Web conferencing have been with us for many years. But until now they have been expensive to acquire, deploy, and manage, and difficult, if not impossible, to integrate. By converging digital voice and data traffic onto a single physical network, IP communications enables companies to deploy a robust infrastructure that will support a wide range of integrated and cost-effective collaboration solutions, including: voice, data, and Web conferencing; unified messaging; call center systems; even IP gateways that can be used as universal translators to enable incompatible RF systems to communicate with each other — an important collaboration tool for first responders to disasters like Hurricane Katrina or the terrorist attacks on 9/11.

Unfortunately, because many companies acquire IP communications systems as a cost saving measure, they sometimes overlook the ways in which IP communications can be used to transform their business operations. Companies that marry a collaboration strategy to their IP infrastructure investment will gain the most benefit from this transformative technology.

There are many good reasons for making the move to IP Communications. Maybe you are moving, growing, merging, or consolidating offices. Maybe your old PBX maintenance costs were too high, important features were lacking, or you decided to include IP-based voice into your next generation IP-based data network. Whatever the reason, you are in good company. Most experts agree that IP Communications has crossed the chasm from early adopters to mainstream technology. According to the Enterprise Communications Association, IP Station shipments are growing at an annual rate of 36 percent and will top 4.5 million in 2005. Conversely, TDM (define - news - alert) station shipments are shrinking by almost seven percent per year. Synergy Research Group reported a 30.6 percent year-over-year growth for Enterprise IP Communications in the third quarter. According to Nemertes Research, 71 percent of IT professionals surveyed have deployed IP Communications in some fashion.

Now is the time to focus on maximizing your investment. Expect more than just a phone system replacement. As you make the transition from TDM to IP, you must distinguish between *Infrastructure* and *Applications*. While designing and implementing the proper infrastructure for IP Communications is critical to success, it is the implementation of IP Communications Applications that will drive the productivity and enhanced customer experiences intrinsic in the potential of this new communications paradigm. By deeply integrating IP Communications

By David Hart

applications into your environment you can reap a truly significant return on investment that goes far beyond mere cost cutting.

Building Blocks: Laying the Foundation

Like any new technology, the IP Communications infrastructure needs to be engineered to be robust and resilient. Beyond the IP Phone, the basic building blocks for an IP Communications system include:

- Next Generation Local-Area Network and Wide-Area Network;
- Server-based Call Control;
- Distributed Trunking Gateways;
- Management and Monitoring Tools.

Next Generation Local-Area Network

Since IP will now be the transport for your voice traffic you will be reliant on the LAN to provide the "five nines" reliability your users have come to expect from their phone system. This means you need to be sure that your network is both redundant and intelligent. An intelligent network can use Quality of Service (QoS) techniques to make voice packets and prioritize them

It's not just about cheaper phone calls — it's about transforming the communications experience.

throughout the transit of your network. An intelligent network also uses sophisticated routing techniques to ensure reliability in the event of link or device failure. Also, be sure to have standards-based IEEE 802.11af Power over Ethernet for each end-station LAN port in your network. Finally, be certain to include tools to manage your network infrastructure. Beyond the rudimentary tools that are provided by the LAN/WAN hardware manufacturers, there are many third-party companies that offer excellent tools to provision, instrument, isolate faults, manage, and report on your network infrastructure.

Server-Based Call Control

After much debate, the best practices architecture for deployment of IP Communications has been settled. The fundamental device in this design is centralized call processing on redundant server or appliance platforms. These devices make the routing decisions for all call processing. These centralized call processing platforms can be integrated with lower-cost survivable call processing engines to service remote/SOHO locations in your enterprise. Further, these devices can be integrated with other major hub sites to provide for a global dial plan. Also, these devices provide advanced services like Call Detail Reporting and CTI application integration.

Distributed Trunking Gateways

Though enterprise communications is rapidly migrating to IP, the Public Switched Telephone Network (PSTN) remains the transport for "off net" calls. Trunking Gateways are the devices that provide termination and transcoding services to interconnect the PSTN to the private IP Communications network. Because IP is a routable protocol, location limitations are eliminated in IP Communications networks. Therefore, you should distribute PSTN Trunking Gateways throughout your LAN and WAN to provide high availability, flash traffic control, and optimized toll bypass services to each endpoint in the network, regardless of location on the network.

Management, Monitoring and Ongoing Support

In most every organization, dial tone is the most critical application in their environment. Now that you've decided to migrate from your TDM network and put voice on your IP network, you need to be certain that you have the appropriate tools to support and monitor your environment. Start by understanding your requirements. Besides uptime, what do your employees and clients expect from your IP-based voice applications? What does "voice quality" mean? What is the recovery time objective in the event of an outage? Once you have identified your requirements, map them to the service level agreement (SLA) or service metrics that define these requirements. Once that mapping is complete, survey the tools offered by your primary IP Communications vendor, as well as third-party providers. These tools will prove critical in implementing your support strategy. Finally, define service management processes within your organization. These processes define the roles and responsibilities required to measure, analyze and react to management data as it is presented.

Turning IP Telephony into IP Communications

Once you've deployed the infrastructure required to replace your TDM phone system with IP Communications, you are ready transform the way your organization communicates with your clients, your partners, and your colleagues. IP and IP-based applications enable one-to-one, one-to-many, and many-to-many communications in a multimodal fashion. Beyond voice, an IP-enabled network removes the physical and geographic limitations of TDM and affords many new forms of communication. Some examples include:

- Unified Messaging
- Next Generation Contact Centers
- Presence/IM
- Rich Media Conferencing

Unified Messaging

Unified Messaging is the capability to integrate your Voice and Electronic Mail into a single message store for retrieval and management. While available on a limited basis in the TDM world, Unified Messaging has become the most common application deployed as part of the transition to IP Communications. In the Unified Messaging environment, a voicemail is simply another object in the message store. It can be retrieved, forwarded, deleted, modified, saved, backed-up, and restored in the same manner that any mail object would be handled. Using IP as a transport, end users no longer need to "call in for messages." Voicemail can be presented as an attachment to an e-mail available from PC or PDA with access to the enterprise mail environment. Some UM solutions provide additional services, such as text to speech engines, which allow for e-mails to be read to a user over an IP or PSTN voice call.

Next Generation Contact Centers

Contact center technologies, including Automated Call Distribution (ACD), Integrated Voice Recognition (IVR), and Voice Response Units (VRU), are critical for business operations today. They create access points for clients to be serviced and reduce the friction that can come with client service communications. They also have tremendous value in providing historical data as to how an enterprise communicates with its clients and how well it manages customer service representatives. These technologies have been available in a TDM world for some time. However, they have typically been stove-pipe solutions with little or no integration with the rest of the enterprise communications environment. They are further hampered by limited capabilities to integrate with other enterprise applications such as customer relationship management (CRM) or enterprise resource planning (ERP) systems. Again, by breaking the rigid physical, geographic, and proprietary limitations of TDM, IP Communications has opened up new avenues to improve client service and organizational efficiencies with next generation contact centers.

With IP Communications, voice becomes an application that can be integrated with most any enterprise application. This produces more and smoother avenues (click to chat, click for call back, short message service (SMS), Web cam/video) for your clients to communicate with you as well as significantly improving the quality of the client experience by putting vast amounts of enterprise data at the fingertips of your service agents.

Finally, significant cost savings can be achieved. By eliminating stovepipe TDM systems, you have lower operational and maintenance costs. The dynamic nature of IP allows for more efficient use of the existing communications network, which lowers telecommunications costs. Agent physical location becomes almost irrelevant so smaller pools of agents spread out geographically (onshore versus offshore) to support more client interaction. Integration of presence technologies allows for more efficient use of agent resources. Integration with enterprise data allows for shorter call times and faster wrap-up, again shaving agent costs. Finally, a consistently high-quality client experience will reduce client churn, dramatically reducing client acquisition costs.

Presence and Instant Messaging

The integration of voice, video, and data over IP has significantly streamlined communications and provided multiple channels for exchange. That said, communication still is not entirely synchronous. E-mail and voicemail, in particular, tend to be non-real-time exchanges. The advent of Instant Messaging brings with it the concept of Presence. Presence allows an individual to advertise his or her availability to communicate and which communication modes are available and/or preferred. Today, all of the major IP Communications vendors are driving the capability of presence into their solutions. The most common emerging standard seems is based on the IETF draft standard SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE). As vendors continue to ship next-generation products, expect SIMPLE-based Presence technology to be a major feature of their end point solution sets.

Rich Media Conferencing

Integration of voice, video, Web, and enterprise data is the ultimate goal of a converged network. Once we have built the infrastructure to support IP-based communication, we can begin to integrate previously stovepiped audio, video, and Web-based data applications into a common user experience. By providing a means for both scheduled and ad hoc conferences where users can share information in a multimodal fashion, we can boost productivity, speed up business processes, provide higher quality educational experiences, and improve client interaction.

This means more than simply reducing travel costs and sharing documents. By leveraging our IP network, we can provide for real-time communications and collaboration between multiple parties, regardless of geographical location. Teams share ideas and data in a virtual conference center where documents can be shared and mutually edited. A virtual white board can create a knowledge sharing environment for all attendees. Instant Messaging technology allows for off-line, person-to-person exchange. Integration of IP-based video makes the communication as close to an in-person experience as possible.

The Promise of IP Communications

It's not just about cheaper phone calls — it's about transforming the communications experience. IP Communications means eliminating the redundant, stovepipe approach of separate voice and data networks.

IP Communications enables customers to invest and grow in the part of the network they plan to keep (IP), not the part they plan to phase out (TDM). By consolidating call control, distributed gateways, and new management and support on a next-generation IP network, we can build a robust platform for advanced communication and collaboration.

Applications like Unified Messaging, Contact Centers, Presence IM, and rich-media capabilities are key cost savings and productivity enablers for converged IP networks. Add these applications to give your employees and clients the true benefit of IP Communications. Make it more than just a phone. IT

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Next Generation Messaging:

Understanding the Market Drivers and Managing in a Changing Environment

Faced with globalization, a shifting marketplace, and regulatory and competitive pressures, today's telecommunication carriers must continually differentiate products and services to increase the average revenue per user (ARPU) and actual margin per user (AMPU), all while improving customer retention rates. Next generation messaging (NGM) services offer carriers an opportunity to dramatically differentiate themselves from competitors and to target offerings toward specific market sectors. However, as traditional voice mail systems evolve into IP-based next generation messaging systems, carriers also experience a tremendous amount of digital data growth that must be managed and stored for customers. As a result, carriers will need to dramatically enhance their information management and storage technology to support the system's capabilities while improving the efficiency of customer-facing business activities, such as billing and customer service.

Trends Towards Next Generation Messaging

The next generation messaging evolution will assimilate all other message types to enable users to send a compound message without really knowing or caring about the capabilities of the recipient. So, when one user sends a message to another, the message will be 'groomed' by the network so it can be delivered to a device irrespective of the client type. The network may also be able to add value elements, such as predictions based on past behaviors, current location, or even link to calendar sensitive issues, such as travel schedules or holidays. In that scenario, the network is pivotal in making decisions concerning format delivery, timing, and all other message parameters. In other words, the network continues to add value.

However, this presents some challenges. The growing trend of implementing IP-based communication at the core of the network will affect functions and operations at the edge of the network. Consequently, all message format and presentation could be handled on a peer-to-peer 'thick-client' basis, ultimately eroding the need for network-based components and gateways. In addition, the advent of the IP network breaks down boundaries between what is mobile and what is fixed line.

The carrier then has to manage the transition towards convergence while still adding value above and beyond the capabilities of a client-based communications service.

The ability of the network, particularly the mobile network, to add presence, location, and preference information, while offering a guaranteed storage location for sensitive and valued data, should be central in the network-centric value proposition.

NGM and Content Storage

In addition to evolving messaging technologies, telecom operators will require other technology enhancements to drive revenue. The explosive growth of customer digital media in the form of photos and music files will be a potential area of revenue generation, both as a value-added service, as well as with additional bandwidth utilization.

For instance, offering online photo albums and music facilities requires back-up and storage as well as other capabilities such as sharing, printing, jukebox, and more. It is unlikely that

Go To Table of Contents | Go To Ad Index

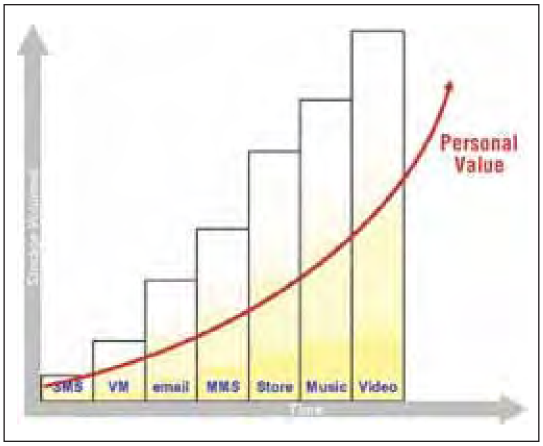
By Steve Toft

NGM content stores and 'my photo,' 'my music,' and 'my video' will exist in isolation. Rather, a customer receiving a complex message may well elect to store components of the message in different ways, moving a photo to the online album, for example. Similarly, the online stores will offer the user a rich choice of data to compose and enhance their messages.

Challenges

Today's IP-based voice mail systems typically plan for one to ten MB per user. This is a function of the volume and relatively transient nature of the data with a typical voice mail being held for three days. However, a trend towards rich data will result in both an increase in data volumes and likely, an increase in the duration for which messages are held. Future messaging systems will typically have a requirement of 10–100MB per user. However, as noted above, adding the online photo/music/video store to the environment could well drive storage requirements to the multi-GB level. This will likely increase memory capabilities of the network storage devices used to access the messaging system.

Figure 1.



While the ability to handle data volume alone is important in the storage layer, the input/output performance of the system is also a critical factor. This can be seen in the requirements for today's voice mail systems. Clearly, as the volume of data increases, this factor becomes even more important, particularly when the 'real-time' nature of messaging is considered.

In summary, today's messaging systems are transactional in their nature. However, the trend is towards a system with both transactional and bulk storage characteristics. Designing a 'right cost' architecture to support such systems will be critical to their success.

Information Management

The volume of data to be stored is only one element when considering the storage and content management requirements for next-generation messaging.

The message illustrated in Figure 1 is clear. As the carrier progresses to handling more and more personally orientated data, not only do the volumes increase, but the value attached to data also increases significantly. Carriers will have to guarantee this data will not be lost and can be retrieved in the case of disaster, and give a concrete guarantee covering the integrity of the information stored on behalf of their customers.

An information lifecycle management (ILM) strategy utilizing tiered storage and information management will securely and cost-efficiently manage the storage of information while offering high input/output performance. To intelligently manage information through its lifecycle, carriers will establish processes and procedures for moving data to the appropriate tier of storage based on the changing value of information and archive less pertinent data more cost effectively. For example, the complex messages proposed above will comprise a number of components including video, text, and voice. It may be possible to 'best guess' which of these elements are of most value to the customer. A received message could be treated as an object by the messaging system and its components held in different storage areas depending on its characteristics.

Voice would need to be held on high-performance storage to ensure it's availability for playback, whereas images may be placed on in 'near line' mode in anticipation of archiving and/or to guarantee integrity. It may well be that each component is then subject to separate information management rules throughout its lifecycle. In another variation, it may well be that all messages are managed according to standard lifecycle management rules unless the customer intervenes. For example, the customer may wish to add a message containing images to their online photo album. Or, it may well be possible to offer the user options against discrete components of the message. As this would have priority over the 'automatic' information management rules, some form of charging could be appropriate.

What To Plan For In A Next Generation Messaging System

There are several distinct value propositions that carriers should be pursuing:

• Partner with a vendor who offers superior price/performance characteristic when implementing the tiered storage necessary to support next generation messaging. Don't look only at acquisition costs, but consider the total cost of ownership of those systems, including maintenance and staffing.

The explosive growth of customer digital media in the form of photos and music files will be a potential area of revenue generation.

• Seek a partner who enables NGM by providing data storage security allowing carriers to gain the trust of the customer in handling data of high personal value.

• Find a partner whose applications and infrastructure are able to utilize a superset of the telecom's storage layer functionality.

• Select a partner who can assist the carrier in conducting a cost-based analysis of the complete NGM solution.

• Look for market leading ability to implement tiered storage layers that enable cost efficiency in message handling to drive down the system TCO through smart information handling techniques and while offering users new storage options (automated or manual).

As more carriers embrace next generation messaging as a critical service offering, they will need to brace their IT infrastructure to handle the increased data growth the service will generate. Each of the above value propositions has a direct cost impact on message transportation, storage and content manipulation. In an automated environment, determining the right architecture and determining the storage cost base to feed into the business plan will be complex, but could maximize the performance of the next generation messaging architecture, as well as improve the efficiency of customer-facing business activities. IT

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Multi-Point Voice Conferencing Will Boost Online Communities

About 30 percent of human active time is spent talking. Before making important decisions people seek advice, and they like being consulted, sharing experiences and discussing outcomes.

In many cases people prefer talking live (i.e., while physically being in the same place). Making deals, brain storming, debating about problems and opportunities, progress tracking: all of these are ongoing routines. Private group communications — meetings of family members who live separately, or holding a party with friends — are as important as they are useful. Several people talking through an issue leads to better decisions, eventually improving one's life and business. It is widely held, that people who are part of the decision making process have already committed themselves to the decision. Group communications are standard, common and are a must for enterprise performance.

The introduction of the telephone has been instrumental in bringing people together; however, in most cases it allows only two people to speak at a time. Regardless, this breakthrough in voice communication technology often eliminated the need for face to face meetings. Distant talks became common — easy, cheap, comfortable. Up to now, phone conferencing in distributed groups was used in enterprises only. With the emergence of the Internet, typed instant messaging and online chats have offered attractive alternatives to live conversations or e-mails. The 'chat-room' option for groups has become the nucleus for online communities to grow around.

Is networking adequately helping this urge? How is voice conferencing supported today? How will this technology rub off on communications and global markets?

A quest for real-time voice collaboration

The Internet has revolutionized both text messaging and voice communication. Voice over IP(define - news - alert) is competing and winning against PSTN legacy and mobile in terms of immediacy, price, and sound quality. Initially, IP networks — designed for pure data transfer — were hard pressed to ensure high sound quality, but gradually began to provide an acceptable audio experience. This primarily concerns bilateral connections. Up to now, multi-point (i.e., involving multiple participants) VoIP conferencing was not famous for its high quality, due to intrinsic technical complexity. The market is still anticipating an optimal conference solution that will combine phone call simplicity and 'chat-room' capability, while benefiting the emerging communities as widely affordable. It is a foregone conclusion that this solution will be IP-based, and should provide, in addition to basic voice functionality, a variety of value added services.

The next approach to networking became known as Real-Time Collaboration. RTC is software that integrates a set of online collaboration and conferencing applications: IP telephony, text messaging/chat, presence management, document sharing, media streaming, and Web conferenc-

Go To Table of Contents | Go To Ad Index

By Andrew Sviridenko

ing. At a new level of productivity and cost-effectiveness, RTC helps both enterprises and consumers discard their outdated communication equipment and applications: stand-alone telephones, e-mail/IM/chat/VoIP clients, conferencing facilities, together with their SMS/Voice mailing/logging. RTC integrates all communication routes, radically improving efficiency and convenience.

According to Merrill Lynch, VoIP system sales dwarfed those of traditional voice systems for the 12-month period ending in June 2005. META Group research indicates that the RTC market will reach at least $10B by 2008, growing exponentially with increasing demand for, and application deployments in, IM and IP telephony, Web and video conferencing, and other value-added applications; with maturity of streaming media; with wireless and broadband bundling voice, data, and mobile. Analysts conservatively estimate a $4.5B combined revenue for the entire market of RTC-related components. In 2004, the total return of sales in U.S. submarkets reached approximately $550M for Web conferencing and $275M for enterprise IM, while well-established markets such as audio conferencing and video conferencing accounted for $3B and $600M, respectively. Reportedly, audio and video conferencing segments are growing at 20 percent annually. Web Conferencing is expected to soar at 40 percent annually (*Collaboration Strategies*). Is there any product ready to win the RTC voice market?

Multi-Point VoIP: The Core of Industrial RTC

e-phone stands for VoIP basic technology in IBM Lotus Notes and IBM Workplace. In Microsoft's worldview, Live Communications Server will be the centerpiece for instant messaging, voice and Web conferencing. Oracle Collaboration Suite integrated real-time collaboration and communication platform targeted at enterprises. Similarly, Macromedia's initial release of Flash-based Breeze for Web conferencing missed a useful multi-point VoIP functionality, which their newest product Breeze 5 includes.

These proprietary solutions aside, hosted Web conferencing services apparently missed the multi-point feature in their core as well. Service providers like WebEx still do not provide true Multi-point VoIP services.

In most cases, multi-point VoIP conferencing is based on a dedicated equipment set (Polycom's, for instance), either leased or purchased. None of their many options address the basic needs. They are of limited use in business and useless and expensive for consumers. Furthermore, compared to integrated software solutions, the marketed hardware solutions fall short both in voice quality and performance: several hundred channels per unit are available, compared to thousands of channels per PC, and with PC's extra economic benefits.

The impossibility of instant group conferencing using any available desktop debases the very idea of real-time collaboration and communication. That's why the multi-point feature is becoming the key factor of conferencing leadership.

The impact of multi-point VoIP is not limited to the performance of RTC-enabled business. Multi-point voice conferencing platform is a perfect tool for boosting online communities of like-minded people. Voice communities will soon become an integral part of our lives, and a valuable evidence of social integrity.

Multi-point VoIP challenges

The VoIP challenges in multi-point versus point-to-point, (PP) regarding both sound quality enhancement and voice data streaming optimization, are of completely different orders of complexity. The technological competence required should exceed that necessary for IP hardware OEMs, IM, and/or in the softphone industry.

In case of many conferees talking concurrently through a non-multi-point application, voice quality is generally poor. Just try speaking with a couple or more people at once through any softphone like Skype. In fact, you won't be able to invite more than five participants. The voice flow will be corrupted by gaps and distortions as the system is picking up the loudest party while damping, clipping, or muting the rest, who will be eagerly trying to interrupt the leader. The result cannot even be called a conversation. If several people start arguing, such communication will lose whatever sense it had before, as they just won't be able to catch everything said.

In case of multi-point, the number of participants is only limited by the number of people that really need to be involved. Everything said is heard. Speech is clear and the voice stream is smooth, so the audio is better than that of PSTN, and it is safely preserved for most of conventional hands-free and headset-free modes.

Compared to PP, ensuring excellent voice quality for multi-point conferencing is not that simple. The voice engine must:

- cancel acoustic echo repeatedly spawned by remote participants talking simultaneously;
- suppress background noise to keep speech levels sufficiently high for comfortable conversation while preventing gaps and clipping;
- support the interaction of people who use different channel capacities coming from IP, wireless or PSTN networks, and
- eliminate the possibility of voice data loss and sources of misunderstanding in order to prevent expensive mistakes.

Multi-point voice conferencing engine

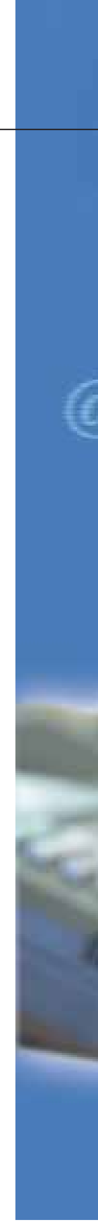
Crystal-clear sound is provided at a cost of the lowest possible resource consumption. Is this truly feasible? With an optimized multi-point solution which employs tandem-free mixing, the server is used dozens (or hundreds) of times more efficiently, when compared to direct transcoding applied in hardware conferencing. This allows increasing the supportable number of conferences at least tenfold, while each of those conferences will be able to unite dozens of participants.

In case of multi-point, the number of participants is only limited by the number of people that really need to be involved.

Unlike conventional solutions, intellectual adaptive voice coding ensures the transmission of speech flow to everyone participating in the conference, in accordance with the bandwidth available. Everyone fully understands what is being said, but those having wider channels also enjoy improved audio, regardless of the limitations of modest channels. As a core software solution for VoIP-enabled real-time collaboration, the multi-point voice engine is highly beneficial to the consumers, since it enables to decrease the TCO (Total Cost of Ownership) through the elimination of conventional phone calls and all PSTN hardware expenses. The provided voice quality is twice as rich as narrowband PSTN. It also ensures the great convenience of a single tool for audio, video, data communications and cross-platform availability for PC, PDA, and mobile platforms.

Further penetration of really high-quality VoIP multi-point conferencing capabilities to everyday life will provide the new level of human convenience, efficiency, and productivity. It will ensure communications that are much easier and less expensive, making obsolete multiple communication devices and applications together with their separate histories and archives. Real-time multi-point VoIP conferencing will integrate all communications and all the stories. IT

Andrew Sviridenko is the CEO of SPIRIT DSP (news - alert). For more information, please visit the company online at http://www.spiritDSP.com/voip.



Triple Play Myths and Magic

IP-based triple play deployments enjoyed a breakout year in 2005, both in terms of hype and reality. Most vendors — and certainly any member of the IPlay3 Consortium — will agree it was exciting to see the concept take flight and even morph into unexpected, but interesting variants. The entire industry woke up and it was encouraging to see investments in equipment and service deployments back up the hype — most of the time. As the triple play industry grows a year older — perhaps into its "adolescent years" — it is most important to separate fact from fiction, magic from myth, behind the forces driving triple play.

The Major Benefit of Triple Play is Getting One Converged Bill

Myth: The point has been previously made, but it warrants restatement: Triple play must be more than just a converged bill, only capturing a small portion of the total value. We have seen early forays into converged services — various service providers, using separate equipment and networks, offering customers the option of subscribing to all three services (voice/video/data) from them. But by now, most service providers have realized that this "1+1+1" approach is shortsighted, often giving customers the wrong impression. The customer receives a 20-page billing statement, and perhaps $5 off his total bill. What's more, the service set-up and support are no easier, and sometimes even more convoluted and frustrating, leaving subscribers scratching their heads and wondering where the value is.

Quad Play Is Something More Than Triple Play

Myth: This myth derives from a basic human desire to show up your neighbor. It is reminiscent of the cult classic movie, *This Is Spinal Tap*, where brain-fried band member Nigel Tufnel tries to convince us that Tap's Marshall amps could extend beyond the standard high volume mark of 10. Says Nigel: "You see, most blokes will be playing at 10. Where can you go from there? Nowhere. But ours goes to eleven. It's one louder, isn't it?"

Quad play refers to wireless/mobile access to services. But wireless is just another access technology, as are DSL, Fiber, or Cable technologies. It is understandable to think of it as a separate service because the addition of wireless access and mobility applications certainly improves the overall business model (and complexity) of triple play services. But a true definition of triple play is a service mix based on the trio of voice, video, and data — regardless of the access technology. Each of these services must be looked at for how they can be uniquely tailored and integrated across multiple access devices (e.g., phone, TV, PC, mobile handset). Whereas wireless and mobility add a range of possibilities, "quad play" is a misnomer if it is meant to introduce something more than the three platform services that triple play has to offer. What would be next, "Quintuple Play"? When it gets to "Undecuple Play," start laughing. Perhaps a reasonable compromise is simply "Multi-Play."

There is a Killer App for Triple Play

Myth and Magic: Fair enough, this is, in fact, a trick question. There is not a so-called killer application for triple play, at least not in conventional terms, which usually involves one single service as the real draw and catalyst for rapid adoption. Obviously, in this sense it is a myth. That said, the real magic comes from the service mix itself. Paolo Tavazanni, head of advanced services at Italian service provider FastWeb, explained it at last year's Triple Play Symposium. He noted that FastWeb, which has grown to more than 800,000 lines, began with a fairly modest "all you can eat" service package of voice, video,

Go To Table of Contents | Go To Ad Index

By Brian Mahony

and data. As the firm grew, they kept finding more targeted versions of new and existing services to meet the needs of a specific vertical. Some of these service bundle integrated voice, video, and data in unique ways. Others provided greater choice for a specific application. For example, FastWeb rolled out a "pay-per-use" service for Internet access where customers could pay by the minute. While it seems odd to offer pay-per-use over an always available high-speed DSL or fiber connection, FastWeb found there was a significant minority (about 10%) who did not use the Internet very often and did not like the idea of paying a monthly flat fee (even though FastWeb's analysis found many of them rang up charges more than the flat monthly fee would have cost them). The lesson is that a flexible service mix combining a range of all-you-can-eat and à la carte options is the real killer application to meet the needs of many different subscriber demographics.

Triple Play Will Usher in a New Era of Service Creation

Magic: The major milestone for any triple play provider is delivering the services via a converged IP network. For many of them, this represents their first time building a converged network; it is a watershed of sorts for them as they realize the potential for service creation goes well beyond basic voice, video, and data services. In fact, once the IP network is in place (regardless of access technology) it becomes much easier to deploy other advanced IP services (e.g., gaming, presence applications, e-learning, e-medicine, music, collaboration, security, monitoring, meter reading, dating). All that is required is new service logic and, perhaps, some new CPE (e.g., video cameras for home monitoring).

The Triple Play User Experience Transcends the Services Themselves

Magic: This is the core message about the value of triple play — not only that the converged services will save subscribers money, but also that the services themselves may be exceeded by the way they are experienced by the subscriber. Let's say you have subscribed to a converged triple play package of high-speed internet, IPTV/VoD, and VoIP. (define - news - alert) This is good, and you are probably already the envy of your neighbors. But let's also look at a scenario for how you might use them. You are settling in for a Friday night, and have selected a VoD movie to watch with your family. Grandma calls, her caller

The provider that masters video calling may well win the triple play wars.

ID is displayed on the screen, and with a click of the remote, you initiate a video calling session so Grandma can say good night to the kids before going to bed. Later an old college buddy calls. Again with one click of the remote, you send Larry to voicemail. Still later, when you take a break from the movie to make popcorn, you retrieve Larry's voicemail and hear it through your TV's speaker system. Then, through a similar look and feel on your laptop interface, you program all the phones not to ring but to go directly into voicemail so as not to wake the baby. You also set up your call forwarding rules for the next day, so that all calls automatically get routed to your ski house where you will spend the weekend. Not only do you get to enjoy the services, but the way you interact with them can enhance your overall communications and entertainment and, indeed, your living experience.

The Future of Triple Play Will Be Built on an IMS Architecture

Magic: All indications are that IMS is the converged architecture of the future. Based largely on the SIP (define - news - alert) protocol coming out of the softswitch world, combined with mobile concepts like HSS/HLR intelligence, IMS is here to stay. One of the key value points is that IMS defines a layered architecture that allows applications to be created independently of the underlying access technology and with little knowledge of the network equipment. Since a major driver and benefit of triple play deployments is service creation, IMS will play a significant role in motivating both third-party ISVs and service providers to innovate ways to combine and build upon the basic voice/video/data building blocks triple play has to offer.

Cable Company MSOs Have a Huge Lead in the Triple Play Wars

Myth: It defies conventional wisdom, but when you peel the onion back a little, you will see a more level playing field between the cable MSOs and their older Telco siblings. First, cable companies do not exactly have the reputation of being purveyors of premium customer service. Indeed, some have even "earned" years of enmity from subscribers who do not easily forgive (or forget) past grievances.

Second, Telcos are now leaner and better equipped to fight the nascent battle for subscribers' triple play loyalties. Voice revenues have declined significantly and have forced telcos to trim down and become more aggressive, while video service revenues have stayed comparatively fat. This means video revenues have a way to go before becoming fully commoditized due to competitive pressures, and cable companies' core business will be pinched more than the telcos'.

Finally, looking at the underlying complexity of the technologies themselves, Mr. Tavazanni of FastWeb fame reminds us that the telephony portion of triple play becomes the true measure of service quality, because the ears are less forgiving than the eyes. Furthermore, the future of triple play seems to lie in interactive services. Telcos have become adept at management of massive real-time conversations, while it could be argued that video broadcast, or even VoD, services are inherently less complex. Whereas telephone calls and signaling require constant management and vigilance of the quality and state of the "session," TV channels, once set up, are mostly stateless push services (or simple pull services in the case of VoD).

The best merger of the two worlds is video calling. In one sense, video calling requires a mastery of video streaming codecs and other technologies with which the cable companies are quite familiar. On the other, video calling requires an entirely interactive communications environment, where sight and sound need to be blended in a precise way, and adjunct non-real-time applications, such as video voicemail, need to be integrated seamlessly as part of the experience. The provider that masters video calling may well win the triple play wars.

Triple Play Service "Stickiness" Can Build Customer Loyalty

Magic and Myth: By now, it should be clear that triple play will create quite a spark. However, defined, deployed, and supported improperly, it could also cause disaster for service providers that do not invest wisely. For sure, service providers will be able to build service revenues and customer loyalty in ways never before possible, but the stakes are also considerably higher — the "triple threat." Now, if a service provider fails in one area, perhaps not providing a quality VoD product, the customer could take ALL his triple play service business to a provider that can provide a more compelling service mix or better customer service. The balance of power still rests with the incumbents, all of whom are trying to lock up the loyalty of as many customers as possible before the real head-to-head triple play competition heats up. But the definition of what constitutes sufficient service "stickiness" will evolve over time as even more innovative services are offered by the competition (perhaps driven by IMS) and customers are given a greater range of broadband upgrade options (DSL, fiber, fixed/mobile wireless). This is where magic and myth will give way to legend — that story is still being written. IT

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Go To Table of Contents | Go To Ad Index

QoS: The Nuts & Bolts of Performance & Profit

In one regard, "Quality of Service" mandates for voice and multimedia over IP services are nothing new: failure to meet them can quickly result in lost customers and revenues. Delivering profitable multimedia services means giving customers exactly what they expect — and pay for — every time they use the service.

In another regard, sustaining high-quality services represents a whole new arena for service providers where they are confounded by issues unique to IP infrastructures every day as subscriber numbers grow. Fraud, evolving regulations, and the idiosyncrasies of IP itself all pose challenges that must be met in order for services to keep pace with the performance of the PSTN (define - news - alert) and improve upon its profitability.

Outwardly, "Quality of Services" simply means delivering the consistently high levels of performance end users routinely expect. Internally, sustaining QoS is not so simple. Service providers must manage many different facets of the communications path as a call progresses from the end user through the service provider's network and potentially into a third-party network. In IP-based multimedia services, quality must be achieved and ensured using broadband networks that were never intended to handle real-time traffic with such predictable precision.

A closer look at today's networks reveals numerous factors influencing the successful delivery of profitable high-quality services. To start with, two unique facets of IP multimedia infrastructures must be countered: the bursty nature of data networks and the elastic nature of IP connections.

Broadband Access: Burst Versus Best

The market for broadband voice and multimedia services is growing rapidly. iLocus estimates that by mid-2005 there were 14.5 million consumer voice over broadband subscribers, and that the number of subscribers in the top 10 countries alone will exceed 130 million by 2009.

But despite an apparent abundance of bandwidth, most access networks were designed to deliver the required service levels to consumers based on the bursty nature of data services. Networks could be overbooked and still deliver the required service so that, in the case of downloading an MP3 or video clip, for example, a one- or two-second gap in the stream would go unnoticed. (However, the same one-second gap in a voice stream would be fairly annoying.)

In other words, although a connection rated at 1Mbps may seem to be more than sufficient in making a voice call, the service provider could face the thorny problem of all or most subscribers trying to make multimedia calls over the access network at the same time. Without adequate planning and precautions to ensure QoS, such an occurrence could compromise quality, not to mention SLA compliance and profits, fatally.

IP is Elastic (Voice is Not)

The second factor in the QoS equation is the elastic nature of IP connections. In the traditional telephony world, if you have a connection capable of supporting 32 concurrent calls, it does exactly that — the thirty-third call will not fit and the caller will get a busy tone. In the world of IP, because each connection is just a stream of packets, the thirty-third call will be admitted too — to the detriment of all 33 calls. Each IP stream will suffer from some delay and dropped packets resulting in a reduction of voice quality. Blocking a call may seem harsh but the alternative is 33 unhappy customers!

Enter QoS: Imposing Order on Chaos

Combining the inherent overbooking in the access network with the tendency

By Mike Wilkinson

of IP streams to compete for available bandwidth sounds like a recipe for disaster. In response, service providers and infrastructure suppliers continue to evolve comprehensive strategies and innovative mechanisms for imposing order on would-be chaos.

Policing the Access Network

The access network will not police itself, so it is necessary to impose some control on who can do what and when. Doing so requires an understanding of the relationships between the different parts of a multimedia call. Each call is made up of a signaling stream and a media stream. When most network devices encounter these streams they are blissfully unaware that there is a relationship between them. This is where Session Border Controllers (SBCs) come in. SBCs, (define - news - alert) as their name implies, are session aware, meaning they understand which media stream belongs to which signaling stream.

This in turn means that, unlike most network devices, SBCs exert control on the traffic at the level of an individual session when required. First developed to provide secure NAT traversal, SBCs have evolved to encompass such a broad range of security and quality management functions that they are rapidly becoming indispensable in creating secure, manageable multimedia networks.

For starters, SBCs now furnish the nuts and bolts of quality management in the access network:

• *Session Admission Control:* understanding the topology of the access network and the traffic flowing through it.

• *Controlling parasitic traffic:* preventing unauthorized use of signaling and media streams.

• *Resisting Denial of Service attacks:* blocking or limiting malicious IP or signaling traffic.

• *QoS policing and remapping:* ensuring correct priorities are assigned to traffic

• *Anti-tromboning:* preventing unnecessary media traffic in the access network.

Session Admission Control

The basic principle of session admission control is preventing the admission of more calls than the network can handle at any given time. At the simplest level, Session Admission Control involves modeling the bandwidth in the access network and monitoring the resources consumed by each new call. Once the limit is reached, further calls can be refused, thus guaranteeing the quality of existing calls.

Consumers of voice services are fickle and don't like to encounter the unexpected.

At a more detailed level, a corporate customer with an IP PBX may be permitted to make up to ten concurrent calls — equivalent to ten external lines on the PBX. Once the agreed limit is reached, further calls can be refused. However, one important exception must be made to the policing: If an emergency call is received it will always be carried, regardless of any restrictions.

Parasitic Traffic

As a simple example of service theft, a user might signal that a voice call is being made, then initiate the exchange of high-capacity video data once the path is established. This hits the service provider on two fronts: a) loss of revenue by billing for only a voice call, and b) potential degradation in service quality for other users resulting in dissatisfaction.

The structure of a VoIP call with separate media and signaling streams can lead to some innovative ploys. For example, rogue PC clients can transport media in the RTCP quality monitoring stream without ever being policed in most networks. Another ploy involves transporting media in the call signaling stream then failing the call before billing commences resulting not only in unbilled calls but in repeated call sets that can cause huge signaling rates that themselves constitute a DoS attack.

To counter creative new strains of fraud, SBCs and other devices must police all components of a call to ensure that the call is executed as requested and RTCP traffic is within expected bounds.

Resisting Denial of Service Attacks

All devices that support voice and multimedia over IP are subject to DoS attacks including traditional IP logic and flood attacks as well as newer exploits at the SIP signaling level and application level. While conventional firewalls go part way toward preventing IP level attacks, firewalls are still typically session-unaware and unable to recognize SIP level attacks.

Here it falls to devices such as SBCs to prevent the propagation of disruption in access networks. Measures must be taken to keep malicious traffic out of the network, thus preventing it from becoming clogged. Service provider infrastructures must be shielded from being overwhelmed by limiting signaling rates to levels that Softswitches in the core can handle.

QoS Policing and Remapping

All calls carry information in IP packets that determines the priority a packet receives as it traverses the network. Service providers must ensure quality settings are enforced so that they can assign the correct service to different types of traffic and prevent some users from manipulating the settings at the expense of other users. True carrier class SBCs equip providers to control quality settings based on traffic source, type, and even individual users.

This function is invaluable for interconnection of service providers networks as well since QoS policies and policing capabilities may not match the settings used by a peering providers. Here, the SBC steps in to remap quality settings as traffic is exchanged.

Anti-Tromboning

This strange term refers to the trombone-like shape that a call path takes when a local call is made via a distant SBC. When both end points of a call are in the same network, it may not be necessary or desirable to route the media via the SBC. When the SBC detects that the call can be more efficiently routed directly between the end points it will release the media, thus freeing up valuable bandwidth in the access network.

Regulation of Services

We have already mentioned that networks must provide Emergency call capabilities under all conditions, and evolving regulation of VoIP and MoIP services increasingly requires operators to provide Lawful Intercept capabilities as well. While this may not be seen as part of the quality management of the network, the imposition of these requirements means that inappropriate or illegal activities are more likely to be discouraged. Implementing lawful intercept capabilities demands that service providers know about all the signaling and media flows in their network. Once again, the SBC is a natural candidate to support these requirements.

Premium on Performance = Premium on Profit

Consumers of voice services are fickle and don't like to encounter the unexpected. Carriers are literally betting the business on their ability to strike the right balance between cost, quality, and control in provisioning access networks. Overprovisioning the access network doesn't translate into a reliable service delivery platform in the face of bursty data traffic.

Only a multi-faceted approach to policing available resources can ensure customer confidence and lock in loyalty by delivering consistent levels of service, day in, day out. Providers must anticipate and engineer for varying traffic patterns, employ contingency planning that sustains customer communications while upholding the law during emergencies, and literally put their money where their mouths are, in terms of SLAs and fraud prevention.

As global communications migrate toward all-IP infrastructures a little each day, service providers continue to seek ways of rapidly embracing a more profitable future while preserving the legacy of quality taken for granted in the past and present. Fittingly, as the realities of VoIP become clearer and more present each day, a balance between innovation and tradition is being struck in a level of the network for which the PSTN has no counterpart.

And if service providers and their partners do it right, end users need never even know this level is there, even as they gladly pay a premium for the quality it guarantees. IT

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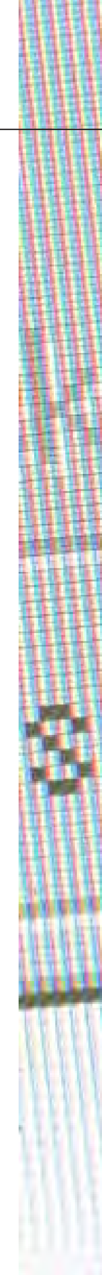
Go To Table of Contents | Go To Ad Index



SIP: Enabling The Hidden Potential Of VoIP

GEMAYA is a term coined by David Kirkpatric in a recent *Fortune* article as an acronym for Google, eBay, MSN, AOL, Yahoo, and Amazon — the Internet heavyweights. Like IBM (quote - news - alert) and the BUNCH from the heyday of large computers, the distinguished GEMAYA group is likely to pioneer a new generation of interactive services layered atop their current core offerings. The integration of voice, video, banking, gaming, and other real-time applications with instant messaging, chat rooms, and e-mail is already well underway. One example is eBay's recent acquisition of VoIP provider Skype (news - alert). Ponder for a moment the potential of combining a global marketplace with robust financial services (PayPal) — both backed by voice, video, multimedia messaging, and more.

Will one-stop-shopping with such sophisticated capabilities enable GEMAYA to leapfrog traditional voice carriers with fully bundled and low-cost (or free) services? There is no way of knowing today, but two things are fairly certain at this stage. First, driven by IP cost models and service potential, these are (finally!) exceptionally interesting times to be in the Internet telephony business. And second, regardless of the direction the market takes, the Session Initiation Protocol (SIP) is destined to play a critical role. This article analyzes both the solid core and ragged edges of SIP (and some related protocols), and investigates those areas where evolution is most intense and current implementations are most challenged. These areas include interoperability, security, quality (as a superset of QoS), middleware and 'middleboxes,' and support for rich application deployment.

The key to creating winning solutions in this rapidly evolving space will be the ubiquitous user experience — all applications, anytime, anywhere, on any device. Such ubiquity is most likely to be derived from formal standards, as opposed to a single de facto standard, largely because no single player has enough critical mass to force-feed a proprietary solution that satisfies the absolute need for interoperability. As in many leading-edge areas, the standards are evolving and competing. Still, Session Initiation Protocol (SIP) (define - news - alert) is the new standard for real-time services. The reasons are straightforward. SIP is a flexible, extensible, rich, and highly-leveraged specification. SIP puts few bounds on potential applications, allows for extensions and enhancements, and reuses some of the best Internet technologies to date (for example, encryption and authentication mechanisms, and MIME data types). Of course, with flexibility and richness comes the potential for complexity and confusion. Today, SIP is in its adolescence; much has been learned since its infancy, and much more will be learned as it matures.

Interoperability

Interoperability, or even operability, continues to be an issue in today's growing SIP-based infrastructures. While there are still significant discussions around interpretations of the SIP specification, many of these issues occur in areas the specification simply does not address. Still fundamental in this area is the issue of NAT and firewall traversal — SIP uses IP addresses in order to set up sessions (How else would it be done?), and these addresses are invalidated by Network/Port Address Translation — and firewall behavior.

By Todd Simpson and Alan Hawrylyshen

Many non-standard approaches are in use to solve this problem, but none can claim to be 100 percent effective owing to the overwhelming variety of NAT and firewall devices (for example, port-rotating firewalls can still be problematic). Emerging standards and processes, such as STUN, TURN, and ICE should soon allow this area to become more cohesive.

Other interoperability issues can occur based on the implementation. For example, SIP does not specify an upper bound on header sizes, but many implementations have hard-coded bounds; messages beyond these bounds are rejected. Improper implementations of registrars and proxies, either through bugs, incomplete designs, or misinterpretation of the specification can also lead to interoperability problems. Lack of support for full DNS resolution (as documented in RFC 3263) including NAPTR and SRV support, basic security mechanisms, and flawed CODEC negotiations might also lead to interoperability issues. Fortunately, the industry recognizes the need for solid interoperability, and efforts such as the SIP Forum's SIPit events have enabled much progress. And yet, the industry is likely to end up with pseudo-interoperability, much like with HTTP, where content works better in some browsers than in others. Of course, unlike the World Wide Web where only a handful of browsers are used, the world of VoIP currently involves myriad different systems and devices from a plethora of vendors.

Security

Now that several large VoIP access networks are operational on the Internet, the issue of security is becoming increasingly important. And like all security concerns, there is a mix of real issues and fear-mongering. The SIP specification leverages the best of mature Internet security models, which, when fully implemented, distill the true areas of concern to border cases. Unfortunately, the state of deployments today tends not to include even basic security provisions, leaving networks open to many sorts of breaches.

SIP contains excellent support for ensuring point-to-point confidentiality and encryption, including the adoption of digest authentication, S/MIME, and TLS encryption, and the ability to share keys for media encryption. Many of these mechanisms, however, still have subtle technical or management drawbacks; for example, sharing keys for digest, or managing the tradeoffs between end-to-end and point-to-point architectures. For these reasons, implementing full cryptographically secure end-to-end authentication remains a challenge, especially given the realities of disparate domains of trust and the existence of the middleware boxes needed to overcome other interoperability issues. Finding and deploying a full solution that addresses this problem will also be essential to combating SPAM over Internet Telephone, or SPIT, as well as other nefarious attacks.

Another example where security could be compromised is with forked requests — where a single invitation is sent to multiple contact points (a home and cell phone, for example). Because the means to authenticate and authorize the response from each fork is not well specified (a proxy many only return the response from one endpoint, somewhat arbitrarily chosen) many different behaviors are possible. This problem area is known as the Heterogeneous Error Response Forking Problem, or HERFP, and remains under discussion at the IETF.

An additional example is the interplay of SIP with other Internet protocols. Routing of SIP requests is often handled via DNS using the NAPTR and SRV records. SIP itself does not specify how to validate or authorize DNS results, so tampering with SRV records can be used to misroute messages. The interplay between SIP and other protocols is a fruitful area of research and implementation innovation. Of course, security breaches that compromise underlying (and unprotected) protocols and resources are not unique to VoIP, in general, or SIP, in particular. This widespread problem is why work is ongoing by the DNSSEC group and others at the IETF to enhance DNS integrity.

Quality

Overall user experience, or "quality," is also still hit or miss in today's networks. Again, SIP itself is not the culprit, or perhaps even the solution, but the interplay between SIP and other functions needs to evolve to address this issue. Quality includes always having connectivity, the speed of connection (ringing, for example), and the quality and consistency of media delivery. Within one homogeneous environment, adequate quality may be delivered by making logical network and bandwidth decisions, and enforcing these across the network. Across heterogeneous networks, however, the problem is significantly more challenging. Where one network may employ MPLS or VLANs to guarantee QoS, another may simply use DiffServ and, where an IP network meets the PSTN, numerous interface issues (such as echo and security) can also occur. Even between peering partners carrying purely IP backbone traffic, there may be different design choices, especially involving encoding. For this reason, a network that is optimally designed for larger packets at slower intervals may not work well with a network optimized for smaller, more frequent packets. Jitter and latency correction at the endpoints may not be sufficient to compensate for arbitrary media routing.

Deploying services beyond VoIP across SIP-based networks has its own set of challenges. Environments like IMS and TISPAN (the 3GPP's IP Multimedia Subsystem and ETSI's Telecoms & Internet converged Services & Protocols for Advanced Networks) anticipate video, TV, gaming, access to back-end databases, and many other session-related applications. Administering, controlling, authorizing, and guaranteeing quality of service across these multimedia networks is non-trivial. For example, while a few seconds of delay in ringing another user agent may be acceptable for telephony applications, it may be completely inadequate for a massive online gaming environment. Likewise, while non-repudiation may be manageable within voice-only applications, it must be rock solid for banking

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applications. Thus, the scope and complexity of the issues mentioned above become more acute in a rich application environment. And overcoming interoperability issues between SIP and existing security mechanisms, like X509, S/SMIME and the related PKI technologies, will be essential to widespread adoption of these services.

At the other end of the spectrum from IMS (define - news - alert), which has a heavy back-end infrastructure, is the work on P2P SIP, which attempts to remove any dependence on a back end. The ability to quickly set up ad hoc, real-time sessions using a P2P system (along the lines of Skype) has obvious advantages to the end user — and disadvantages to service and equipment providers. There is nothing fundamentally difficult in having P2P endpoints communicate with an IMS infrastructure, however, other than the previously highlighted management and control of such services. Pure P2P SIP applications will occur; planning for their integration and control is only just beginning to be discussed.

Conclusion

As the clash of the titans heats up — GEMAYA versus The Incumbents — SIP is positioned to play a central role. While SIP has certainly proved its flexibility and worth to date, it still has much room for improvement and growth. This makes the SIP battlefield an exceptionally exciting and (potentially) prosperous place to be. SIP is solidly positioned to be the underlying standard for real-time services on the Internet. And given the relentless growth in connectivity and available bandwidth, what application can afford to not be real-time! IT

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Go To Table of Contents | Go To Ad Index

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VoIP Market Enters New Era

System-On-Chip (SoC) Media Processors Combine with Open Source Software to Accelerate Already Rapid Market Growth

After years of mounting expectations, the worldwide VoIP market is experiencing significant growth as service providers, businesses, and consumers realize the deployment benefits of converged, packetized voice and data services. Designers are rapidly adding these mixed-media capabilities to what have heretofore been islands of voice-centric or data-centric networks.

In the residential market, the focus is on integrating basic voice features into a single packet-based gateway. Meanwhile, enterprise class designers are focusing on adding full voice quality and secure VPN data routing into converged mixed-media platforms. Carrier class systems also can benefit from this new mixed-media architectural model. In all three cases, the advent of open source voice applications has stimulated important new opportunities, providing designers with a head start on basic convergence features, and a foundation upon which they can add customized features and capabilities while optimizing quality, performance, and cost, with faster time to market.

The open source model — including the ability to run open source and third-party applications and access a variety of hardware integration options — enables manufacturers to cut development costs and execution risks while accelerating time to market for broader product offerings with expanded feature sets, improved real-time performance and scalability. At the same time, manufacturers also can leverage their development investment across a wide range of residential, enterprise, and carrier applications.

Design Examples for Small Businesses

VoIP (define - news - alert) migration has now reached the doorstep of small businesses as an exciting alternative to expensive private branch exchange (PBX) technology. The same concept illustrated by VoIP in the home applies to VoIP-based PBX technology in the small business environment with minor complications. First, the enterprise gateway has to support four or more phone lines, fax machines, and point-of-sale terminals. Additionally, voice quality and the telephony gateway application have become major issues. For the small business, even a small PBX is out of reach from a cost perspective. In contrast, a combination of off-the-shelf and open source VoIP technology building blocks combine to produce an elegant telephony solution.

The main required building block is the telephony application that runs the entire system and interacts with the telephony provider. Up until now, PBX vendors have kept their telephony applications proprietary and charged high fees for service contracts. The open source revolution disrupted this business model when the Asterisk PBX software solution finally was released to the public. Asterisk is an open source PBX and Internet Protocol (IP) PBX telephony application that offers both traditional and next-generation network features such as Session Initiation Protocol (SIP)-based IP telephony (define - news - alert) or peer-to-peer directory services. Installing and maintaining Asterisk requires some effort but is justified given the savings in the operating expense.

The other required building block is the hardware for running Asterisk. The standard hardware is an off-the-shelf PC with the desired telephony adapters. Asterisk runs with no modifications on a standard PC; however, this configura-

By Diya Soubra

tion may not be ideal when voice quality, scalability and reliability are key concerns. When special requirements exist in these areas, designers need to consider using an embedded media processor from companies that supply semiconductor devices to the telephony industry. In this application, Asterisk requires some modifications to offload the media processing to an embedded device. The latest embedded media processors offer direct access to off-the-shelf open source applications and don't require extensive software redesign, opening up a host of new opportunities for designers.

Working with Embedded Media Processors

Today's latest voice and data convergence platforms offer, on a single chip, the horsepower of two independent CPUs that share a common SDRAM and yet independently handle signal and packet processing. They also integrate a flexible high-performance encryption engine, all security and other VoIP features into a single device, and include a variety of high-speed interfaces. Their extensive software suites support all wireless and wireline voice/fax codec standards, echo-cancellation technology, and IP protocols and modulations. Additional functionality can be included through software upgrades. They run off-the-shelf open source Linux and Linux applications, and use standard open source development tools.

There are several examples of open source VoIP applications where the use of an embedded media processor offers significant advantages. The first is a company that has already invested in a PBX and wants to conserve its capital investment by moving the PBX onto a VoIP network. Another example is a small company that has invested in a small stand-alone telephony box that provides Internet access, routing functionality and IP PBX capabilities. In each of these examples, an open source VoIP application supported by an embedded media processor yields the desired voice quality, scalability, and reliability.

Multiple criteria should be considered in selecting a media processor platform. First, dual processor designs offer the advantage of an architecture with deterministic carrier grade voice quality and the ability to scale from data-only applications to full packet telephony capabilities operating on up to 32 complex voice channels. Secondly, the device should have a security engine for voice security and VPN connections. Thirdly, the device must have an extensive and programmable software suite, and deliver a sophisticated open source processing system for a complete mixed-media processing solution on a single chip. And finally, the system-on-chip (SoC) should not require extensive software redesign when integrated with an open source application, which eliminates the risk of any impact on the real-time functional blocks like voice processing.

The Asterisk Open Source telephony application provides an outstanding foundation for realizing the benefits of SoC media processors. It is now a mainstream application that is allowing businesses around the world to benefit from the VoIP revolution at a fraction of the original cost using traditional approaches. Any possible voice quality and scalability shortcomings of the Asterisk telephony application are easily overcome by augmenting the system architecture with single-chip media processing devices. Asterisk's modular architecture was designed to allow the straightforward insertion of media processor devices, enabling systems to guarantee the voice quality that is absolutely required in a business setting.

Emerging Developer Ecosystems

Along with the availability of media processors and open source software applications have come comprehensive new development ecosystems consisting of multiple vendors collectively supporting a common open source hardware and/or software platform. These ecosystems enable VoIP system designers to quickly and easily integrate single-chip VoIP media processors with products and services from a growing network of participating hardware, software and contract manufacturing companies.

Open source development ecosystems give both new market entrants and established voice equipment players alike, the necessary resources with which to quickly create baseline designs and then value-add their own customized features and capabilities, as desired. Design options range from complete turnkey solutions to flexible designs that can be easily customized and differentiated for specific market segments. Basic elements of the development ecosystem include a processor platform optimized for open source VoIP applications, a rich library of open source and third-party applications, and a comprehensive development/manufacturing ecosystem.

New Applications, New Profit Opportunities

The open source approach differs dramatically from traditional component-centric approaches that require extensive internal development work on the part of the equipment manufacturer. In contrast, open source initiatives create a robust and significantly more comprehensive solution while offering numerous advantages to customers and partners, alike.

The design possibilities are broad and growing. For instance, a vendor can integrate a customized API channel module for a media processor into the Asterisk Open Source PBX application, enhancing the time to market advantage of enterprise equipment customers designing open source-based PBX systems. Another vendor might integrate session initiation protocol (SIP) software into the media processor, creating an enhanced system-on-chip voice processing solution for enterprise PBX, gateway, and voice-enabled router equipment. A third vendor might integrate a media processor into a family of voice-gateway modules, allowing equipment manufacturer to quickly add robust voice services to existing router, enterprise PBX and carrier gateway equipment.

The more comprehensive the development ecosystem — including software application developers, design services partners, and turnkey ODM solutions providers — the richer the design opportunities, and the faster the time-to-market. Increasingly, raw silicon per-

Design options range from complete turnkey solutions to flexible designs that can be easily customized and differentiated for specific market segments.

formance isn't enough. Today's open source media processors with built-in encryption engines are capable of processing up to 32 channels of highly secure VoIP and routing data at up to 150 Kpps, with up to 60 Mbps of throughput for highly secure virtual private networks. While it is possible to find better performing SoC processors, this means sacrificing the ability to support off-the-shelf open source software, and the result is much more costly software rewrites and customization requirements. Indeed, it is increasingly important that designers consider the accompanying software development and/or customization, which is substantially streamlined through the use of open source software run on an embedded media processor and supported by a comprehensive development infrastructure and ecosystem.

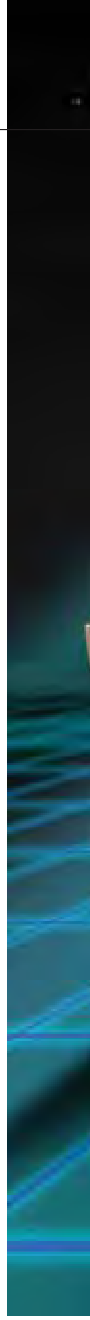
VoIP is here to stay. The technology is now in widespread use at millions of homes worldwide, and the open source model is poised to further accelerate its adoption. Everyone benefits. Manufacturers increasingly can leverage developer ecosystems and take advantage of reduced designs costs/cycles for modular and scalable products. Open source vendors benefit from the synergies inherent with offering a holistic and coordinated multi-vendor approach to VoIP equipment design based on a foundation of open source applications. Service providers benefit by diverting voice call profit from traditional operators. And finally, users benefit by having lower operating cost and access to next-generation applications that better integrate communications with the overall business process. IT

Diya Soubra is the director of marketing for the enterprise product line and manages the European Engineering Design Center for Mindspeed Technologies. (news - alert)*For more information, please visit the company online at http://www.mindspeed.com.*

Go To Table of Contents | Go To Ad Index

Can't We All Just Get A Line?

Private and public sectors must play together to connect the disconnected.



As the first light of dawn emerges from over the horizon, somewhere in rural America, down an old, dusty dirt road, a weathered farmer nestles in front of his antiquated computer, waiting patiently as the screen slowly fills inch by inch with news of the continued drought. Once again, the bulk of this year's income will come from government subsidies. At that same moment, just outside the city limits, a teenager sits before his laptop, his eyes come alive from a flood of sights and sounds; it's barely six in the morning and already he's watched the morning news, a music video, and he's downloaded a dozen songs to put on his brand new MP3 player.

This contrast of scenes, in its many forms, plays itself out millions of times each day as the disparity continues to widen between this nation's broadband-enabled citizens and its economically and access-challenged. With each year that passes, technological progress provides a greater quantity and quality of broadband offerings to those with the opportunity of access and affordability. The consumers who remain outside this circle become increasingly disconnected from the flurry of advances that make communication faster and easier.

In growing numbers, municipalities across the country are working to bridge the gap, providing low-cost, high-speed connectivity to their citizens. But, in many cases, these municipal governments are going it alone, adopting the appearance of a competitive company without the experience necessary to thrive. Current service providers — with their years of experience, technological know-how and knowledge base — stand to provide an enormous contribution to the success of these projects. The private sector incumbent companies, however, have been slow to form partnerships and many have begun to stomp the fires of competition. The companies are fighting the process through the staunch support of anti-municipal broadband legislation, citing unfair pricing, funding options, and competition as some of the reasons why municipalities should not get involved.

Despite the lack of support from the private sector, all parties agree that affordable and ubiquitous broadband access is a benefit to all. The challenge now remains to get municipalities and incumbent service providers to collaborate to close the connectivity gap and make broadband a reality for the millions of Americans beyond its reach.

Lagging Behind

Those who believe the United States is leading the charge to bridge the digital divide would be surprised to know that according to the International Telecommunication Union (ITU), the United States ranks 16th among nations that have adopted the use of broadband. Countries not known for their technological supremacy — Israel, the Netherlands, Denmark and Canada — have all surged ahead in the race to ubiquitous broadband connectivity.

Travel to South Korea, where the world's most broadband-enabled populace enjoys relatively inexpensive, prevalent connectivity, and you'll find a hyper-connected society. More than 75 percent of South Koreans use broadband, which has become not only a pervasive cable-based technology, but wireless as well, with millions of citizens using broadband-enabled 3G mobile phones. The surge in broadband use over the past decade can be attributed to an initial government campaign, backed

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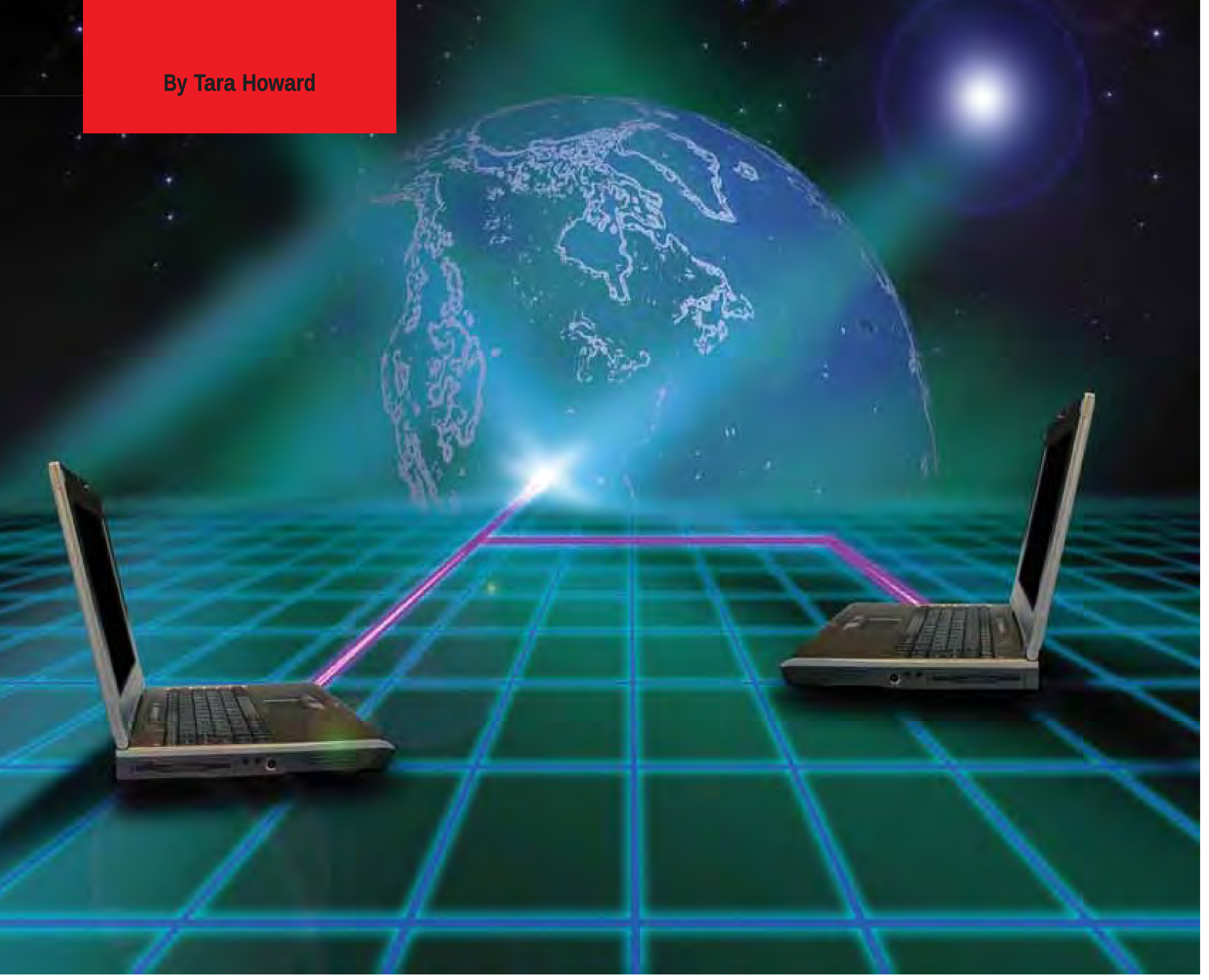


By Tara Howard

by billions of dollars in public investment, to provide high-speed Internet to educational institutions. As competition pushed prices down and availability spread, Internet-savvy consumers — 50 percent of whom play online games — rushed to adopt high-speed service.

Although demographics (a largely urban-centered country) and a national obsession with video games and online culture have played an enormous role in the rapid rise in Internet penetration in South Korea — 3 to 75 percent in less than 10 years — the prosperous Asian nation stands tall above other countries, including the United States, in providing its citizens with broadband capabilities. In 2005, it's not uncommon for someone in Seoul to have a 20-megabit connection — 15 times faster than a normal cable modem in the United States.

The Homefront Push

In the United States, communities ranging from the metropolis to the rural town are seeking to lessen the disproportion of accessibility and provide taxpayers with a valued and desired service are spearheading their own broadband initiatives with varying levels of success. Cities such as Denver and San Francisco and small towns such as Adel, Georgia and Nantucket, Massachusetts have all entered the ranks of communities developing municipal broadband projects and each faces its own set of challenges.

For those small communities that are often without any broadband provider, the challenges are many. Reminiscent of the push to bring electricity to rural towns in the late 1800s and early 1900s, the municipal broadband movement echoes the movement of municipalities to convince power companies to extend service beyond metropolitan areas. Not until local power utilities were formed did these communities turn the lights on, modernize, and flourish as industrious segments of society.

Helping to speed the process from dial-up to broadband, many communities are increasing their efforts to develop a network to support government, education, and public safety applications. Business development is yet another stimulus for bringing broadband to an economically stagnant or waning local economy, although profit is not always the deciding factor for

Communities seeking broadband networks face opposition from providers who fear the direct competition.

developing a publicly owned infrastructure. Often, the initial development is extensive (and costly), so governments see the projects as an investment, a means for attracting more taxpaying citizens and businesses.

Entering the broadband business without private sector partnership leaves municipal governments in a difficult situation, as municipalities lack the inherent technical and business expertise needed to make such projects successful in the long run. Currently, communities seeking broadband networks — whether fiber or wireless — face opposition or indifference from providers who either fear the direct competition or that can't see the value of a possible partnership. While nearly all agree that the propagation of broadband can only lead to positive economic and social growth, municipalities and incumbent telecommunications operators are often at odds as to who should implement the rollout of these networks.

Two models of municipal broadband realization currently reign as the most common of options — retail and wholesale. When private companies choose to pass on partnership with municipalities — which is so often the case — local governments adopt a retail model and become the owner and operator of the broadband facilities, as well as the provider of Internet, video, and telephony services. The municipality will generally leave the network as open access, allowing smaller providers (local ISPs or CLECs) network access while the incumbents ignore the partnership opportunity. The wholesale scenario brings a private sector provider into play, as municipalities establish the network, but lease its potential to a willing provider.

Playing Together

For community-driven broadband networks to succeed, both the private and public players will need to take responsibility. Local governments must first do their homework by assessing a number of factors. Most importantly, they should first determine the feasibility of the project, and then once viability is ensured, an evaluation of end user demand, technology choices, local infrastructure availability, and vendor selection will help in developing a strategic plan. From the private sector angle, telecommunications companies hesitant to saddle up should first accept the inevitability and long-term benefit of municipal broadband. Their involvement in these projects is crucial to the long-term success and eventual profitability and their movements to curb municipal broadband through the support of legislation will only serve to broaden the digital divide and lessen their opportunities for customer reach.

To assuage the private sector's fears of high-cost implementation, communities can dip into various funding programs such as education, public safety, healthcare, and transportation, adding value to a proposed partnership. In addition, communities can not only leverage their infrastructure rights of conduits, buildings, and fiber assets — using existing utilities can often make or break community broadband networks — but they can corral interest from their consumer and business constituents with various national and local organizations, including education, healthcare, and public safety.

To woo the private sector into partnership, municipalities will need to stay out of direct competition with local telecommunications companies, as well as cellular carriers, cable MSOs, and WISPs. The more a municipality can partner with these entities and leverage existing infrastructure and public monies, the easier a partnership will form and the greater the chances for successful implementation and measurable ROI. As partnerships form and telecoms warm to the municipal broadband movement, the network design will need to utilize existing technologies and equipment that take into account the area's specific needs, geography, and economics. A rural community — seeking reliability and cost-effectiveness — will not require the same features and functionality provided to residents of an urban center. Therefore, the private sector will need to take a more strategic look at its partnerships and its provision of technology.

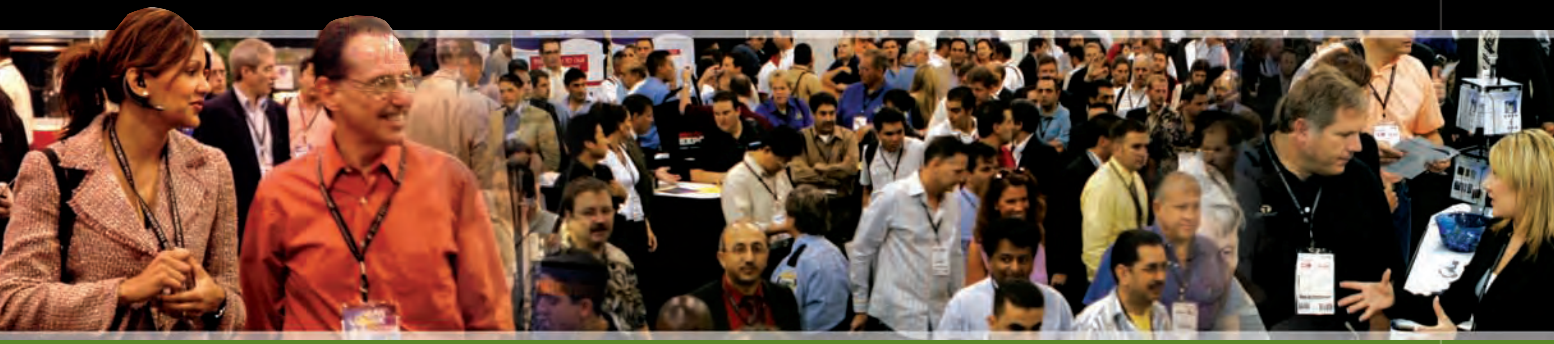
Outlook

Fiber networks, especially from a cost perspective and geography, will prove a more successful venture for rural communities with the right private sector partners than a wireless solution. Urban communities find the most feasible and cost-effective solution for a successful partnership in wireless broadband. As the price and complexity of next-generation infrastructure declines, a window has opened for municipalities to build their own networks with the support of vendors and carriers. As we move forward and more emphasis is placed on bridging the ever-increasing connectivity gap, the key technology will be WiMAX (define - news - alert) .

Although strong business models and best practices don't yet exist, partnerships between the private and public sector will prove to be the lynchpin behind the development of rural and low-income communities. The existence of hundreds of municipal broadband projects loudly signals a demand for high-speed connectivity. The economic future of many communities relies on this much-needed technology. It's in the hands of municipalities and telecommunications partners to ensure that broadband becomes a reality for the millions without as this nation continues to feel the effects of the deepening digital divide. IT

Tara Howard is an Associate Analyst with Yankee Group, (news - alert) a global leader in providing knowledge, tools, and support to help clients make winning decisions when business opportunities intersect with technology solutions. For more information, please visit the company online at http://www.yankeegroup.com.

Go To Table of Contents | Go To Ad Index



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VoIP MARKETPLACE

CEO Spotlight

Tim Burne
CEO
VegaStream

Formed in 1998, VegaStream (news - alert) is a veteran player in the VoIP market. The company manufactures VoIP gateways for telephony providers to allow interoperability between today's various phone systems and IP networks. CEO Tim Burne, who has been with VegaStream since 2000, had this to say about his firm making its way into the next generation of Internet telephony.

GG: What is VegaStream's mission?

TB: Today, VegaStream's mission is to accelerate businesses' deployment of VoIP (define - news - alert) by providing reliable, easy-to-use technology that interconnects traditional TDM networks and equipment with the VoIP platform. Going forward, we anticipate our core gateway technology to be a pivotal enabler of future platforms by providing the essential link between ensuing generations of network technology.

We are dedicated to the needs of business. We approach the market in partnership with both telephony systems integrators and service providers. Many business customers are reluctant to forklift out their entire existing telephony systems, but can achieve real business benefit by deploying VoIP in call center and customer contact applications. Our Vega gateways enable systems integrators to offer their customers cost-effective connectivity between new VoIP solutions and the PSTN and existing TDM equipment. These hybrid solutions are very much a part of the evolution from TDM to VoIP, and our gateways are making them happen.

Tier 2 and 3 service providers that have made a full commitment to VoIP are another critical catalyst for the evolution of VoIP. Their success is an important force driving the Tier 1 carriers toward VoIP. To be successful, they must be able to sell their services to customers who may not have made a similar commitment (i.e., those who want to retain their traditional equipment). VegaStream gateways are helping service providers around the world to satisfy this demand. For example, Bulgaria's second carrier, Orbitel, is bypassing the traditional PTT resale model by offering customers a voice service delivered over the ISDN network and Vega gateways.

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

TB: VegaStream is at the heart of the next-generation telecom market. That market exists today and is typified by multiple network bearer technologies from core to edge. There will be no overnight wholesale switch from TDM to VoIP, just as there was no wholesale switch from fixed to mobile. In the business space, few enterprises make a decision to switch to new technology purely for the sake of it — there has to be a business case. Until such time as the capital cost of those billions' of dollars worth of PBX and handsets is written off, there will be hybrid VoIP/TDM networks that need a gateway.

Furthermore, the Tier 1 carriers have only made their commitment to IP networks in recent years; it will be a decade or more before they are fully implemented. By that time, who knows what the new next generation of bearer network will look like and what gateways will be required. That's where we will continue to deliver value.

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

TB: The strategic shift from VoIP to TDM is underway and I can see nothing that will change it. Fundamental laws of physics are driving it. With IP you need less network to carry more traffic. It simply is more efficient. The fact that you can do a whole lot more with voice traffic over IP is icing on the cake.

There are many with vested interests who fear the rapid take-up of VoIP, and they are quick to raise issues that may deter customers and impede its progress. Security is the high on the list. However, it was not that long ago, in strategic terms, since the telecom industry migrated its central offices from inefficient, but intrinsicly secure, electromagnetic switches to digital switches — or huge, hackable, network servers as we might call them in modern parlance. However, there are no perceived security issues with these switches, both because the engineers have solved them and because few people had a vested interest in raising the spectre that they might present a security risk.

I believe that engineers will solve what security issues there are in the VoIP space. However it is fair to say that, as of today, we see little real evidence of VoIP insecurity where it is actively deployed by enterprises that have already secured their IP networks.

The other major hurdle the industry has to overcome is in its marketing. We are not making it easy for customers to understand our proposition. We still fall into the trap of selling technology, not solutions. People don't buy VoIP, they buy what it can do for them, and I feel that the business market at large is still

Go To Table of Contents | Go To Ad Index

waiting to hear what we can do for them beyond talking to mates overseas via the PC.

> People don't buy VoIP, they buy what it can do for them, and I feel that the business market is still waiting to hear what we can do for them.

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

TB: While VoIP technology is accelerating beneath a constellation of open standards, a "Milky Way" of standards illuminates the TDM (define - news - alert) world. So our focus has been, and will remain for some time, interoperability, interoperability, interoperability. For VoIP to really take off in the enterprise space, particularly in the SME space, it has to be based on "plug and play." The prize will go to the first gateway manufacturer to produce a device that the user can install between PBX, PSTN, and VoIP services and set up a seamless, integrated dial plan — by themselves. We aim to be that manufacturer.

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

TB: I foresee a future where all new telephony implementations, fixed line or wireless, will be IP. That future is not too far away. However, I also see a prolonged life for existing TDM networks and equipment. You simply do not discard a technology that has served the world so well and for so long, in a decade or so. The gateway providers will be the essential membrane that will ensure the integrity of a single voice network where all subscribers can connect to each other.

Meanwhile, as technologists and customers come to understand that multiple bearer networks based on different technologies can successfully interoperate, so can we expect an acceleration of new bearer technologies. After all, in the 100-year history of the telecom industry, we have made but one tectonic shift in the core "operating system." I predict that, in the next few years, we will see something that will represent a sufficient advance on IP telephony that will require a gateway between it — VoIP, as we know it today — and TDM networks. IT

CEO Spotlight

Michael Robinson
CEO and Director
Citel Technologies

In the CEO Spotlight section in *Internet Telephony*®, we recognize the outstanding work performed by exemplary companies. Each month we bring you the opinions of the heads of companies leading the Internet telephony industry now and helping to shape the future of the industry. This month, we spoke with Michael Robinson, Chief Executive Officer and Director at Citel Technologies. (news - alert)

GG: What is Citel's mission?

MR: Citel's mission is to simplify and accelerate VoIP (define - news - alert) migration for enterprises and service providers.

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

MR: Our vision is to empower enterprises and service providers to realize the cost and productivity benefits of IP telephony, at the same time leveraging as much of their existing infrastructure as possible. Billions of dollars have been invested by enterprises of all sizes in digital PBX equipment over the last five to ten years. Many are not aware that the handset they use today can be upgraded to provide all the most useful VoIP features instead of having to buy a replacement.

One of VoIP's greatest benefits is cost-savings, yet one of the greatest inhibitors to deployment is the expense and business disruption associated with rewiring, installation, and training. It is extremely satisfying for us to hear our customer's stories — stories about banks, universities, retail chains, hospitals, and others who were able to deploy VoIP overnight, resuming normal operations the next morning with little or no disruption.

These are everyday examples of enterprises deploying VoIP for the purpose of realizing real business benefits, rather than simply "deploying VoIP for VoIP's sake." This puts Citel in an excellent position to serve the 90 percent of the market still using digital PBX handsets to run their businesses. In fact, digital PBX (define - news - alert) equipment will constitute the majority of the installed base well beyond 2009.

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

MR: Our customers tell us that complexity and confusion are two of the primary inhibitors to deploying VoIP. Most enterprises understand the benefits of VoIP in terms of feature enhancement coupled with cost savings. We educate customers on how to leverage existing infrastructure to realize the benefits of VoIP without expensive equipment replacement or disruption of their operations. The end result is a customer who is comfortable migrating to VoIP on their own terms, at their own pace, and at a far lower cost than that of other migration solutions.

In addition, many organizations have disparate systems from various manufacturers across multiple locations – a common scenario in the retail and finance sectors. This presents an additional deployment challenge: which PBX's or handsets should be replaced first? What interoperability issues might we face during the rollout? Citel's technology enables both digital and IP PBX handsets from a variety of different vendors to converge on a single IP telephony platform, dramatically extending the useful life of existing investments and paving a roadmap to the future.

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

MR: First, we are investing heavily in the hosted VoIP market, as we believe this segment has a promising future. Important new IP-based services are coming to market that can serve small to medium-sized businesses on a very granular level — offering features previously reserved only for the largest enterprises. This allows service providers to offer new features that have a profound impact on the productivity and efficiency of their customers.

Second, we are continually enhancing our current hardware and software to offer a greater amount of scalability and flexibility, whether an enterprise plans to migrate to IP telephony utilizing their existing PBX, a new all-IP PBX, or a hosted solution.

Lastly, we are working to make it easier for enterprises to decentralize their communications networks to meet their business needs. Historically, a distributed architecture included headquarters and branch offices, each using a number of disparate systems and technologies. Today, IP technology is creating an ever-flattening global landscape. Every employee can be connected as an extension of the headquarters, regardless of their location or device. Our roadmap is focused on developing solutions to accelerate this trend, where every employee can be as productive as they can be.

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

MR: Although VoIP as a technology is well established, the transition to it is still in its infancy, and enterprises are just beginning to realize the substantial benefits of an IP telephony platform.

Applications that improve productivity and service will drive the adoption of IP telephony, and many of these applications will be integrated into the hardware platform. For example, Citel has developed an application that enables remote workers to share a single phone number for their desk phone and their mobile device, with a single voice mail box and similar functionality between the two. These services are agnostic to the mobile service provider and the PBX vendor, but are integrated into our platform to add value.

It's important for an enterprise planning their migration to carefully review all options, whether it's a rip-and-replace move to an IP system or various forms of balanced transition utilizing existing infrastructure. Citel's role is to help enterprises find the solution that fits best for them. IT

Go To Table of Contents | Go To Ad Index


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