TMC

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VoIP Developer Wars! (page 6)

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VoIP NEWS

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October 2005

TMC INTERNET TELEPHONY CONFERENCE & EXPO

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The team at Internet Telephony Conference & EXPO has been hard at work this summer, putting together an event that promises to be the best-attended VoIP event in history. Based on the success of the last Internet Telephony show in Miami, and upon early registration numbers, this year's conference, which takes place October 24th– October 27th is expected to draw a crowd of nearly 8,000 attendees to the Los Angeles Convention Center.

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The VoIP Authority

By Greg Galitzine

In Between Events

Just back from the West coast, and it's already time to start planning my next trip across the country. I've just returned from the VoIP Developer Conference, which took place in the beautiful Bay area in South San Francisco. Aside from perfect weather (it was 95 degrees and humid back home in CT; 70 and breezy in San Fran) the show was well attended and for the most part folks were happy.

One of the highlights of the show was a live demonstration of bicycle powered VoIP, (define - news - alert) courtesy of Inveneo. Inveneo has built a Linux-based PC with a VoIP connection that's powered by either a bicycle or solar panel. This "ruggedized" solution is designed to withstand severe climates (such as can be found in Bukuuku sub-county, Kabarole district of Western Uganda, where the solution has already been deployed) and has no moving parts and is designed to be serviced remotely, if the situation so demands.

Inveneo, working with various international relief organizations, is installing this "VoIPcycle" in remote African villages that have neither electricity nor phone access.

Perhaps the greatest need faced by the Third World is safe, plentiful access to clean drinking water, not to mention protection and salvation from the scourge of AIDS and other deadly diseases, but access to communications is critical to the future of these countries as well. Communications empowers the spread of ideas, which in turn fuels a promising future for people who have historically been unable to see beyond the hills of the very next town. My hat is off to CEO Mark Summer and his team at Inveneo for devoting even a little time to coming up with a solution that will help bring real benefits to the farthest reaches of the world. Thank you.

As I said in the opening of this column, I am "in between" shows. I have the opportunity to unpack my bags, get the dry cleaning done, and next thing you know it's time to start packing again: This time I'll be heading to southern California, Los Angeles actually, for Internet Telephony Conference & EXPO.

Next month's event promises to be one of the largest shows that we've ever had the privilege of producing, and I for one am looking forward to it. I especially like meeting the readers of the magazine, and hearing from them first-hand what we're doing right and where we need to shore things up.

I'm very much looking forward to hearing what Michael Powell and Carly Fiorina have to say about the state of the industry too. They are the headliners of a keynote lineup that is just out of this world. For more details on who's speaking and what else is going on at the event, check out the show preview on page 58.

I've said it before: Sometimes it feels like we're all playing a part in a Harry Chapin song. You know what I mean. Here's wishing everyone safe travels. See you in Los Angeles!

-Greg Galitzine, ggalitzine@tmcnet.com

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QUOTE OF THE MONTH:

Businesses that are serious about what VoIP can do for their customers, efficiency, cost reductions, and revenue must be as serious about the benchmarks and tools used to plan, deploy, and continually evaluate VoIP services. There are a variety of challenges including deciding on the tools for deployment and support of VoIP services; assuring quality through service availability monitoring and service performance monitoring; and even billing considerations. There is also a complexity and diversity problem when it comes to selecting VoIP products to deploy

– Yves Cogne (page 85)

WHAT'S ON TMCNET.COM RIGHT NOW

To stay current and to keep up-to-date with all that's happening in the fast-paced world of IP telephony, just point your browser to http://www.tmcnet.com for all the latest news and analysis. With over 5.1 million unique page views per month, translating into over half a million unique visitors, TMCnet.com is where you need to be if you want to know what's happening in VoIP.

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

FCC VoIP Tap Rule: An Open Invitation For Hackers?
The Federal Communications Commission (FCC) gave a green light to law enforcement agencies to tap VoIP-based telephone calls. Exposing computer systems to this potential security breach is an open invitation for hackers, according to digital privacy and civil liberties groups...
http://tmcnet.com/157.1

Separation Of Church And State: Fixed To Mobile Convergence (FMC) Services And Business/Personal Contact Modalities
Enterprise organizations really have to start looking at the new person-to-person contact alternatives that are coming down the pike, in order to intelligently plan their migration to multi-modal business communications.
http://tmcnet.com/158.1

FCC Clears Wiretaps Of VoIP
In order for service providers to comply with the Communications Assistance for Law Enforcement Act (CALEA), the FCC last week ordered interconnected voice-over-Internet-Protocol (VoIP) and facilities-based broadband access service providers to accommodate law enforcement wiretaps.
http://tmcnet.com/159.1

411 Update: Directory Assistance In The VoIP World
Directory assistance provides much more than a telephone number and an address for VoIP; it serves callers as an experienced concierge. When callers dial 4-1-1 via VoIP, they can now get their hands on movie tickets, stock quotes, sports scores, and even driving directions.
http://tmcnet.com/160.1

Get Ready For The Era Of The Virtual Contact Center
For many years, people have been saying that offshore contact centers are the wave of the future. From a cost perspective, this view was hard to argue with. Yet now we see VoIP bringing about a new era of home-based U.S.-CSRs. Because the cost of providing service has come down significantly, American centers can now compete again on a more even playing field.
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By Rich Tehrani

VoIP Developer Wars

A while back I wrote about comments from fellow blogger Skibare and his theory that Google may purchase Skype and how it would mean "game over" for everyone else. I have been thinking about these comments subconsciously for a while and I have to admit that whether or not Google decides to buy Skype it seems only a matter of time before they launch Gphone.

I can see it now:

"What is Gphone? …As part of Google's mission to make the Web more useful we have decided to launch Gphone, short for Google (quote - news - alert) Phone. Gphone is a free PC to PC program that allows you to make unlimited calls to other Gphone users..."

What happens next, of course, is that Microsoft needs to become a phone company and launches Mphone and of course, then there is IBM, Sun, etc.

Interestingly, I keep hearing how handset manufacturers are considering adding the Skype protocol to their devices. Skype has become that popular. Is this a barrier to entry for Google? Perhaps, but when you have access to just about every Web user out there by simply adding some text to your search page you have to wonder if anything is beyond your reach.

I have been thinking a great deal about developer programs and VoIP ecosystems since TMC's just completed VoIP Developer Conference. I saw with my own eyes how developers were so attracted to companies that are pushing ecosystems like Avaya.

Skype realizes this and last year, right around the time that Niklas Zennstrom made his first U.S. speech at Internet Telephony Conference & EXPO, they launched a developer program. I recognized this as pure genius immediately.

You see, the future of VoIP is going to be in building ecosystems where many companies can make money. Think of this as a platform business. In other words, the future of VoIP is going to depend on interconnected parts and pieces to make large-scale solutions possible.

Skype gets this and is helping to further push their brand by making others rich in the process. Avaya understands this and their developer program is mission-critical to them. Inter-Tel gets this too, but doesn't have the resources to be Avaya — yet.

The future of VoIP is going to be in building ecosystems where many companies can make money.

One thing is for sure, the company that wins the PBX war is going to be the one with the most developers. Cisco knows this and Nortel does as well but Avaya is winning the race for the hearts and minds of new developers according to my informal polling at recent industry events.

This is important because the future of the PBX may be similar to that of the PC. The company with the largest developer program wins. What sorts of development are we talking about? Vertical applications such as legal, real-estate, hospitality, and so on… You see a company in a certain niche in the market will always want a phone system customized to what they do. It is that simple. This brings us back to the size of the developer program.

The result of the VoIP developer wars, as I call them, is more choice for companies looking for solutions. Everyone wants to feel special. Everyone wants to think their software and in fact all products they purchase are designed to best suit their needs. The companies that get this are the ones that will win tomorrow's VoIP wars for sure.

Services Over IP

One of the more interesting comments I heard at the VoIP Developer Conference was from Michael Stanford, Director, VoIP (define - news - alert) Strategy, Digital Enterprise Group at Intel who said "VoIP is old news," said Stanford. "Long Live SoIP, or Services over IP." He went on to explain that VoIP is merely the first drop in the coming deluge, the first significant application over IP, and that we have already moved beyond simply seeking to offer cheap minutes.

Stanford offered a veritable laundry list of promising numbers from a variety of research analysts. Cell phone lines are far outstripping fixed lines globally. Dual mode shipments are slated to grow beyond 100 million units shipped by 2010. Data-capable phones will overtake voice-only phones by 2008. (Today's conventional wisdom dictates that virtually all cell phones will be smart phones soon, yet there is starting to be some pushback from the cell phone industry. People want simplicity, and perhaps the increasing amount of services may stall sales.)

Broadband subscriptions are growing, with an expected 300 million broadband users globally expected by 2008. To keep up with service evolution and advent of IPTV, as well as other applications, we will need 10MB+ by 2008.

Stanford mentioned that WiMAX (define - news - alert) field trials have begun with over 100 trials being conducted

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globally today. He expects to see notebook integration by next year, and the first WiMAX phones and mobile network roll-out by 2008. By 2008-10 we should see global network roll-out.

Stanford wrapped up his presentation by reminding everyone that VoIP is the beachhead to services over IP, and that the technologies underlying VoIP (RTP, SIP QoS, IMS…) support more than just voice. VoIP is the baseline feature that opens the door to new services such as wideband audio, video, IM, push to talk, presence, document sharing, rich collaboration, and the further rapid innovation of new features and service combinations.

The opportunity, according to Stanford, is New Services and New Applications. SoIP. Services over IP. Any access connects you to any service. Opportunities abound for developers.

What amazed me most about the keynote was how many people were asking about WiMAX after Stanford finished. This is not surprising as right around the same time as the show, Vonage announced they are working with a company called TowerStream to deploy VoIPoWiMAX (this is a brand new acronym I just made up!) Check out http://tmcnet.com/154.1 for more on this alliance.

I moderated an interesting panel on HMP or host media processing that included David Duffet who heads up Aculab Academy; Alan Percy, Director of Business Development, AudioCodes; Amir Zmora, VP Marketing and Product Management, Surf Communication Solutions; Greg Pisano, Market Development Manager for Carrier Enhanced Services, Brooktrout Technologies; and Daniel LeCour, Vice President of Business Development, Envox Worldwide.

We had a tremendous amount of expertise up on stage and what struck me as the panel proceeded is how complex it is to make a decision as to whether to use host media processing or use boards in your applications. The one real obvious point I came away with is that HMP is a great solution for small-scale projects but there are almost no service provider solutions that are better served with an HMP solution.

If you have a medium-scale application you really need to do some intricate calculations to figure out which solution is most cost-effective. Interestingly the panelists kept throwing around Moore's law and used it as an argument to justify going HMP. I countered that DSPs also evolve according to Moore's law. The panel agreed and actually said a common problem they see in the market is that users don't realize that Moore's Law also applies to DSPs.

Interestingly David Duffet made sure the audience was well-aware of what he coined as Duffet's Law, which says that the faster processors get the more that is demanded of them. As an example he mentioned that some service providers in South America are running VoIP on low-quality lines and as such now need the DSPs to do extremely intensive echo-cancellation.

Speaking of having more demands on your server, it seemed unanimous that when you deal with video, the HMP solution gets expensive fast. By the time you factor in licenses for the OS and space in a cage, the costs add up quite quickly.

Mark Spencer, the founder of Digium and creator of the Asterisk open-source PBX spoke to a standing room only crowd and fielded a bunch of questions about the Asterisk platform. One of the more entertaining aspects of the conference sessions was a snippet of code Mark showed that was written by a friend. The code was aimed at eliminating contact with an ex-girlfriend and it basically connected calls from the ex-girlfriend's cell phone to her home phone and vice-versa. This apparently resonated with crowd based on the laughter in the audience.

I later heard a story from someone else who had found out their spouse was beginning to develop a relationship with someone else (let's call this person Pat) and subsequently they programmed their home-PBX (It amazes me how many people have PBXs in their houses these days) to do the following. Regardless of who called whether the spouse called Pat or Pat called the spouse, the calls were rerouted to a PBX extension that never picked up. Apparently this solution helped end the relationship. I tell you these VoIP developers are very creative. If you could shrink-wrap some of these, there is no telling how much money can be made.

Opportunities abound for developers.

Secure VoIP

It may come as a surprise to learn that not all corporate security issues originate from outside the network. A growing number of threats are coming from inside networks. In fact some security analysts have found that there are more successful attacks coming from within than outside your network. Cyber criminals working inside a company can steal the identities of coworkers on a large scale. Usually these criminals are difficult to spot. They don't look like criminals.

This concept is not lost on crime syndicates who equip members of their organizations with fake identification and send them to apply for and obtain jobs in Fortune class corporations. Once inside, these criminals have access to the inside of networks — on the "soft" side of the firewall — and from there can unleash an array of internal attacks on a company's computer network in an effort to steal identities.

Other times, such syndicates direct their efforts towards the corporation and in some cases are able to steal money from banks and other financial institutions. When they are caught it is often too late to get the money back.

Enter VoIP.

Internal and external hackers are always looking for new ways to get confidential information they can use to make money from. Techniques such as VoIP eavesdropping allow a hacker to listen-in on phone calls. Conversations with banks where PIN codes are used can be saved for later analysis. Think of unprotected VoIP networks as a hacker gold rush.

VoIP encryption is one way to deal with the problem and if you read Internet news sites like TMCnet you probably have noticed a growing trend by companies to encrypt voice packets. The problem is that encrypting voice packets in a way that an enterprise cannot unencrypt them causes problems for law enforcement agencies as well as the corporation. Skype is

an example of a product that cannot be centrally unencrypted. HIPAA and Sarbanes Oxley are two laws that require corporations to record certain employee conversations in order to be in compliance with the law.

Phil Edholm the CTO of Nortel Networks told me a while back how concerned he is about peer to peer encryption of SIP messages as these messages can contain viruses and other malicious code. Encrypting SIP messages on a peer to peer network can lead to absolute disaster if you aren't careful. If all of these problems aren't bad enough, there are issues relating to latency caused by encryption you also have to deal with.

Encryption — unless it is centralized — is a bad idea for enterprise VoIP. In this world where security is so important to all of us, any sort of p2p VoIP security protocol that governments can't break is bad news for the population as a whole. Rich Mendoza, the Managing Director of SIP Solutions at BorderWare tells me that the firewall, not encryption, is going to have to deal with the VoIP security issues and you know what? He thinks we will need specific firewalls for various applications such as e-mail. If Rich is correct, the more applications you have, the more firewalls you will need.

As Rich tells me, general purpose firewalls don't generally do the deep packet inspection needed to protect organizations using VoIP. He goes on to say that service providers aren't in the packet censorship business, this is why we have desktop applications for antivirus, anti-spam, and anti-spyware (he forgot anti-adware — how can anyone live without this?) We need endpoint applications to protect VoIP calls.

BorderWare sells firewalls so they are obviously biased towards firewall use. Their appliances sit behind your general purpose firewall and when deploying you just open up ports 5060 and 5062 on your general firewall so that the VoIP firewall can handle the job of dealing with the VoIP traffic. Another application of these firewalls is deployment by service providers to their customers so that the softswitch is segregated and customers can't get to other customers on the same softswitch.

This past December 2004 I wrote the following about VoIP E911 in Internet Telephony Magazine:

VoIP E911

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

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

I consider this a stumbling block that needs addressing on our way to achieving VoIP 2.0. Companies like Vonage, who use

On DSL Deregulation

The following blog entry (tmcnet.com/155.1) was written by me on my blog at tehrani.com a day before the FCC "deregulated" DSL (tmcnet.com/156.1).

The FCC Will Bother Me

If you are a CLEC, the FCC will certainly bother you and that bothering may come sooner than you would like as the Wall Street Journal reported in an August 3 article titled *FCC May Set Rules Allowing Bells Exclusive Access Over DSL Lines*. The article said that Chairman Martin, who many people say looks like Harry Potter, plans on freeing up the LECs from having to share their networks so they may compete more favorably with the cable companies. The FCC was going to do this eventually. We all knew it was coming. I am just so saddened at the whole incident; I don't know what to do but rant. Consumers are going to lose big time when the decision is made... Whenever that is.

LECs sob out loud and proclaim cable companies have an advantage over them in providing broadband as cablecos are unregulated. Lost in this argument is the Greek-god like levels of incompetence that have been a hallmark of the local exchange carriers for years. Does anyone other than me remember ISDN, the original broadband solution? This was a technology that worked and it has been around for well over a decade.

You know why LECs couldn't roll ISDN out successfully? There is no technical reason except for a lack of common sense, the inability to execute a business plan that didn't have to do with selling voice and finally, they were just too fat and happy to have to worry about it.

These are the same companies we are now helping. Spoon feeding in fact.

Here is a chilling paragraph from the Wall Street story that should make the average consumer's blood boil:

*The change, which likely would take effect this fall, would allow phone companies to kick competitors such as **EarthLink Inc.** and **America Online**, a division of **Time Warner Inc.**, off their DSL systems. If independent Internet providers can't reach terms with phone or cable companies, they could be forced to either focus on providing dial-up Internet service or emerging technologies such as high-speed wireless Internet.*

Now if you missed this point... The U.S. is rapidly losing the broadband race to over a dozen other countries. We are not deploying fast enough. The few solid competitors that we do have who have innovated and given consumers real broadband choice with excellent service are about to be run out of town.

This wouldn't be so bad except for the fact that the FCC is eliminating competitors to improve the competitive landscape. I have said this before, I feel like I am in a Twilight Zone episode when I read statements like this. Somehow less competition is more.

If you really want universal broadband, give tax credits to the young hungry ISPs that move like lightning and can't wait to roll out the latest technology.

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technology from an innovative company called Intrado, are taking bold steps to ensure the safety of their customers. They should be commended for their efforts and others need to follow.

Few service providers listened. These providers are now scrambling to meet the FCC deadline for 911 compliance.

Now I'm at it again. I am telling you that if you are a service provider or an enterprise putting VoIP on your network, you need to understand the security implications of not having a VoIP aware firewall in the mix. Understand full well what you are doing.

If you are unsure, come to Internet Telephony Conference & Expo this October to learn everything you need to know to roll out VoIP safely and securely. We have extensive and in-depth education on this topic. Remember that if a 911 call doesn't work on your network because of some sort of attack, someone will be held responsible. The same goes for sales calls and revenue that may be lost. Be sure you know as much as you can and do as much as you can to ensure a successful and secure VoIP deployment. IT

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Simply stated, if you were going to bank on an airline to provide us with the future of high-speed transportation, would you bet on Southwest and Jet Blue or American and Delta? Entrenched players are not innovators. They are masters at gaming the system to their advantage and whining endlessly about how unfair their lives are.

Moreover entrenched cable companies and LECs are generally hated by customers because of their slow service. These companies should be the last to be rewarded, not the first.

LECs were notorious for doing everything they could both legal and illegal to kill the CLEC market. They seem to have finally succeeded.

Tomorrow could be the saddest day the U.S. broadband market has ever seen. In a few years I predict we will slip further behind in the broadband race. At this time it will become abundantly clear that killing off broadband competitors to help companies that are universally hated and have a poor track record in innovating was a bad idea.

I truly hope I am way off base and the U.S. does take its rightful place as the world's broadband leader. Hopefully Chairman Martin knows a magic spell or incantation that will ensure we can all communicate at 1 gigabit per second in the near future. IT

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NEWS

Vodavi Expands, Enhances XTS-IP Apps

Vodavi Technology, Inc., (news - alert) announced enhanced IP capability for the XTS-IP and XTSc-IP converged family of communications products. Targeted at telecommuter, remote, and traveling worker applications, the XTS-IP platform offers enterprise customers the ability to extend additional IP services to remote locations via a remote IP services gateway, deploy IP soft phones to laptop PCs and take advantage of increased IP system capacities. These capabilities help to maximize business productivity and cost saving advantages when deploying IP communications.

According to Chet Lytle, President of Communications Diversified, Inc., a Vodavi Authorized Gold Dealer, "We believe Vodavi's value proposition for cost of technology is exceptional, specifically the cost to network multiple locations together using IP. The XTS-IP makes the decision to purchase even easier because customers have the option to start with a few VoIP applications while staying firmly grounded with traditional technology. Customers have the ultimate flexibility to choose their own technology path for meeting company goals."

The XTS-IP product family offers Vodavi customers the benefits of the following new features and devices:

Fully-Functional Host Extensions for Telecommuter and Remote Office Workers. The introduction of the XTS-IP product family brings to market the Nomad RSGM (Remote Services Gateway Module) that allows users at remote locations to become fully-functional extensions of the host telephone system.

IP Soft Phone for Traveling Worker/SOHO Applications. Designed for road warriors and SOHO (small office/home office) applications, Vodavi's new Nomad SP is a soft phone that operates as a fully functional extension of the host telephone system when used with a USB headset and a laptop PC.

Seamless Multi-Site, Multi-Applications Networking. Up to 32 XTS-IP and XTSc-IP converged telephone systems can network together for seamless, limitless opportunities for digital and IP voice communications.

http://www.vodavi.com

AdventNet Intros Zoho Virtual Office; Web Collaboration Software

AdventNet (news - alert)has announced the release of Zoho Virtual Office — a Web-based, groupware and collaboration software for Linux and Windows. Zoho Virtual Office applications include: Webmail, documents, calendar, tasks, notes, contacts, bookmarks, group discussions, instant messaging and many other useful features. It allows several concurrent users at different locations to create multiple groups, and facilitate the sharing of information from both inside and outside the company.

Information can be accessed from any computer that has an Internet connection and a Web browser, allowing teams to work securely as if they were in the same physical location. Individuals and groups can collect, communicate, organize, and share information, as the virtual collaboration platform is designed to deliver the functionality for many participants to augment a common deliverable. Virtual office runs on Windows and Linux machines, and is designed to increases employee productivity by allowing individuals and groups of users to seamlessly share information.

With Virtual Office, you can:

- Have easy access to information from anywhere.
- Instantly bring together virtual team members.
- Share, review and edit information securely.
- Communicate through instant messaging, discussion forums.
- Know which team members are online and offline.
- Track, manage and organize meeting and projects.
- Set task reminders and notifications.
- Create Notes to collect important information.
- Integrate e-mail, task, calendar and notes to form a seamless collaboration platform.

http://www.zoho.com

iPhone2 Prepares for Launch of VoIP Video Phone Service

iPhone2, Inc., (news - alert)the creator of a proprietary software video phone called ImagePhone, recently announced that the company has begun conducting an extensive beta test of its video phone and VoIP service. The service will initially be made available to a closed group of approximately 100 testers who have signed up for a pre-launch test.

iPhone2's ImagePhone is designed to be an extremely user friendly, feature-rich VoIP video phone service representing a new type of consumer VoIP offering that has the potential to revolutionize the way that people communicate. ImagePhone allows customers to make unlimited point-to-point voice and video calls, as well as make or receive voice calls from anywhere in the world right from their computer using iPhone2's proprietary Softphone technology. The Company's service offerings include unlimited Video/Voice Calls, caller ID, Call Forward, Voice mail and several other value added features.

"ImagePhone is a long anticipated product that broadband is helping make a reality. With thirty two million households in America as well as millions of households internationally already having broadband access, iPhone2 is well positioned to capitalize on the dynamic and fast growing consumer VoIP market," commented Chip Greenberg, President of iPhone2.

http://www.iPhone2.com

ATTENTION VENDORS!

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

FCC Deregulates DSL

By Ted Glanzer

In a highly anticipated move, the Federal Communications Commission recently voted unanimously to remove regulations that required legacy phone companies to provide network access to their competitors.

Specifically, the Commission ruled that DSL is an "information" service instead of a "telecommunications" service, essentially leveling the playing field with cable operators. While the distinction sounds rather arcane, the stakes behind the classification change are enormous.

First, RBOCs are no longer required to lease their DSL lines to non-affiliated ISPs.

Additionally, the new ruling will alter (reduce) how much of their DSL revenue RBOCs must contribute to the Universal Services Fund.

There is a "grandfather" provision that requires RBOCs to continue to provide network access to their competitors for one year. Also, RBOCs must contribute a portion of their DSL revenues to the USF based on current calculations for 270 days, or until the FCC adopts a new contribution standard.

To be sure, the rule change has not caught anyone off guard. Indeed, FCC Chairman Kevin Martin has pushed for DSL deregulation ever since the Supreme Court handed down its decision in the Brand X case in June. In that decision, the high court upheld the FCC's rule that cable companies are not required to share their broadband lines with competitors.

"[The new rule] ends the regulatory inequities that currently exist between cable and telephone companies in their provision of broadband Internet services," Martin said. "[L]eveling the playing field between these providers has been one of my highest priorities. . . . [T]he actions we take in this Order are an explicit recognition that the telecommunications marketplace that exists today is vastly different from the one governed by regulators over 30 years ago."

Even though the vote was unanimous, not every commissioner was completely sold on the sweeping change.

"Were the pen solely in my hand, this is not the Order I would have drafted or the procedural framework I would have chosen," Commissioner Jonathan S. Adelstein said. "This Order, however, reflects meaningful compromise by each of my colleagues, and I appreciate the efforts to address many of my concerns about issues including the stability of the Universal Services Fund, access for persons with disabilities, and the ability of competitive carriers to access essential input facilities."

Nevertheless, according to sources within the FCC, Martin and his fellow commissioners were so driven to level the broadband access playing field that staffers worked around the clock to prepare for the announcement.

While many have speculated that the new rule will spell the end of competition for broadband services, some industry analysts disagree.

"[W]ith growing pressure (both regulatory and market-driven) on RBOCs to proliferate 'naked' DSL as primary line erosion continues, the door will increasingly be open to non-facilities-based VoIP competitors," stated an analyst at Deutsche Bank.

http://www.fcc.gov

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Most vendors can't extend your contact center technology resources to service remote contact center locations and, at the same time, enable your local business units to manage their own technology-driven business processes. As a result, multi-site organizations are challenged to gain economies of scale by sharing IP-based infrastructure and software licenses across sites.

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StarVox Launches Next-Gen Voice Services Via Its VoIP Networks

By Ted Glanzer

StarVox Communications, Inc., (news - alert) a facilities-based service provider, announced the launch of its VoIP network for the delivery of next-generation voice services. StarVox works with a variety of channels (ISPs, CLECs, Telecom agents, carriers, affinity groups, etc.) that are interested in migrating customers from traditional separate voice and data communications applications to a converged voice/data connection.

The company, according to a statement, is in a unique position to do so quickly and at a low cost because it owns its core application technologies and the network used to deliver its applications.

"The advanced applications we are offering enable the channel partner to compete effectively against competitors offering next generation solutions," said Doug Zorn, StarVox CEO, in a prepared statement. "We are offering a product suite to both retail agents and wholesale partners who want to integrate the products with their existing solutions."

StarVox' domestic VoIP network has over 300 points of presence (POPs) nationwide reaching 80 percent of business customers. The network is comprised of redundant softswitch sites connected via a private IP VPN network backed up with a diverse MPLS-based ATM backbone.

Among the features contained in the domestic VoIP network are the following:

• Protocol mediation supports a variety of VoIP and PSTN protocols, including H.323, SIP, MGCP, SS7, C7, ISDN and others.

• VoIP origination enables service providers to deploy services including IP Centrex and Unified Communications applications for their customers, providing local and toll-free numbers;

• VoIP to VoIP Serves as an intermediary between non-related VoIP networks, completing calls in a native IP environment while maintaining carrier-grade QoS.

StarVox also owns and develops the following technologies:

• VoIP trunking including local, long distance, toll-free and international calling in metered, flat rate or bundled packages, including all traditional voice services, such as 911, directory assistance and local number portability.

• VoIP Virtual Private Network (VPN) hosted non-metered calling between various subscribers VoIP last mile sites;

• IP-based Unified Communications, including display of voice messages on e-mail screen, attachment of voice messages to e-mail, single number reach and other advanced features.

• Dedicated IP Centrex Service: VoIP network extended to the customer's premise and maintains a network-hosted phone system that eliminated the need for a customer premise-located PBX or key system.

"By owning the applications and the network, [StarVox] can achieve faster reaction to customer requests," Rich Barry, StarVox' vice president of marketing, told TMCnet in an interview.

Indeed, under normal circumstances service providers purchase applications from third party developers and offer network services dependent on these developers. This, according to the statement, typically results in long lead times to react to evolving customer needs and the ability to control vendor development priorities.

"Service providers don't have the subscribers that give them the power to persuade vendors to make them their number one priority," Barry said. "It's really hard to get a third party to change its priorities."

With the transition of corporate communications to a converged IP network, Barry said that business customers can expect the much faster development cycles associated with IP-based products versus the slow traditional deployment of new TDM-based voice services.

As a result of the forgoing, Barry said that he sees a major shift from TDM to VoIP connections. "Millions of lines are going to be switched," Barry said.

http://www.starvox.com

MCI Speeds Up Rollout Of Wholesale VoIP

By Robert Liu

Because of the current red-hot market demand for IP-based communications, MCI has accelerated its product development to roll out a wholesale Voice over Internet Protocol (VoIP) product suite earlier than initially planned.

MCI (quote - news - alert) recently announced immediate availability of the service that was originally set for later this year. The VoIP enablement services include Carrier IP Termination, available throughout the U.S., and SIP Gateway Service, available to roughly half of all U.S. business and residential customers.

The accelerated rollout comes at a time when more than half of all CIO's recently polled by Deutsche Bank stated they either have already deployed or were planning to deploy in the next 12 months some form of VoIP service. Meanwhile, on the consumer side, cable operators and independent service providers like Vonage and Skype are ramping up their marketing and sales efforts.

As the IP marketplace develops, MCI said it has adapted its approach to TDM functionality. The Carrier IP Termination product is ideal for customers who have already purchased media gateway equipment and desire to obtain the cost advantages of originating IP traffic and terminating that traffic over a fully integrated global network.

"MCI's wholesale VoIP enablement services provide our customers with a carrier of choice that owns and operates an IP-centric network necessary to compete in today's market," said John Krummel, senior vice president of MCI Wholesale Services, in a press statement.

To help support the new product suite, MCI has slightly re-engineered the sales process. Wholesale customers interested in getting into the VoIP market will now work with a dedicated VoIP sales team that includes senior management, sales professionals, specialized engineering technical consultants, and program implementation personnel, the company said.

http://www.mci.com

Portugal Mobile Operator Deploys IMS-based PoC Service

By Robert Liu

Motorola (quote - news - alert) has deployed a Push-To-Talk over Cellular (PoC) network for Optimus and the Portuguese mobile operator has signaled that its 2.1 million customers will have the ability for other IP Multimedia Subsystem (IMS) services in the near future, the handset maker announced recently.

Albeit small, Portugal's telecommunications market is among the most advanced in Western Europe. In mid-2005, mobile and Internet usage are among the highest in Europe and broadband adoption is far above the EU average. So it's no wonder that operators like Optimus, formed only in 1998, have embraced burgeoning new technologies like Motorola's IMS platform. This comes despite analysts' predictions that real-world IMS deployments won't like come until 2007 or 2008 in the U.S.

But while U.S. telecom execs continue to sort through the promise and the hype surrounding the emerging IMS standard, Optimus is now offering its subscribers the opportunity to make walkie-talkie style mobile conversations with individuals or groups of contacts. Optimus' subscribers will initially have a choice of PoC handsets; the Motorola V400p and Symbian Series 60 devices with the Motorola PoC Client. Depending on the success rate and market acceptance, Optimus executives believe more IMS services could be forthcoming down the road.

"'Push-to- x' applications are expected to develop considerably over the next few years with compelling roadmaps, for which Optimus wants to be ready," said Miguel Almeida, Chief Operational Officer for Optimus.

Motorola is far from the only handset maker that's embraced IMS-based PoC functionality. Nokia and Ericsson have also issued white papers detailing their Push-To-Talk over Cellular offerings. Unlike what Motorola manufactures for Nextel for its so-called "Direct Connect" service in the U.S., PoC is an IP-based technology that uses cellular access and radio resources as opposed to circuit-switched cellular services. The platform allows operators to not only deploy voice but other applications like Push-To-Text, Push-To-View, Push-To-Video, etc. But Optimus turned to Motorola because of existing relations — Motorola helped deploy the operator's 3G network in 2004.

http://www.motorola.com

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CallWave Launches First Prepaid VoIP Mobile Phone

By Ted Glanzer

People who either can't afford or don't want to commit to a long term cellular phone contract have a new option to obtain mobile service: a pay-as-you-go mobile phone that has VoIP features.

Indeed, CallWave, Inc., (news - alert) a provider of VoIP enhanced services for consumers and small businesses, has announced the availability of the CallWave Mobile prepaid cell phone, which, according to a press release, is the first pay-as you go phone service with VoIP enabled features.

According to the release, the VoIP features, which include Mobile Call Screening, Mobile Call Transfer, and Follow Me Home, are designed to assist customers to "stretch their airtime, control their costs and enjoy higher quality phone service."

For example, the Follow Me Home feature enables subscribers to receive their mobile calls on their home phone instead. FMH is automatically activated when the prepaid cell phone is turned off or if the phone is on and loses network coverage.

Mobile Call Screening allows users to listen to voice messages in real time; a user has the option of interrupting the message at any time to take the call.

Another feature, Mobile Call Transfer, permits users to instantly transfer a live cell phone call to a home or office phone. If a subscriber has run out of minutes, the service will still take a message and deliver it to the subscriber's e-mail or PC software.

Along with the VoIP features, the press release states that the first 1,000 customers to purchase the service at CallWavemobile.com will receive a free Nokia phone with 60 minutes.

All subscribers are provided AT&T nationwide coverage. Additionally, there are no charges for in-network roaming, in-network long distance, or service termination.

The service costs $3.95 per month; "refill" minutes cost extra, starting at $10 for 40 minutes.

http://www.callwavemobile.com

NetCentrex Adds To Its IMS Portfolio Expansion, Acquires NeoTIP

By Ted Glanzer

NetCentrex (news - alert) has moved a step closer to offering a complete IMS architecture by acquiring NeoTIP.

NetCentrex, Inc., a developer of next-generation converged voice and video solutions, added key component in its effort to offer a complete IP Multimedia Subsystem (IMS) architecture with the announcement that it entered into an agreement to acquire NeoTIP, a European-based supplier of session border control technology (SBC).

The two companies, both of which are headquartered in France, are "kindred spirits" with similar cultures and, therefore, aren't expected to experience the growing pains associated with many acquisitions, NetCentrex' Vice President of Marketing Brian Mahony told TMCnet in an interview.

It also didn't hurt that NeoTIP is profitable.

The crown jewel in the deal, however, is NeoTIP's SBC technology, which delivers security and quality assurance to VoIP and other services, according to the release. SBC will be folded into NetCentrex' IMS developments component (the Proxy Call Session Control Function in particular).

Managing network borders cannot be performed by regular network services such as routers, softswitches or firewalls, NeoTIP's Web site states.

"As IP technology is becoming the foundation for delivering advanced applications to consumers and enterprises, operators demand increased security and quality of service without increased complexity or cost," said Olivier Hersent, chairman and CTO of NetCentrex, in a press release.

Mahony said the acquisition is a natural progression for both companies in light of the rapid evolution of IMS.

"It just makes sense," Mahony said.

NetCentrex' full commitment to IMS puts the company in a favorable position to assist service providers in delivering data and fixed-mobile services to consumers and enterprises quickly and efficiently.

"NetCentrex is growing; we're waking up and being aggressive," Mahony said. "We're doing what we need to do to be a major player."

http://www.netcentrex.net

Zultys Intros WiFi VoIP Phone

By Johanne Torres

Gadget lovers rejoice! Zultys Technologies has announced that it will introduce the WIP 2 wireless IP phone. The WiFi VoIP device will be available this month. With this release Zultys joins a growing number of VoIP product and service providers that are currently designing such devices. The device is designed to allow users to take their VoIP service from a stationary PC to a more mobile environment.

"Nearly 80 percent of our customers have asked us about WiFi VoIP products," explained Iain Milnes, president of Zultys Technologies. (news - alert) "They tell us they would like the ability to take their desktop IP phone with them rather than have a second mobile unit with reduced capability. Other WiFi IP phones coming into the market today don't provide users with enough features or sufficient battery life. The WIP 2 excels in all areas."

The company had announced a desktop version of the WIP 2 wireless IP phone back in March. The wireless device has all of the features and functionality found in the company's desktop version such as open standards, Linux operating system and SIP compatibility. The device also features two call appearances and supports voice encryption, paging, three-way conferencing, presence, instant messaging (IM), and all other telephony functions commonly found in landline telephone services.

The phone's battery life lasts for four hours of continuous talk time and 12 hours of standby time. The phone also allows users to access functions like call hold, call transfer, and DND. The device bundles the company's jitter buffer, speaker-phone and full duplex acoustic echo cancellation (AEC).

The paging feature is supported on the MX family of IP PBX products from Zultys. The company explained how this feature worked in a news announcement released today. "Using a single button on the WIP 2, a user can initiate paging to a group, thereby emulating "push to talk" capabilities within the enterprise. Such a feature is invaluable in construction areas, medical facilities, or any other organizations where immediate communications is mandatory."

The WIP 2 is in trials now with quantity shipments beginning in September. Pricing for the WIP 2 will also be announced in September.

The announcement goes to show that WiFi IP phones are becoming increasingly popular, and industry insiders have started to notice the trend. In fact, according to a recent study conducted by Infonetics Research titled *WiFi Phones Biannual Worldwide Market Size and Forecast*, revenue for these devices totaled $54.7 million in 2004 and units totaled 143,000. The firm predicts a strong growth through 2009 as steady adoption of voice over WiFi continues.

http://www.zultys.com

WiFi VoIP Handset Market Expected To Grow Significantly

By Johanne Torres

Internet telephony industry insiders know that the technology buzz is all about the gadgets. This year I've noticed that developers have been pretty busy designing Internet telephony devices that let us take our VoIP calls wherever we go. After all, I can only hope that these devices will one day replace my expensive contract-bound cell phone. Analysts have noticed the trend and predict that mobile WiFi VoIP handsets' worldwide revenue is set to multiply at a rapid speed.

According to a recent study conducted by Infonetics Research titled *WiFi Phones Biannual Worldwide Market Size and Forecast*, revenue for these devices totaled $54.7 million in 2004 and units totaled 143,000. The firm predicts a strong growth through 2009 as steady adoption of voice over WiFi continues.

What's the latest in mobile devices you ask? I have noticed that developers are designing hybrid WiFi/cellular handsets. The devices roam between cellular and IP-based networks bundling the connections seamlessly to prevent phone call disruption. The study found that the revenue for these devices hit $6.7 million in 2004. Analysts believe that "WiFi capability will eventually become a common feature in cell phones, just as it is becoming standard in laptops today."

Research revealed that despite the WiFi VoIP handsets market being currently small; it is indeed one with great potential. Analysts believe that "in logistics and healthcare verticals in particular, VoWLAN is already gaining momentum and will become widespread throughout the enterprise as VoIP and wireless LAN adoption continue."

The study also found that there is potential for notable growth in the consumer space, "as VoIP services and wireless gateways are bundled with a broadband connection. More dual-mode WiFi/cellular handsets will reach the market, enabling enterprise users to roam across 3G networks, home networks, corporate wireless LANs, and WiFi hotspots.

"Some challenges remain, such as QoS, roaming across different wireless platforms, and the relatively short range of WiFi signals, but with vendors currently addressing these issues, it is likely we'll be at the foot of the adoption bell-curve by mid-2006," concluded Richard Webb of Infonetics Research, author of the report.

http://www.info.infonetics.com

Trinity Convergence Announces Availability, New Customers Of VoIP Software

By Ted Glanzer

Trinity Convergence, (news - alert) a provider of voice and video over IP software solutions, announced the availability of VeriCall 4.0, a "turnkey voice-over-Internet-protocol software framework" for telecom infrastructure and equipment manufacturers.

According to the release, the new version offers equipment manufacturers a platform to develop connected and wireless VoIP products, such as "wireless gateways, integrated access devices, IP-PBXs and media gateways" and "delivers even more flexibility, enabling transcoding between any combination of TDM and IP networks." New enhancements to the software include the following:

- Support for multi-party conferencing and caller ID; and
- Integration with the Freescale MSC8122 digital signal processor.

Trinity wasted no time in announcing in a separate release that Pannaway Technologies, Inc. and Quantm Voice Systems are the first customers for the new VeriCall VoIP platform.

"The VeriCall framework supports Pannaway's desire to create the most comprehensive and robust platform available for delivering converged voice, video and data services over broadband," Kevin Brown, vice president of marketing for Pannaway, said in a prepared statement. "Our award-winning Broadband Access Switch family is a key component to our end-to-end IP platform and requires VoIP functionality that is capable of evolving with the needs and requirements of our customers."

http://www.trinityconvergence.com

RADVISION Teams With TTPCom, Adds Video Telephony To 3G

By Johanne Torres

Attention handset developers: if you are looking for a way to quickly integrate video telephony capabilities to your 3G designs, RADVISION (news - alert) and TTPCom (news - alert) might just have the solution. RADVISION, a protocol toolkit provider for developers made news as it joined TTPCom Ltd., a wireless terminal developer, for the companies to integrate video telephony capabilities into its AJAR mobile applications platform for wireless handsets using RADVISION's 3G-324M Toolkit.

TTPCom's AJAR allows handset manufacturers wanting to design low to high-end multimedia phones to quickly customize the end-user interface as well as implement third-party apps. The AJAR platform features include multimedia messaging, browser functionality, player/recorder/camera support, and intelligent text-entry.

"RADVISION is the acknowledged leader in Internet Protocol (IP) and 3G signaling protocol development kits," said Morten Iversen, director of partnership programs at TTPCom. "When we began looking for a partner to add high-end multimedia and video telephony capabilities to our AJAR handset development platform, RADVISION quickly rose to the top due to its field-proven technology, market leadership, and up-to-date, reliable, industry-standard protocols."

RADVISION's 3G-324M Toolkit includes features with the necessary capabilities to develop multimedia communication systems for 3G networks and terminals. The company said the toolkit is "fully compatible with 3G-324M enabled devices is optimized for minimal dynamic memory usage, yet offers outstanding performance."

Version 3.0 of the 3G-324M Toolkit is currently available worldwide.

"We are very happy to team with TTPCom and have our advanced 3G-324M technology integrated into AJAR," said Adi Paz, senior director of product management and marketing for RADVISION's Technology Business Unit. "TTPCom is rapidly gaining traction in the handset development market and its customers can gain a competitive advantage by quickly adding video telephony capabilities to their next-generation wireless handsets with AJAR."

This latest news follows RADVISION's recent announcement about the company releasing Version 2.5 of the RADVISION Session Initiation Protocol (SIP) Server Platform. The new release offers interfacing to external databases as well as enabling the development of presence servers and events packages. Other features include the SUBSCRIBE/NOTIFY mechanism and presence message format, Winfo, PUBLISH, and XML encoding/decoding.

http://www.radvision.com
http://www.ttpcom.com

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Empirix Unveils Voice Self-Serve App And VoIP Net Combo Tester

By Johanne Torres

Empirix, Inc., (news - alert)just unveiled the Hammer CallMaster 5.0, an automated testing tool for both voice self-service applications and VoIP networks. The new device tests voice and speech applications as well as VoIP environments, allowing users to launch higher quality apps in a short amount of time.

The company described the Hammer CallMaster as a "robust software solution that runs on top of Empirix' Hammer test systems to enable users to quickly and easily create test scenarios using a visual scripting interface. Those scripts can be deployed in a pre-deployment setting for testing, or in production to proactively monitor and manage application performance." Empirix claims that "users report that Hammer CallMaster simplifies the testing process considerably, thus lowering overall testing costs and enabling them to perform more testing in less time."

"Speech and VoIP technologies are becoming more prevalent in contact center environments, but as with any new technology, they both present Quality of Service challenges," said Wes Hand, product manager for the company's Contact Center Solutions Group. Automated testing from the caller perspective is the most efficient way to identify and correct customer-impacting issues. Hammer CallMaster reduces the most significant barrier to implementing automated testing by making it very easy to map out call flows and create test scripts.

The Hammer CallMaster has a visual interface which enables drag-and-drop, point-and-click test creation and execution, along with expanded reporting and analysis capabilities for quicker identification of problem sources. The tool features an Integrated Grammar Administration, which allows IT personnel to remotely manage both test scripts and supporting grammars from a single remote desktop interface; enhanced scheduling capabilities for monitoring based on time-of-day, and day-of-week; support for Windows 2003; and enhanced ISDN support, which provides extended programming capabilities for testing.

The Hammer CallMaster is available starting at $25,000 per seat plus $25,000 per server.

http://www.empirix.com

GL Launches PPP Protocol Analyzer

GL Communications, Inc., (news - alert) a leading provider of test and measurement products for the telecom industry, announced the release of a new protocol analyzer, PPP Protocol Analyzer, designed to capture and analyze various PPP protocols over IP network.

The Point-to-Point protocol (PPP) is a link layer protocol, which is generally used to establish a direct connection between two network nodes over a serial link. Today, the PPP protocol standard finds wide use in synchronous connections between LANs, bridges, routers, and other intermediate devices.

GL's PPP Protocol Analyzer is designed to capture and analyze a suite of PPP protocols such as LCP, NCP, PPP BPDU, PAP, CHAP, HTTP, SNMP, FTP, DNS, and DHCP. The tool allows a test engineer to monitor, capture, and perform numerous measurements across WAN-LAN or LAN-LAN connections. It also includes the ability to test and analyze HDLC-based PPP protocols in a synchronous environment.

Some of the main features of PPP Analyzer are listed below:

- Compatible with Windows 2000/XP operating systems with user friendly GUIs.
- Works with GL's field proven Ultra E1 or T1 internal cards or Laptop E1 or T1 external units. Other interfaces also available soon.
- Supports analysis in real-time as well as offline.
- Supports both PPP routed and PPP bridged protocols.
- Supports data transmissions on single channel, sub channels or hyper-channels.
- Allows exporting detailed information to an ASCII file: all captured frames or only filtered frames defined by a comprehensive filter criterion and raw frame data as hexadecimal and ASCII octet dump.
- Allows exporting summary information to a comma-delimited file for subsequent import into a database or spreadsheet.
- Supports statistics display based on frame count, byte count, frames/sec, bytes/sec, and more for the entire capture data.
- Supports call records and message trace to be monitored remotely and centrally using NetSurveyor.

http://www.gl.com


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Network Timers
Speech Quality
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Voice Path Delay
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Pandora Networks Chooses Sangoma

Sangoma Technologies Corporation (news - alert) announced Pandora Networks has selected Sangoma's AFT Series of TDM hardware as an integral part of its On Demand IP communications solution.

Capable of handling both voice and data and supporting all popular open source projects, Sangoma's AFT Cards are designed for performance, reliability, compatibility, support, and ease of installation. With less demand on the host CPU, drivers take advantage of the AFT technology to substantially reduce the processing required to handle TDM voice calls. This reduces the CPU's workload and results in fewer dropped calls, less jitter and better voice quality for callers.

"Sangoma's family of AFT cards are among the most robust and complex the industry has to offer – the result of our more than two decades in LAN/WAN networking and numerous hundred thousands produced and successfully installed," says Sangoma Technologies President and CEO David Mandelstam. "That Pandora has selected our hardware after a meticulous evaluation of the marketplace is proof-positive of our four tenets for which we are becoming increasingly known: quality; compatibility; performance and unparalleled support."

Some of the AFT card features include:

• ***Compatible and Flexible***. Sangoma's voice/data cards are self-sensing for 3.3v and 5v PCI slots and software configurable for T1, E1 or J1. They share interrupts properly between themselves and other PCI compatible devices, supporting unlimited numbers of cards per PC chassis. Conforming to the 2U form factor, both in height and length, the AFT cards allow users to install many cards in a slimline 2U chassis to maximize server capacity.

• ***High performance***. Sangoma cards have been carefully designed to reduce CPU loads in TDM environments, improving system performance and reliability on larger systems.

• ***Trust and quality***. As part of its commitment to quality assurance, each Sangoma card is individually inspected and burned-in prior to shipment in protective anti-static wrap complete with cables, manuals and CDs. The result is an almost zero dead-on-arrival rate and high reliability in service.

• ***Unparalleled support***. Sangoma's fully engaged engineering support for both its hardware and software is unrivalled in the industry.

http://www.sangoma.com
http://www.pandoranetworks.com

TelTel Unveils New SIP-Based Public Internet Telephony

TelTel, (news - alert) a provider of SIP-based global Internet telephony with presence-enabled features, announced the availability of a new Public SIP Telephone Network (PsipTN). While maintaining an interface with the PSTN, the PsipTN powered by TelTel is designed to enable a new generation of SIP-based applications and services, and create new revenue opportunities for service providers and application developers. Unlike a traditional PSTN that only handles voice traffic, the new SIP-based virtual network is capable of carrying voice, multimedia, and audio/video content.

TelTel is also using the SIP protocol to enable TelTel users, service providers, application developers, and product vendors to join together to take advantage of SIP capabilities to enable communication and commerce. By taking advantage of the open SIP standard, new value-added services and products will be regularly available to TelTel's rapidly growing community.

The new platform provide numerous commerce and co-branding opportunities such as customizable softphones and Web pages, billing, call center, and provisioning applications, as well as information/entertainment media channels for devices and services. For example, an Internet service provider can offer customers VoIP calls under their own brand and use PsipTN to route the calls and provide billing services. Additionally, vendors can offer their customers so-called "TelTel Ready" handsets and personalized ring tones.

"The goal of TelTel is to fulfill the great potential of the open SIP standard." said Benedict Tse, VP of Product Management of TelTel. "PsipTN creates an environment that fosters and facilitates innovation, as well as revenue potential for developers and service providers."

http://www.teltel.com

Tekelec Expands IMS Portfolio With Acquisition Of SIP Routing Provider iptelorg

Tekelec (news - alert) announced they have purchased German and Czech-based iptelorg GmbH, developer of leading-edge Session Initiation Protocol (SIP) routing software, securing a critical IP Multimedia Subsystem (IMS) capability for Tekelec.

"We've been collaborating with iptelorg for the past year because of the fast, purpose-built Call Session Control Function [CSCF] capabilities of its SIP Express Router," said Fred Lax, Tekelec CEO. "What became clear to us was that we wanted to do more than license this technology — we wanted to own it, build on it, and make it a central part of our IMS strategy to bring the same level of carrier-grade reliability, scalability and innovation to our customers' rapidly growing SIP signaling needs that we provide for their SS7 networks."

"The fact that IMS networks are very signaling intensive plays to our strength," Lax continued. "Integrating SIP-signaling applications with our SS7 applications provides a perfect complement to our portfolio, supporting customers with legacy and next-gen environments alike. We help operators evolve their networks at the pace and scale that make sense for their businesses."

http://www.tekelec.com

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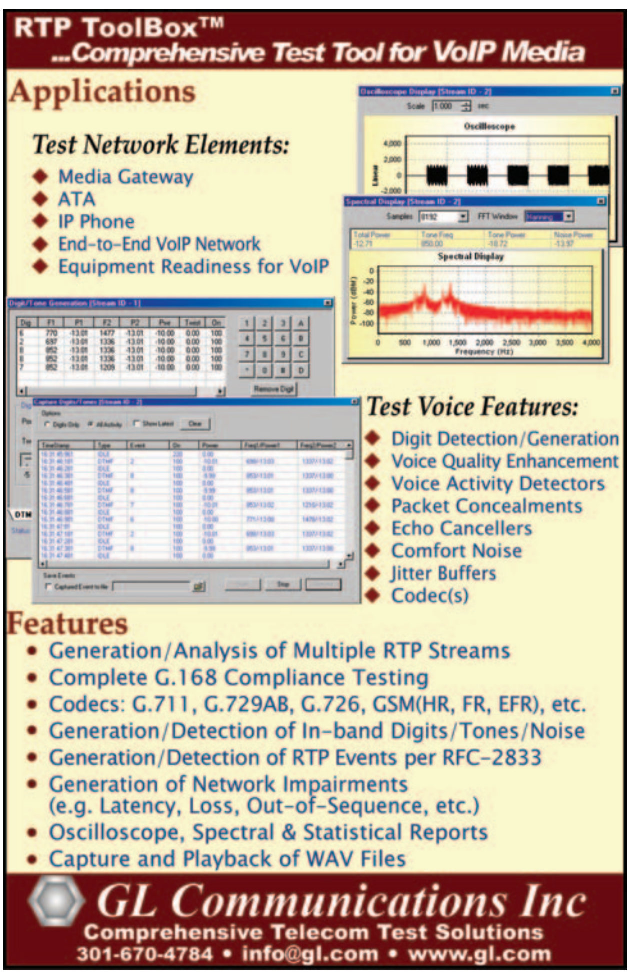
Eicon's Diva Server To Enable SIP-based VoIP Telephony

Eicon Networks Corporation (news - alert) announced recently that it will provide SIP-based gateway software to enable VoIP calls to easily traverse between Microsoft Office Communicator 2005 and legacy PBX systems by using Diva Server telephony adapters and Microsoft Office real-time collaboration (RTC) applications. The SIP-based gateway software is designed to enable users to seamlessly make PC-to-phone and phone-to-PC audio calls through the standard public switched telephone network (PSTN).

Microsoft Live Communications Server 2005 is a real-time collaboration platform that enables people to connect and work together in real-time using web and data conferencing along with instant messaging and audio/video collaboration. It supports linking corporate telephone systems and computers together to integrate private branch exchanges (PBX) and the PSTN with an IP network. This requires an IP-to-PSTN Gateway to allow users to make and receive calls to and from enterprise-wide PBX extensions or external PSTN phone numbers using Communicator 2005.

The forthcoming SIP-based gateway software from Eicon adds this IP-to-PSTN gateway functionality to any Diva Server telephony adapter (PCI-board) and this can be installed into a server along with Live Communications Server 2005 to provide an integrated gateway. This allows both PC-to-PC as well as PC-to-phone and phone-to-PC voice calls.

http://www.eicon.com/divaserver



FrontRange Releases IPCC 5.0

By David Sims

FrontRange Solutions USA, Inc., (news - alert) a service management, CRM, and voice application vendor is announcing the availability of IP Contact Center 5.0.

The new version of FrontRange's Voice Over Internet Protocol-based software suite features quality management and integration with other FrontRange product "families," including the company's HEAT, GoldMine Corporate Edition and the new IT Service Management modules.

The release is pitched squarely at "contact center managers" who "continue to scrutinize and manage costs meticulously," emphasizing its integration to business applications, which reduces both implementation time and costs.

FrontRange IPCC 5.0 is designed, company officials say, to "enable users to train staff more effectively with features like the optional module quality management" with which a supervisor can record calls with server based recording and call rating.

It's a growing field: The Pelorous Group recently released a study predicting that the total global market for contact center recording systems will grow from $487.9 million in 2004 to $705.8 million in 2009. Sales for selective recording systems will flatten and full recording systems increase due, primarily, to "declining storage costs, legal requirements, and contact center needs to mine deeper databases to find root causes of performance deviations," the group predicts.

Kevin J. Smith, FrontRange Vice President of Products called 5.0's "rapid deployment" and "reduced implementation" key benefits for their target demographic: "Customers want a solution that is easy to use without sacrificing power and flexibility."

Other product features include the quality management module, which allows supervisors options of one-way listening, coaching agent without customer hearing it or conference participation. There's also a business application integration which is designed to lessen the requirement for middleware and costly professional services.

http://www.frontrange.com

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Pillar Data Systems Chooses Nuasis IP-based Contact Center

By Johanne Torres

IP contact center provider Nuasis Corporation (news - alert) announced it was picked by Pillar Data Systems, (news - alert) a provider of enterprise storage systems, to deploy its Nuasis NuContact Center for Pillar's customer service contact center operation.

"Our decision to purchase the NuContact Center was based on our desire to have the right technology in place from the start. The IP-based system gives us the flexibility to customize the system to fit our needs. Whether it be fulfilling a disaster recovery requirement or establishing 24/7 availability for our customer service center, Nuasis has created a product that empowers its customers to meet their business goals," said Dave McCroskey, vice president of Customer Service, Pillar Data Systems.

The hub-and-node architecture of the Nuasis system contains failover and bypass capabilities to protect against the loss of a voice call under a variety of outage scenarios. According to the company's announcement, "if communication is lost between a hub and node or if a node experiences a network or PC outage, calls in progress and in queue are automatically redirected to another node on the network." For this particular deployment, Pillar's main node of their Nuasis system is located at their primary customer service contact center in California while the failover or second node is located at the company's development center in Colorado.

http://www.pillardata.com
http://www.nuasis.com

Salesnet Announces 'Significant' Upgrade

By David Sims

Salesnet, (news - alert) a provider of on-demand CRM software, is announcing what company officials call a "significant" upgrade to its enterprise product offering, introducing "over 50 product enhancements.

These new product additions are a prelude to Salesnet's upcoming 25th Anniversary Edition, anticipated for a winter release, which will include over 250 wide ranging features that, in the words of a company spokesman "broaden Salesnet's CRM footprint into Campaign Management, Lead Management, Product Catalogs, Order Management, and much more."

Jonathan Tang, President and Co-Founder at Salesnet, promised the so-called "Anniversary enhancement" will be "the most extensive enhancement in the company's history."

In this current release, geared towards enterprise customers, Salesnet is introducing over 50 direct feature product enhancements to its flagship on-demand CRM product.

Key new product enhancements include:

- ***Deal Fields in Communication Manager***. This allows users to create proposals, quotes, and opportunity summary documents using the Salesnet MS Word integration, at four times the speed.
- ***Custom Field Enhancements***. These allow finer-grain control of custom field attributes: user/position-based default values, user/position-based access to individual dropdown field values.
- ***Reporting Privileges***. These include the ability to control multiple levels of access to the reporting engine.
- ***Deal Import Wizard***. This has the capability to create and import deal records, extensive to round-robin 'deals' to sales groups within the organization and newly added ability to "stagger" deal creation across a period of time.

http://www.salesnet.com

Popular Telephony's Peerio Signs An SMB Deal With Vocalpad

By David Sims

Popular Telephony, (news - alert) provider of "serverless peer-to-peer communications products," has signed a distribution and licensing agreement with Vocalpad, a provider of VoIP services to establish the VAR channel for SMBs in Spain, South Asia, Middle East and African countries.

There's also a licensing deal to integrate Peerio in Vocalpad's core product lines. Popular Telephony's Peerio products will be part of Vocalpad's product portfolio, offering SMBs serverless end-to-end peer-to-peer with the direct connectivity to PSTN and mobile networks.

Vocalpad offers long-distance and international carrier services with access to VoIP products and technical and customer support.

Hassan Ghandour, the CEO and founder of Vocalpad, says Peerio's "unique serverless design" opened up "a whole new outlook that will potentially impact our corporate offering," as well as their Web-based and device-based services.

Peerio serverless middleware resides in the endpoints (PCs with PeerioBiz soft-phone or Peerio-embedded IP phones) connected over a corporate LAN. Peerio-intelligent endpoints collectively create an enterprise telephony system, providing traditional PBX features along with modern functionalities.

You can make calls with it on Baby Bell POTS as well.

Specializing in the VoIP connectivity and system integration services for SMBs, Vocalpad is currently integrating the PeerioBiz serverless soft phone in a number of financial institutions in Europe and the Middle East. Vocalpad plans to use Peerio SDK to develop and customize functionalities specifically tailored for its customers, as well as a variety of custom interfaces to the PeerioBiz serverless soft phone system.

http://www.populartelephony.com
http://www.vocalpad.com

GIPS Joins Symbian Platinum Partner Program

Global IP Sound (GIPS) (news - alert) announced that it has joined the Symbian Platinum Program to support the growing market for smartphones based on Symbian OS.

As a Platinum Partner, GIPS is extending its VoiceEngine Mobile platform to the Symbian ecosystem. Symbian develops and licenses Symbian OS, the operating system that powers today's most popular smartphones. Symbian OS is licensed by the world's leading mobile phone manufacturers. To date, more than 32 million Symbian OS phones have shipped to over 200 network operators worldwide.

"This partnership reflects our commitment to the Mobile Voice- over-IP market and our intent to build upon our mobile VoIP strategy," said Gary Hermansen, President and CEO of Global IP Sound. "By becoming a Symbian Platinum Partner, we can further expand our VoiceEngine Mobile platform and strengthen our relationships with users, application developers, smartphone manufacturers and network operators."

VoiceEngine Mobile for Symbian OS empowers OEMs to extend the mobility of VoIP to end users. The platform helps these companies create products that are easy to use and provide better-than PSTN voice quality even without in-house Voice-over-IP expertise. VoiceEngine Mobile allows manufacturers of mobile, IP-enabled products to capture all of the benefits of GIPS VoiceEngine technology-superior voice quality, accelerated product development and technology specifically designed for packet networks.

http://www.symbian.com
http://www.globalipsound.com

Voda One Streamlines Delivery Of IP Telephony Solutions

Voda One, (news - alert) a specialty distributor of networking and communications technology solutions and a division of the multi-national Westcon Group, Inc., announced that it is serving its enterprise customers more efficiently by handling final channel assembly on the Avaya Modular Messaging product. By performing these functions at the distributor level, Voda One is able to deliver IP telephony solutions to its enterprise customers by up to 75 percent faster than previously possible.

By bringing Avaya Modular Messaging products to market faster, Voda One enables BusinessPartners to realize revenue recognition on an accelerated basis. Voda One already performs final channel assembly for Avaya Communication Manager IP telephony and TDM servers.

Robert Linder, vice president of sales for Voda One, commented: "Avaya's new program helps us make our BusinessPartners more competitive by enabling us to deploy complex convergence solutions on an expedited basis. These solutions, which are assembled in a dedicated integration facility in our warehouse, are totally customized to fit the need."

http://www.vodaone.com

By Marc Robins

Pushing The VoIP Envelope

Summer is a great time for reflection, for mulling about where you've been, where you're going, and setting goals for the coming, cooler months ahead. In these dog days of summer, I've had some time to reflect on where we are as an industry, and where the currents and eddies of technological progress are taking us.

As a technology and as an industry, IP telephony has come an astoundingly long way in the decade or so since the road of mass commercialization was taken. It's gotten to the point where the quality and reliability of a broadband IP call is pretty much on par with regular POTS (define - news - alert) service, and most IP-based enterprise solutions are gaining the degree of end-user confidence typically afforded to legacy TDM (define - news - alert) equipment.

But herein lies the rub: it seems to me that much of the work up until now has been to create "functional equity" with TDM technology, and in effect replicate the telephony features and functions we've been "enjoying" for the past numerous decades. IP-PBX engineers have gone to great pains to duplicate the hundreds of features of their companies' TDM-based gear, employing SIP and proprietary SIP extensions to get them there. VoIP service providers, meanwhile, have succeeded in replicating most of the Class 5 features we all take for granted.

Forgive me for saying this, but I have to question the point of "going IP" if the switch or migration is going to simply replicate what we already have in the TDM domain. I know much of the justification for an enterprise migration usually revolves around cost savings and various networking efficiencies — and these reasons are certainly not insignificant. But in order to realize true, new value, users have to go beyond the ordinary and start taking advantage of new applications that rely on the power and flexibility of IP communications — and have the potential to catalyze deep, positive and fundamental changes in business process.

Of course, there are a slew of new IP-centric enhanced features and services that have been developed — in fact, most are quite near and dear to my heart — and I applaud those vendors and providers that have developed them. However, among many users, there seems to be a hesitation, or timidity, to embrace the newer, more innovative capabilities of the technology. Perhaps it's a matter of users moving up the learning curve, and gaining a good comfort level with the technology before they start extending beyond the tried and true.

So what are the new apps and capabilities that get me excited and glad to be part of such a dynamic industry? It's a long list and I won't be able to run through them all here, but here's a short list of some of my favorites:

Knowledge Management. A few companies, such as Mitel Networks, have a nifty app that can link an incoming or outgoing call to information resources that reside on a user's PC or anywhere in the corporate network. Think of it as CTI (Computer Telephony Integration) on steroids, allowing the automatic search and retrieval of documents, spreadsheets, e-mail messages, etc., that relate to the person at the other end of the line.

Mobility Solutions. What I have in mind here goes beyond the inherent "number portability" feature of VoIP, which involves the ability to have an office extension or home number "travel" with a user to remote locations, allowing seamless connectivity to corporate communications resources. WiFi telephony certainly rates as a killer mobility app. But beyond basic WiFi telephony are emerging solutions that meld wireline and wireless resources into a seamless system that keeps a user connected to the network whether he's roaming around town or the corporate campus with a multimode mobile phone, PDA or laptop.

Supply Chain Integration. A number of innovative players are developing apps that link supply chain "events" with communications resources to generate powerful, new business opportunities. One such company, Texas-based Ipcelerate (www.ipcelerate.com) is developing ways to leverage supply chain RFID technology to trigger a variety of notifications. For example, new DVD players (or any other item) arriving in port, or in a store, can be scanned, and customers on a waiting list can be automatically notified that their order is ready for shipment or pickup.

Presence Power. Leveraging IM buddy lists and a host of presence-based collaboration apps promises to drive a whole new level of efficiency and productivity for today's workers. Think solutions like Siemens' OpenScape, which takes collaborative computing to the next level by offering intelligent, real-time access to people, calendars, and files through presence-based communication and multi-resource collaboration.

Security. Foremost in the minds of most people today, IP telephony can provide an important "platform" for tying video surveillance and variety of sensors into corporate-wide communication resources, in effect leveraging the eyes and ears of all employees of a company.

We've seen easy savings come from reduced networking costs, free or reduced long distance charges, the elimination of expensive audio and video conferencing, lower real-estate and travel costs, and other operational efficiencies IP telephony enables. But perhaps the greatest value of IP telephony is its use as a strategic business tool. It's clear that we've only scratched the surface of what's possible. If you have a favorite "VoIP 3.0" app, or know of an intrepid user taking things to the next level, I'm all ears. IT

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

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By Tony Rybczynski

IP Telephony Across The Public Cloud

Meeting IP telephony QoS, security, and reliability requirements across public packet networks requires special attention. While leased lines are always an option to interconnect sites, virtual private lines using frame relay, ATM, and — increasingly — IP VPNs (including MPLS or Multi Protocol Label Switching) and optical Ethernet offer better price/performance. What are your challenges in handling IP telephony across the WAN?

Engineering The Bandwidth

Unlike LANs, MAN and WAN bandwidth carries a monthly recurring cost. QoS allows the enterprise to use expensive WAN bandwidth most cost-effectively. Traditional voice engineering methods can be used to determine the number of voice and fax calls that need to be handled over the WAN link, factoring in communities of interest, the number of busy hour call attempts, and the average call holding times. The bandwidth required for voice and fax calls can then be calculated, dependent on the codecs used. Over highly-utilized high-speed links, up to 85 percent of the available bandwidth can be used for voice traffic. For low-bandwidth (<1 Mbps) connections, no more than 50 percent of the available bandwidth should be used for voice traffic. This minimizes the maximum queuing delay that the VoIP traffic experiences.

In packet-based services such as frame relay, ATM, and optical Ethernet, tariffs are based on the access link speed and some form of committed rate: committed information rate (CIR) in frame relay, peak cell rate (PCR) in ATM, and committed access rate (CAR) in optical Ethernet. Adding IP telephony traffic results in the need to increase the committed rate of the link.

Flexible QoS Mapping At The WAN Edge

QoS is required to ensure that IP telephony traffic receives priority handling. Running IP telephony over leased lines leaves QoS and traffic management totally under the control of the enterprise. Support for QoS mapping when working into carrier packet services is another matter.

Optical Ethernet, generally available in metropolitan areas, provides native Ethernet connectivity with support for IEEE802.1p/Q Ethernet QoS. The high-speed, low-latency attributes of this service make it ideal for connectivity among metro sites. The CAR may need to be specified such that it supports the maximum number of simultaneous voice channels plus any data traffic.

Frame relay is a highly popular service available from 56Kbps to T3 rates, and even higher with ATM interworking. While frame relay QoS standards and products exist, service providers have not generally offered QoS-based services, though some publish statistical bounds on frame relay latency. Separate virtual circuits (VCs) with appropriate CIR should be established for IP telephony, to minimize interaction between voice and data traffic. ATM, on the other hand, is designed for multi-service transport, though it is relatively bandwidth-inefficient in supporting IP telephony; a voice stream coded in G.729 (8 Kbps coding) could take up over 80 Kbps across ATM. Voice (and optionally data) should be carried over appropriately sized VCs, either using constant bit rate [CBR] or real-time variable bit rate [rt-VBR] VCs.

QoS-enabled MPLS-based services offer a new option, which are starting to become generally available. These are well suited to IP telephony, and are generally premium priced over frame relay in hub and spoke configurations. In many cases, for reach or to work into the installed base, these will use leased line, frame relay, ATM, or Ethernet to reach your site, thus resulting in some complexities as already discussed.

With the increased availability of low-cost broadband access, running voice back to your site using IP VPNs over the Internet is very attractive for remote access and for connectivity to remote offices. There's no need to map QoS at the WAN edge since there isn't any QoS in the Internet. The good news is that for the vast majority of the time, this will work very well, though you may want to have a contingency to handle critical calls when the Internet performance goes downhill, by using cell phones or the PSTN.

QoS is required to ensure that IP telephony traffic receives priority handling.

Reducing Delay Through Packet Fragmentation

Data packet fragmentation is another tool that should be used to minimize voice delay and jitter over bandwidth-limited (<1 Mbps) connections. For frame relay connections, the provider can use the FRF.12 standard; once fragmented, recombination only takes place at the remote site. ATM (define - news - alert) natively provides fragmentation, since all packets are fragmented into 53-byte ATM cells. Over leased lines, Point-to-Point Protocol (PPP) fragmentation allows higher-priority VoIP packets to interrupt and transmit ahead of the remainder of larger, lower-priority data packets that have already been queued, this process being done on a link-by-link basis. The fragmentation size is adjusted to achieve a maximum delay of 20 ms over the different connection speeds. The recommended fragmentation size is N times 128 bytes for a link speed of N times 64 Kbps (e.g., 512 bytes at 256 Kbps).

The Right Balance Between Price And Performance

When you run voice over the WAN, you need to pay careful attention to bandwidth engineering, QoS handling at the WAN edge, fragmentation over slow speed links and of course, price/performance. While leased lines, Optical Ethernet and QoS-enabled MPLS VPNs offer the simplest QoS handling at the WAN edge, tariffs and service availability may move you to leveraging frame relay, ATM and even the Internet, with their implications on complexity and performance. IP telephony systems have addressed these realities by including proactive voice quality management capabilities which help IT better manage IP telephony quality of experience running over the WAN technology you have chosen. IT

When you run voice over the WAN, you need to pay careful attention to bandwidth engineering.

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

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By John Cimko

Should Municipal Governments Be Banned From Broadband?

A growing number of municipal governments want to provide broadband Internet services in their communities. Cable and phone companies want to stop them. Which result would better serve public policy goals?

The battle has been heating up since the Supreme Court ruled last year in Nixon v. Missouri Municipal League that states have authority to bar municipalities from offering telecommunications services. Several states have done just that, and cable and phone companies are lobbying other states to follow suit.

They argue that the private sector is already providing sufficient broadband services to businesses and consumers, that entry by municipalities would distort the market and suppress investment, and that municipal governments would have unfair competitive advantages in offering broadband services (such as operating on a tax-free basis).

Cable and phone companies shouldn't be faulted for lobbying against municipal broadband, since they have a duty to shareholders to guard against perceived threats to revenues. But it's helpful to look beyond the companies' arguments for a moment to ask what's really going on here.

The answer: market failure. Most municipalities providing broadband are acting for two reasons. First, broadband access is critically important. It will help attract new businesses to their communities, and these businesses will spur economic growth, create jobs, and enhance tax revenues. Broadband Internet access will also serve the educational, healthcare, and cultural needs of their communities.

Second, many local governments are convinced they can't rely on the marketplace to bring affordable broadband to their communities any time soon. In short, the market is failing.

This failure is not surprising. The deployment and pricing of broadband by private companies are necessarily driven by profit. Decisions about bringing broadband to particular communities are based on whether the return will justify the investment.

Governments in localities left behind by the cable and phone companies because of this revenue-based calculus can either gamble on the hope that the companies will eventually bring affordable broadband to their communities, or they can try to solve the problem themselves.

The history of municipal electric utilities provides precedent for the latter course. These utilities began operating in the 1880s and mushroomed during the Depression. They met a need and fixed a problem. The need was to deploy electric service as rapidly and extensively as possible. The problem was that private utilities could not be counted on to serve all areas, since they concentrated on lucrative urban markets. Municipal utilities were successful in filling the void. Today more than 35 million Americans receive their electricity from nearly 2,900 publicly owned utilities and cooperatives.

The fact that most municipalities involved with broadband are simply trying to solve the problem of market failure addresses many of the arguments raised by cable and telephone companies. The fact is that cable and phone companies aren't doing a good job in many smaller and rural communities — they're not providing any broadband services at all, or their rates are not affordable.

Municipal broadband services are not likely to cause market distortions or suppress investment because entry by municipalities in most cases is prompted by a failure by the private sector to serve their communities. The absence of any private investment is spurring entry by municipal governments. In fact, private companies have an incentive to keep older, existing facilities in place as long as possible before investing in new broadband infrastructure, especially in less lucrative markets.

Moreover, the Florida Municipal Electric Association has pointed out that "where municipalities invest in broadband, there are more private providers of broadband service." FMEA says that municipalities sell broadband to private communications firms, "and the result is a more competitive and symbiotic environment that benefits both consumers and the private sector."

Finally, it's important to remember that municipalities providing broadband services are not for-profit entities seeking to compete against cable and phone companies, but instead are trying to provide economic and social benefits for their communities. Even if municipalities did "compete" with cable or phone companies, it's not likely they would have any unfair advantage.

Local governments can't match the resources of the major cable and phone companies. Nor are they likely to offset this disadvantage through tax-free operations. Taking an example from a comparable industry, an American Public Power Association report (based on 2002 data) shows that net payments and contributions to local and state governments (such as "payments in lieu of taxes") made by 573 public power systems amounted to 5.8 percent of operating revenues. Meanwhile, investor-owned utilities paid 4.9 percent of operating revenues in taxes and fees to local and state governments.

President Bush has recognized the contribution that municipalities can make toward advancing the country's goals for universal broadband. In a June 2004 speech, the President endorsed efforts by municipalities to set up WiFi hot zones, calling these initiatives "a great opportunity." And the President's observation applies with even greater force to bringing broadband access to business and residential premises.

It's hard to escape the conclusion that, when viewed as a response to market failure, provision of broadband services by municipalities serves important public policy goals. **IT**

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

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By Hunter Newby

Claiming Your Place In The World Of VoIP Peering

Where we all are in the grand scheme of VoIP peering has much to do with where the service provider and customer are, geographically. It also has a lot to do with who the customer is. In and around the major cities where broadband access is generally highly available, VoIP services exist for both businesses and end users. Out in rural America there isn't much broadband because there isn't much of a business case to justify the build-out cost. They're lucky to have POTS in many of these locations and mobile signal in many parts is non-existent. There does exist a great divide in this country between the haves and the have-nots and it and will only grow wider if the RLECs don't do something about it. VoIP Peering may actually be the solution.

As important as it is to see who is deploying and using VoIP Peering it is equally important to see who is not there yet and why. Recently a special group of carriers known as the Intercarrier Compensation Forum (ICF) consisting of some of the big names like AT&T, SBC, Sprint, Global Crossing, and Level 3 and a couple of other vested interests created a document/proposal called the Intercarrier Compensation and Universal Service Reform Plan. http://www.tmcnet.com/145.1. The executive summary is a three-page document presented about a year ago and it essentially mapped out a way — their way — to reduce and ultimately eliminate the terminating fees they have to pay to the RLECs for calls bound for the RLEC networks. As the ICF states "Today's rules are broken beyond repair and must be replaced."

The reasons for this ICF plan include managing (removing) operating expenses and also increasing profitability, namely theirs. The profitability they are concerned with includes that of the flat rate service packages those carriers have come up with and are forced to offer under pressure from the VoBB and MSO unlimited packages that they compete with. Most of those plans are "Russian Roulette" if the caller has unlimited termination for a monthly fixed rate, but the provider gets billed per minute. If the caller calls a number in the RLEC network a lot the probability for profitability goes way down for the provider. The IXCs and RBOCs just don't want to pay the higher per minute rates and probably believe that since they are being marginalized so should the RLECs. In any case, whatever the motive, it is very clear that they want these rates to go to zero — and that's where they are headed eventually with, or without this plan, so the RLECs need their own plan.

As important as it is to see who is deploying and using VoIP Peering it is equally important to see who is not there yet and why.

It's not that they don't have a plan, either. It just seems that at this point they are trying to maintain the status quo. The Rural Alliance which is a group of 200 rural LECs banded together and produced a 181-page document in response to the ICF plan to outline how it is unfair and discriminatory. You can find it on the OPASTCO site http://www.opastco.org. The point was clear to me at least, they have been getting these terminating rates for a long time and that's what their businesses are based and rely upon. It can't just go away on someone else's schedule as they want it. There is also the small little issue of the Universal Service Fund (USF), which is there to help out the rural networks with the high operational and maintenance costs of providing service. Those costs are certainly much higher than the metro-based networks with their large addressable markets and foreseeable returns on investment. What happens to USF once VoIP takes over and traditional phone lines get cancelled? The money flow stops. Ouch. Now that's not something the RLEC can control, nor fight off. The fact is that POTS is going VoIP in broadband "information service" style faster than they can say rapid depreciation. So, how do they stop the bleeding?

There was one very interesting suggestion by the ICF in their plan about an "edge network" architecture that would justify and make possible their proposed decreases in the compensation charges. This concept is based on an idea that centralizing call hand-off points would be more efficient and could provide savings to the RLECs if implemented that would off-set the loss of the per minute compensation. In the summary of the ICF plan there was not much detail, but in the RLEC response they made it quite clear that they believed that this was not a good idea and would cost them more and be unfair.

The RLEC contention was that the edge interconnection points would be too far away for them to cost effectively reach. They also stated that ISP Peering (free traffic exchange) models — which look very similar to the "edge network" model — didn't work out and that ISPs have reverted to paying for transit. (In case the Rural Alliance hasn't noticed the London Internet Exchange, an IP Peering point, recently hit 76 Gbps — a world record http://www.linx.net). Basically the RLECs don't want to look at any payment method, income, or expense amount other than what they already have. They claim, and rightly so, that the law allows them what they have and that suggestions to change it should be within that law. That's nice, but trying to keep the past alive with laws that probably need changing only prolongs the inevitable and increases the divide between the haves (those in the flat rate model service area) and the have-nots (the ones who pay per minute). It's not that either side is totally right, or wrong. A balance must be reached.

Don't forget, VoIP drives broadband deployment because the business case is there for the consumer to save on the overall combined costs. The broadband access business case at layer 2 is very different than the "voice as an application" business case at layers 3 and up. Clearly this is not optimal for the traditional service providers and their traditional revenue levels, but it is the reality. Once the consumer figures it out, they'll demand it, or move. A physical move is more difficult than a service provider move, but it is possible.

A perfect example of this cycle is what happened in Allegany County, Maryland. The county was losing businesses and people because of one thing — lack of broadband. They were leaving the county physically. When the customers leave, so does the revenue. In this instance that would be tax revenue. Allegany County created AllCoNet and they looked at many providers and designs for a solution and finally found a phase 2 solution in Alvarion, using their pre-WiMax standard equipment. This took a rural, limited access situation and changed it in to a more urban style situation. You can see more about it at http://www.allconet.org.

Although this is not an RLEC and it is a municipality that has its own set of rules and regulations that are heating up the point is that they built a packet-based wide-area network and can now provide the services their customers need at affordable prices. The reality of free calls peered across private networks with direct connections is here now. Fighting it puts those that do at a disadvantage. This is not to say that the RLECs aren't planning on building their own VoIP WAN, creating an ENUM directory and peering all of their calls internally and across networks, but if they are they're not telling anyone. If they did, would they let the RBOCs send calls to them for free? I guess they would only if they could figure out how to make money from it in another way and of course if the RBOCs reciprocated.

The road to VoIP Peering has it peaks and valleys. Different operators are at different points on the road. In some places the weather is nice and in some it's rough. It seems like these folks are on a long, steep hill in the rain right about now, but they'll get there just like everyone else. IT

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

By Richard N. McLeod

It's Not Your Father's Dial Tone

Five Important Reasons To Migrate To IP Communications

Voice over Internet Protocol (VoIP) is reinventing the voice industry — merging voice, video, and data into integrated, interactive media and creating whole new applications, new ways of communicating, and new ways of doing business. The change is so fundamental and broad that, rather than VoIP, IP Communications (IPC) might be a more appropriate term.

We're seeing a sea change in the marketplace; it's no longer a question of if customers will deploy IPC, but when. And IPC isn't just for big business; large enterprises and small and medium-sized businesses are moving quickly to reap its benefits. Further, IPC offers value-add resellers and other solution providers exciting new opportunities to innovate, add value, and grow revenues and profits.

Just as the Internet and VPN (define - news - alert) technology let us take our corporate applications on the road, IPC lets us take our desktop communications, too. So many devices — IP phones in a remote office, PCs in a hotel room, WiFi equipped PDAs, and soon dual-mode WiFi cellular phones — are able to behave like a desk phone anywhere, providing full features and applications. When in the office, the IP phone is able to integrate with the desktop PC to provide LAN access and integration with collaborative tools, corporate directories and video.

Any device, anywhere, anytime: it just takes an Ethernet connection or WiFi (define - news - alert).

Top Five Reasons To Migrate To IPC

1. Reduced expenses and total cost of ownership.

Reducing cost is always an important business driver, and IPC offers compelling value. Having one IPC network instead of three networks for voice, video, and data saves companies deployment and management costs. Having applications resident in the network and shared, rather than dedicated to each location/PBX, saves money too. And for corporations with multiple locations or mobile employees, the savings are tremendous as every call is toll-free. It's not uncommon to see ROI paybacks in less than one year.

2. Technology obsolescence.

For decision makers considering technology investments for their businesses, increased bandwidth, new data security, and communications are surfacing as top priorities. The perfect time to "design in" IPC is when a company is redesigning its network.

The last major motivator for voice upgrades was Y2K. Flash forward six years, and those equipment leases are up and the assets fully depreciated; now is an ideal time for another technology refresh.

3. Redundancy and business resilience.

Redundant PBXs have traditionally been very expensive. But redundancy and failover capability are inherent in state-of-the-art IP architectures; in fact, IP enables levels of redundancy and resilience never possible with time-division multiplexing (TDM) technology. In any failure, voice communications and applications can be served transparently and immediately from servers anywhere on a global IP network. Security is achieved with IP components that address privacy through secure connectivity, protection through threat defense systems and control through trust and identity systems.

4. Virtual networking of business locations (including partners and even customers).

Historically, tying multiple locations into one dial plan network, call center and voicemail system — with all the productivity and improved customer satisfaction that brings — was only attainable by large enterprises and required expensive software investments with PBX (define - news - alert)vendors. With IPC, this capability is also inherent, so even small and mid-size businesses now have many new communications possibilities within easy reach.

5. Enhanced employee productivity and customer satisfaction from applications integration.

In today's global economy, organizations need a variety of real-time and asynchronous communications tools. IPC customers have documented the employee productivity benefits of collaboration, unified messaging and video applications.

For many businesses, productivity and customer satisfaction have been focused on the call center. Now, call center functions — such as customer relationship management (CRM)-based services — are possible across the enterprise. Desktop applications, back-office applications like enterprise resource planning (ERP) and CRM, and even industry-specific solutions can be integrated in ways never possible with TDM.

Any device, anywhere, anytime: it just takes an Ethernet connection or WiFi.

A Note On Migration

Misconceptions and confusion about migration — and about pure IP versus IP-enabled, or hybrid, solutions — abound.

First, it's important to be clear about what constitutes the embedded investment in a voice system: it's not the PBX but rather the voice assets, the LAN/WAN assets, the data applications and so on. Leveraging the existing network assets for a converged IPC solution is often the most cost-effective.

The foundation for these innovations is IP to the desktop. Conversely, like using rotary dial phones today to access integrated voice response (IVR) systems, with traditional digital phones connected to an IP-enabled PBX, certain benefits of IP just will not be possible.

A second fallacy is that migrating to IPC requires a massive "forklift" replacement. Instead, it's a matter of selecting an open-standard architecture and applications — like voicemail, unified messaging and rich-media conferencing — that are designed to support both IP and TDM connectivity. The faster a business moves to IP, the faster it gains the benefits and savings — and with the right infrastructure, migration can occur at whatever pace a business desires.

The faster a business moves to IP, the faster it gains the benefits and savings.

IPC is not only reinventing the voice industry, it is reinventing how business works and how people communicate. As with any fundamental transformation, there will be innovators, winners and losers. What about your business? Are you well positioned for the next wave of innovation? **IT**

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By Tim Lorello

The FCC VoIP E9-1-1 Mandate: Doom Or Zoom?

Over the last few months, we have been discussing the potential impact of E9-1-1 on the deployment of VoIP adoption. VoIP (define - news - alert) has the potential of revolutionizing the telecommunications industry, by providing a cost-effective method, using data communications, to replace various wireline and wireless communication methods in use today. However, it is important to know that today, wireline and wireless both provide the ability to automatically deliver a caller's location to an Emergency Services Responder when the user places a call to 9-1-1. This automatic location delivery bridges all languages, passes information even when the caller cannot speak, and enables the Emergency Telecommunicator (the person taking the call on behalf of Public Safety,) to focus on the nature of the emergency, rather than your current location, thereby saving many minutes in sending help and, ultimately, saving lives in the process. In many cases, VoIP does not provide location information automatically to Public Safety. So we have posed the question: will this Enhanced 9-1-1 (E9-1-1) capability prove to be the stumbling stone to VoIP? Lurking in the background of our discussion was the very real possibility of the FCC mandating some form of E9-1-1 service, as it did for wireless. On June 29th and becoming effective on July 29th, that possibility became a reality — the FCC, under the leadership of its new Chairman Kevin Martin, has required all VoIP Service Providers (VSPs) to provide E9-1-1 service to their customers regardless of how or where they are using the service. Our next step is to analyze this FCC Order and determine whether this bodes well or ill for the VoIP industry.

Competitors have become collaborators. And, in the end, the VoIP consumers will benefit.

What's In A Name?

The first part of this analysis must be to clearly determine to whom this Order applies. As one might expect when lawyers are involved, this particular point is under heated debate. According to the FCC Order, the mandate applies only to providers of "interconnected VoIP service." The FCC defined this to mean that the service must be connected to the Public Switched Telephone Network (PSTN) for the purposes of receiving calls from and placing calls to the PSTN. This is a very "common sense" approach to the problem and establishes the basic underlying principle of the Order: if the VoIP user can expect the service to work like a "typical" wireline or wireless service, then it must also provide E9-1-1. As a result, those services which only exchange data, such as instant messaging, or those voice services which only allow interconnection to other users of the same service, such as certain versions of Skype (news - alert), would not be treated as "interconnected VoIP service." This is the classic "if it walks like a duck and quacks like a duck, it's a duck" definition. The debate now rages around whether certain services that allow PSTN (define - news - alert) interconnection from a soft phone environment (e.g., making calls from your laptop or desktop computer utilizing a headset) are actually "interconnected" and affected by the Order. If I were to place bets, I would place them on the side of the safety of the consumer. If the soft phone directly or indirectly encourages the consumer to use it as a primary communication device, I believe that VSP will be impacted by the Order.

What And When?

The VSP has a number of very specific tasks to perform over the next few months.

First, by end of November, all customers of a VSP must provide Enhanced 9-1-1 as part of their service offering. The subscriber cannot opt-out of the service.

Second, by end of November, the VSP must transmit all 9-1-1 calls to the appropriate Public Safety Answering Point (PSAP) that services the caller's registered location. With the call, the VSP must automatically transmit the caller's registered location and a call back number (in cases where the call is disconnected, the PSAP can take action to reestablish communication). This is a change from the approach many VSPs use today which routes the 9-1-1 call to an administrative line at the PSAP — a 10-digit phone number that pretty much anyone may call. I like to call this the "side door" to the PSAP since it is not always staffed 24 hours a day and sometimes does not have the same capabilities that the "front door" to the Telecommunicator

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possesses, such as call transfer or dispatch access.

Third, by end of November, VSPs must obtain location information from their customers, and this must be done prior to initiation of service. For those VSPs that allow their users to access their VoIP service from multiple locations, the VSP must further provide a method by which the user may update their registered location at will and in a timely manner, including one method that requires use only of the Customer Premise Equipment necessary to access the VoIP service. In short, you must get the caller's location; the caller must be able to update it at will, and he/she must be able to use the phone to do it!

Fourth, by end of November, the VSP must file a letter of compliance with the FCC.

Fifth, by end of July, the VSP must notify the customer of their compliance with the FCC order and how the VSPs E911 offering differs from "traditional" E911 offers. In particular, the VSP must get from the end user a positive acknowledgement of reception of this information.

Doom Or Zoom?

What does all of this mean for the VSP? Is this the death knell for VoIP or will this propel it even faster into becoming a mainstream communication medium? The jury is still out on this verdict, but all indications are that this FCC Order has galvanized the industry into action. Though the terms of the Order are likely to be debated with the FCC, and though there continue to be technical, business, and regulatory impediments to providing E911 to VoIP users, this Order has forced all participants to determine how they should and will provide this life-saving service. Competitors have become collaborators. And, in the end, the VoIP consumers will benefit. It is likely that some VSPs will change the way in which they offer their VoIP services. Others may decide it is no longer feasible for them to offer service, and they will disappear from the VoIP landscape. But it has always been my contention that E911 is a required feature for any serious communications service. The average consumer, who does not anticipate the emergency, expects the service to work automatically when tragedy strikes. Thus, I believe that this Order will accelerate the process by which VoIP will take a rightful place as a mainstream communications medium. IT

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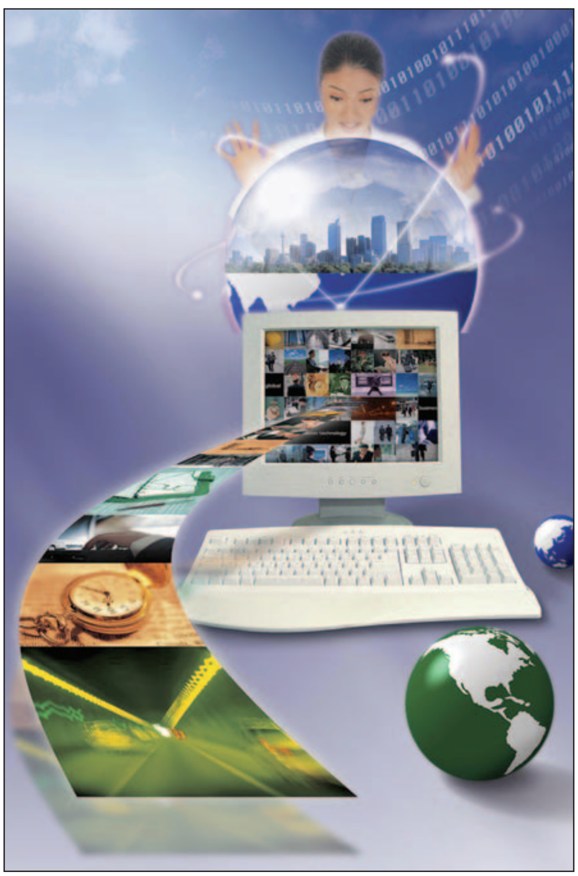
Beyond White Papers And PowerPoint Slides…

The Challenge Of Picking The Right IP Telephony Product

Rewind to just a few years ago. Most IT organizations were still only getting their feet wet relative to IP telephony products by reading, surfing the net, going to tradeshows and seminars — in essence trolling for information on this emerging technology that promised to carry voice interactions over the Internet. And once they gathered what they felt was sufficient data, it was time to begin looking at product options. Unfortunately, most IT chiefs found back then that the only solutions were partial ones. They also found that such incomplete solutions made it difficult to meet 100 percent of their organization's communications requirements.



Today, the "incomplete" IP telephony/voice over IP solutions problem isn't as much of an issue. Instead, the problem is how to sort through the substantial number of products that seem to do everything needed for a successful IP telephony deployment. Or more precisely, how to choose the best IP telephony product for your organization.
Our experience with thousands of deployments suggests there are three distinct factors that any IT chief and business enterprise should consider throughout the decision-making process.

Innovation: When linked with technology like IP telephony, the term "innovation" can easily be overused and watered down. But innovation is still a critical assessment, as in: How "forward thinking" is the product being evaluated? Most likely, you're evaluating an IP telephony product because your organization is using an outdated telephone system that needs to be replaced or upgraded. Therefore "innovation" questions should go like so:

- Does the new offering support the latest open communications standards, such as SIP?
- Does it include an application suite strong enough to extend the functionality my employees and customers need?
- Is it flexible enough for multi-site offices and mobile workers, and scalable enough to add more?
- Does it easily integrate with other products that are already deployed?
- Essentially, will the product you're gauging keep your business out in front of the technology curve?
- Making your evaluation with an "innovation factor" in mind will ensure that you won't be repeating the process again anytime soon.

Experience: Industry knowledge and business experience relates as much to the company you're buying from as it does to the products they sell. How long have they been in business? How many IP telephony installations do they have? Do they have the support infrastructure to complete a successful deployment? Ignoring the "experience factor" will raise your risk level, oftentimes beyond your comfort zone.

Value: Value really is a bottom-line evaluation. Based on the total cost of buying and owning the product, are you getting the best value for the money your business is spending? In other words, consider all system costs as a whole and not just the purchase price. In fact, you'll find the vendors who score high in the areas of innovation and experience are also typically leaders when it comes to the "value factor."

Industry knowledge and business experience relates as much to the company you're buying from as it does to the products they sell.

With these three factors guiding the overall selection process for any IP telephony solution, here are some more specific items to include in your evaluation. Note that some are business related, some are technology related, and that both aspects must be considered.

What is the reason for replacing the old system? It seems like such an obvious question (with many possible reasons), but there often are differing internal views to the answer — which can cause consternation and misjudgment in selecting a new IP telephony system. Arrive at the core reason, ask this question, and then be sure every person involved in the decision-making process is in agreement for the answer.

What are the size requirements for the new system? Can the system scale to meet current requirements … as well as requirements eight to 10 years from now? If so, is its scalability simple and straightforward?

Do remote employees need greater access to the communications system while out of the office? It can be extremely difficult, if not impossible, for some IP telephony communications systems to let remote employees and mobile users work as if they're in the office and serving as an integrated part of the organization. Or essentially, provide "Always In" capability to a growing mobile workforce. Look for a system that offers features such as remote access, Follow-me and Available Forward to make mobile employees available, and responsive, at all times.

Do you require increased coordination between teams, customer service and field services? Across departments? Many organizations have difficulty synchronizing between departments, workgroups and distributed offices to constantly know who's available, who can help with a pressing issue, what department can respond to customer inquiries and so on. Any IP telephony system worth its weight should include real-time presence management, across all locations and departments, for all employees whether in the office or away from it.

Do you need to manage telephony resources for multiple locations, and have your current systems been cumbersome and expensive to administer and maintain? If this is important to your organization, recognize that proprietary "multi-box" phone and communications systems generally have multiple administration interfaces. That is, no single point of administration. Conversely, a single box solution, or one that can be deployed and managed as a virtual single box solution, is essential to managing telephony resources across locations and reducing admin time and costs.

Do you have several high-volume interaction groups and/or informal call centers, such as Customer Service, Sales, Marketing, Technical Support, etc.? Many vendors find it both complex and expensive to offer screen pops and "big call center" functionality for small groups of 10 to 40 agents within an enterprise. Other vendors who can provide such capabilities simply can't offer them at an affordable cost. Products like the pre-integrated Enterprise Interaction Center (EIC) IP telephony and communications system from Vonexus provide more than ample call center functionality for small workgroups and informal call centers.

Consider all system costs as a whole and not just the purchase price.

Are you planning an initial blend between IP telephony and traditional TDM resources? An IP telephony system should provide options for voice over IP that offer maximum flexibility "down the road." Translated, your organization should be able to install the new system on traditional telephony interfaces and migrate to voice over IP later with no costly forklift upgrades. A strong contender for your business should also offer voice over IP for the telephony interface using SIP (define - news - alert) for all incoming calls and stations, or using SIP only for stations leaving traditional trunk interfaces connected to the PSTN (define - news - alert).

What other applications do you plan on deploying in the future? Screen pops, unified messaging, call recording, screen recording … even if your organization doesn't require and deploy applications like these at implementation, planning future needs for such applications as you research new IP telephony systems will help you zero in on the right system. In what should be a major red flag for application flexibility and adding new features later, many IP telephony systems simply are not architected to handle expanded applications without adding more hardware boxes. And adding more boxes only makes any IP telephony system more difficult and expensive to administer.

Some offerings, however, such as the EIC product from Vonexus, allow you to seamlessly integrate new applications and functionality whenever your employees and customers require, all to EIC's single platform and pre-integrated environment.

Consider The Right Factors

While this list isn't all-inclusive — it would take far more space to include every possible consideration — it does provide several key questions and considerations to include in selecting an IP telephony product. More critically, by combining prudent questions and in-depth research with general measurements against a vendor's innovation, experience and value, these factors will help ensure that the IP telephony solution you finally choose will meet your organization's needs for years to come.

Joseph A. Staples is Senior Vice President of Worldwide Marketing for Interactive Intelligence, Inc., and the company's wholly-owned subsidiary, Vonexus, Inc.

As a global developer of software for contact centers and the enterprise since 1994, Interactive Intelligence integrated out-of-the-box IP functionality into its lineup of business communications software solutions in 2002, and along with Vonexus is a leading industry innovator in the IP telephony, VoIP, and SIP movement. For more on their suite of IP telephony, contact center, and enterprise solutions, contact Interactive Intelligence at 317.872.3000 (http://www.inin.com) and Vonexus at 888-817-5904 (http://www.vonexus.com).

What To Ask

- **Why are you replacing the old system?**
- **What are your size requirements?**
- **Do remote employees need access?**
- **Do you require increased coordination between employees?**
- **Do you need to manage resources across multiple locations?**
- **Do you have multiple high-volume interaction groups?**
- **Are you planning to transition between IP telephony and TDM?**
- **What applications do you plan on deploying in the future?**

Internet Telephony Conference & EXPO: The Complete VoIP Event

Leading conference returns to Los Angeles; Thousands expected to invade LA Convention Center

The team at Internet Telephony Conference & EXPO has been hard at work this summer, putting together an event that promises to be the best-attended VoIP event in history. Based on the success of the last Internet Telephony show in Miami, and upon early registration numbers, this year's conference, which takes place October 24th–October 27th is expected to draw a crowd of nearly 8,000 attendees to the LA Convention Center.

Education

One thing is for sure, the crowds at the event will not be left lacking for things to do, see, or learn. The conference program is the most ambitious and thorough offering available anywhere. No matter what you're hoping to learn about, no matter which constituency you represent (Enterprise, Service Provider, Developer, Reseller…) you'll be faced with an educational experience that is second to none. Comprising four full days of conference sessions, Internet Telephony Conference & EXPO is set to deliver. Sessions will be available in a number of tracks, further enhanced with workshops and summits covering every vital topic:

Tracks for Service Providers:
- Service Provider Summit
- VoIP Peering Summit
- IMS Summit
- IPTV Summit
- WiFi Telephony Summit
- Open Source Summit
- SIP Workshop
- Conferencing & Collaboration Summit
- Mobility Summit

Tracks for Enterprises, Government, SMBs:
- Large Enterprise VoIP Deployment Workshop
- Enterprise/Government Solutions
- WiFi Telephony Summit
- IP Contact Center Summit
- Open Source Summit
- SIP Workshop
- Conferencing & Collaboration Summit
- Mobility Summit

Tracks for Developers:
- IP Telephony Development
- SIP Workshop
- Open Source Summit
- WiFi Telephony Summit

Resellers will have a complete day dedicated to teaching them how to become and remain profitable by reselling VoIP (define - news - alert) services and equipment. VoIP is the hottest segment in technology and Internet Telephony Conference & EXPO is the best venue to learn how to make the most of it now.

Keynotes

In addition to the unparalleled conference program, Internet Telephony Conference & EXPO is privileged to welcome the most impressive lineup of Keynote speakers ever assembled to address an audience hungry for insight into the VoIP space.

Former FCC Chairman Michael Powell and Carly Fiorina, former Chairman and CEO of HP, will lead a stellar lineup of industry giants who will be addressing the crowds at the event.

Many people forget that before her high-profile position at HP, Ms. Fiorina was a senior executive at Lucent, shaping that company's strategy and leading them through their spin-off from AT&T. Internet Telephony Conference & EXPO is the first opportunity to hear her perspective on the 'VoIP Revolution.'

Chairman Powell, as head of the FCC, presided during a period of tremendous growth for our industry. Attendees and press can expect to hear what he thinks now as he steps back and takes a big picture view of the future of VoIP.

Here's a complete list of keynoters scheduled to address the Internet Telephony audience:

- **Michael K. Powell**, Former Chairman of the Federal Communications Commission
- **Carly Fiorina**, Former Chairman & Hewlett-Packard
- **Brad Garlinghouse**, VP, Communications Products, Yahoo! Inc.
- **Niklas Zennström**, CEO & Founder, Skype
- **Raymond Pennotti**, Ph.D., Managing Vice President, Lucent Technologies
- **Philip L. Asmundson**, Vice Chairman & National Managing Partner Technology, Media & Telecom Deloitte & Touche LLP
- **Thomas Zimmermann**, President of Enterprise Systems, Siemens Communications
- **Tom Burger**, President, CEO and Director, NEC Unified Solutions, Inc.
- **Jerry Fleming**, President, Vonexus
- **Craig W. Rauchle**, President & COO, Inter-Tel
- **Larry Meyer**, Vice President, Sales & Marketing, Toshiba
- **Atul Bhatnagar**, Vice President & General Manager, Enterprise Data Networks, Nortel
- **Opher Kahane**, VP, Voice Technologies, Juniper Networks
- **Mike Donoghue**, VP of Sales, Americas Region, Aculab
- **Rick Moran**, Vice President, Product & Technology Marketing, Cisco Systems
- **Dr. Donald Brown**, Founder, Interactive Intelligence

Special Panels

Internet Telephony Conference & EXPO is also a great place to come and hear expert panels discuss and debate the hottest issues of the day. For example, the Future of IP Telephony panel has long been a staple of this event and is a great forum to hear from the industry leaders as they offer their perspective on where IP telephony is headed, and try to predict the impact this rapidly developing technology will have on the way businesses communicate in the future. Participating companies include MCI, Quintum, IBM, Intel, Ditech

Communications, and AltiGen.

Making its debut at the Los Angeles conference is a new session entitled the **Battle for the Enterprise/SMB**. Choosing the correct IP-PBX (define - news - alert)for your enterprise is no small challenge. With myriad solutions available, some from new players, other from legacy providers, whose solution is best for your particular installation? One size does not fit all. In some instances strong legacy support will be critical. In others, standards compliance will be crucial. In other situations, branch office support at a low cost or centralized management features will be important to consider. Will the new IP-PBX work well with your current infrastructure? Do you need to rip it out and rebuild? What about support, security, and service? This panel will strive to answer important questions from the audience and give you a unique perspective on what items to consider before selecting a solution that is right for your enterprise. Participants in this debate include NEC, Avaya, Toshiba, Nortel, and Inter-Tel.

Networking Opportunities

Internet Telephony Conference & EXPO Fall 2005 presents a terrific opportunity for you to meet and talk with other enterprises/government, service providers, developers, and resellers to share ideas, exchange business cards, and discuss the virtues of one solution over another. Several planned networking receptions are the ideal location for cementing existing relationships and generating profitable new ones with new partners or vendors. The receptions at Internet Telephony are forever distinguished by the partnerships spawned there.

The Exhibit Hall at this event will feature over 200 exhibiting companies spanning the world of VoIP. There is simply no better place to gather and talk with the companies leading this industry. And newly expanded hours make sure you get ample opportunity to get in and speak with all the companies you need to. This year, the Exhibit Hall will also feature free Learning Centers. Visit these special areas on the show floor and get more unbiased education about key VoIP topics and see some of the most powerful VoIP products and services in action.

Register Today!

Don't miss this opportunity to attend the largest and most complete VoIP event. Internet Telephony Conference & EXPO in Los Angeles is simply THE place to be this October. There will undoubtedly be lines at registration so get ahead of the game by registering online now. Not only will you be able to save money by registering online, but you'll avoid the long lines and get faster access to the conference sessions, exhibitors, and special attractions that make up the show. Log on to the show Web site at http://www.itexpo.com or call 203-852-6800 ext 146 for more information. IT

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Servicing The "Last Chance" Market And Flourishing

The Opportunity

ATSI Communications' (news - alert) chief executive officer, Art Smith, was unhappy with the way telecommunications providers were neglecting the needs of the Hispanic population and was determined to fix it. While larger carriers ignored the needs of this enormous community, ATSI emerged as a leading carrier focused on serving the vast market between Latin America and the United States by delivering International voice over Internet protocol (VoIP) services to previously disenfranchised customers. Today, ATSI's customers include a variety of carriers generating traffic within the United States and Mexico that require both Internet transport and long-distance termination. ATSI also operates an International private network for voice, data, and facsimile transport using IP.

"The Latin American telecommunications market is considered one of the largest in the world. Current estimates suggest it will reach $113 billion by 2005. Instead of seeing Latin America as a very large opportunity, and an economically strong market segment, many carriers have simply labeled it as a 'last chance market' because of the lack of business and consumer credit histories," said Smith. "The reality is, Hispanics spend over $21 billion on telephone services alone and that amount is expected to grow 12 percent a year. In ATSI's home state of Texas, there are 6.7 million Hispanic consumers representing 32 percent of the population. We felt it was time to create a telecom company that could fully understand and fulfill the needs of this unique market."

The Challenge

In targeting the Hispanic market, ATSI first revamped its network by deploying NexTone's Multiprotocol Session Controller (MSC), and then launched the company's retail brand, Telefamilia. Under the Telefamilia brand, local service and long distance calling plans offer both convenience and value directly to the Hispanic/Latino end-user who calls frequently between the United States and Latin America.

Enhanced services such as pre-paid conferencing and pre-paid Internet open doors to a larger base of end-users while the "*Enlacefamilia*" (the family link) toll-free service provides customers with unique billing option for calls originating in Mexico and terminating in the U.S. Beyond growing revenues, entry into the retail market increased ATSI's profits by eliminating the intermediate long-distance carrier and introducing several new high-margin retail products.

"Our customers are cost-conscious and price-sensitive so it is imperative that we offer unique service packages at affordable prices," said Smith. "Through our Telefamilia unit, ATSI currently permits customers to place phone calls to more than 225 countries from the United States and provides International toll-free access from 22 countries. Once our customers gain access to our network via either IP or toll-free access, we route the call to the appropriate destination, which is enabled by our NexTone system."

The Solution

ATSI deployed NexTone's MSC at the network edge of its VoIP (define - news - alert) network. NexTone's intelligent edge device operates alone or in conjunction with a core call control

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Enterprises, Service Providers, Manufacturers and Integrators depend on **NetAlly® VoIP** for network readiness assessment and troubleshooting.

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engine such as the NexTone Multiprotocol Signaling Switch (MSW). NexTone's technology has enabled ATSI to quickly adapt its VoIP network to reliably interface with VoIP solutions found in its partner's SIP or H.323 networks. Before ATSI deployed the NexTone MSC, the turn-up time for new interconnects was hindered because of differences in vendor implementations of VoIP signaling (SIP or H.323), T.38 fax support, and DTMF transport. NexTone's MSC quickly resolves these problems while maintaining the traffic in its native IP form. By keeping the traffic IP, ATSI can quickly extend their service reach and preserve quality while also lowering their equipment and operations costs.

NexTone's (news - alert) MSC also provides advanced media routing capabilities that include NAT traversal, topology hiding, route enforcement, and the regulation of bandwidth consumption to manage multimedia flows across carrier and enterprise boundaries. The product ensures consistent service quality for ATSI while protecting and securing its VoIP infrastructure.

In the three months following the deployment of the NexTone MSC, the number of VoIP minutes that traversed the ATSI network grew in excess of 270 percent. While ATSI expects more moderate growth rates in the future, this is directly related to the adoption of a 100 percent VoIP-based network. This level of speed for interconnection with customers and vendors would have been impossible utilizing traditional telecommunication network equipment.

Delivering Quality And Profits

NexTone's MSC lowers equipment costs, simplifies network operations, and provides faster service turn-up, enabling ATSI to use its VoIP media gateways more efficiently, thus realizing increased CAPEX and OPEX cost savings. In traditional VoIP networks, interoperability issues require carriers to dedicate one or more media gateway ports to each of its customers, commonly referred to as "back-to-back" media gateways; however, with NexTone's interconnect technology, ATSI is able to share port resources with multiple customers. ATSI is also now utilizing NexTone's iView Management System (iVMS), which provides a comprehensive set of management tools to engineer and streamline traffic routing tables.

Hispanics spend over $21 billion on telephone services alone and that amount is expected to grow 12 percent a year.

"The implementation of NexTone's MSC was a key move in allowing us to scale our business and improve our core service offering," said Smith. "We now have a cost-effective and reliable VoIP network that has helped us to secure new customers, while increasing the volume and revenue with our existing customers. NexTone's products allow us to provide exceptional customer service while monitoring performance on a real-time basis."

Summary Of ATSI Results

Utilizing NexTone's technology, ATSI has realized significant ROI, including the growth of its monthly volume of VoIP traffic by over 800 percent, resulting in a monthly revenue increase of nearly 350 percent since May 2004. On a quarterly comparison, ATSI's VoIP revenue increased by 150 percent for the first quarter of fiscal year 2005 versus the fourth quarter of fiscal year 2004. IT

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

I Guarantee the VoIP Education You Get at INTERNET TELEPHONY® Conference & EXPO is the Most In-Depth, Most Comprehensive, Most Valuable You Can Find.

Please join me in Los Angeles for **INTERNET TELEPHONY® Conference & EXPO Fall 2005.** Come spend invaluable time learning, networking with potential partners and vendors, and gathering the critical data you need to plan a successful VoIP strategy.

Some VoIP conferences are known for their parties and 'insider perspectives'. Others provide a forum for their largest sponsors to 'educate' you about what they think is the best solution for you.

At INTERNET TELEPHONY Conference & EXPO, we don't just say we provide the finest, commercial-free conference sessions — we guarantee it.

I encourage you to review the extensive conference program described in this brochure. Find the exact content geared to your business objectives, then join us in L.A. this Fall.

You will leave the event better equipped to tackle your VoIP project than when you arrived, or we'll give you a free pass to come back to a future conference absolutely free.

You can now deploy third-, fourth-, and fifth-generation products offering 100% reliability, scalability, affordability and rich feature sets traditional PBXs can't match.

This program is designed to ensure you leave the event with a strong plan of action for taking advantage of the amazing cost-savings if you are a VoIP user, and profit-generating opportunities if you are a service provider or reseller.

There is specific content geared to your needs as a service provider, large enterprise, SMB, government buyer, reseller or developer.

Sincerely,

Rich Tehrani, TMC President & Conference Chairman

The Complete VoIP Event

This Conference Program is so Thorough that it's Guaranteed*. (See full agenda on pages 6-7)

Including Workshop Summits in the Conference Program Covering Every Vital VoIP Topic

Tracks for Service Providers:

- Service Provider Summit **New!**
- VoIP Peering Summit
- IMS Summit **New!**
- IPTV Summit **New!**
- WiFi Telephony Summit
- Open Source Summit **New!**
- SIP Workshop
- Conferencing & Collaboration Summit **New!**
- Mobility Summit **New!**

Tracks for Enterprises, Government, SMBs:

- Large Enterprise VoIP Deployment Workshop
- Enterprise/Government Solutions
- WiFi Telephony Summit
- IP Contact Center Summit
- Open Source Summit **New!**
- SIP Workshop
- Conferencing & Collaboration Summit **New!**
- Mobility Summit **New!**

Tracks for Developers:

- IP Telephony Development
- SIP Workshop
- Open Source Summit **New!**
- WiFi Telephony Summit

Complete session listings — including detailed content descriptions — begin on page eight.

Contents

Don't Miss The Complete VoIP Event

Spectacular Keynotes: (See page 5)

This year's unprecedented lineup of keynotes includes top executives from equipment manufacturers and service providers, the former chairman of the FCC and one of the most influential executives in technology for the last 20 years.

Each has extensive experience in telecom - dating back to long before the VoIP revolution. Learn from them why this opportunity is far bigger than any in the telecom industry's century-long history.

- **Michael Powell, Former FCC Chairman**
- **Carly Fiorina, Former HP Chairman & CEO**

Plus, top executives from:

Yahoo!	**Cisco Systems**
Skype	**Deloitte & Touche**
Siemens	**Juniper Networks**
Toshiba	**Lucent Technologies**
NEC	**Nortel**
Vonexus	**Aculab**
Inter-Tel	

More Important Topics Covered Within Each Conference Track

Within each major track heading, there are sessions exploring the most important micro-topics for enterprises, service providers, developers, government and resellers. These topics include:

- VoIP Security
- Security/Surveillance over IP New!
- e911/Regulation & Taxation
- VoIP Traffic Management New!
- Consumer VoIP Marketing New!
- Peer-to-Peer Telephony
- Session Border Controllers
- Number Porting/ENUM New!
- Triple Play
- Dual Mode New!

TMC University's IP PBX Certification Courses (See page 22)

The only independent certification program of its kind validating your competency in IP PBX selection, deployment, implementation & management.

"The Future of IP Telephony" and "Battle for the Enterprise/SMB" Panel Discussions

(See page 24)

Always among the most popular events at the conference, these free general sessions give you the chance hear many views and opinions about the best choices for you and your organization.

Future of IP Telephony Participants include:

- IBM
- Intel
- Quintum
- AltiGen Communications
- Ditech Communications
- MCI

Battle for the Enterprise/SMB Participants include:

- Avaya
- NEC
- Toshiba
- Inter-Tel
- Nortel

NEW! Free Educational Seminars – Reseller Solutions Day & Service Provider Solutions Day (See page 23)

These full-day seminars help resellers and service providers sort through the many choices of equipment and service they can choose — whether finding the right solution for a client or for their own network.

NEW! Free Learning Centers on the Exhibit Floor (See page 25)

- **Triple Play**
- **Open Source**
- **SIP Interoperability**
- **WiFi Telephony**

Your visit to the centers supplements what you learn in the conference sessions with demos and explanations about specific VoIP technologies. Participating vendors are prohibited from mentioning their products. Rather, here's yet another opportunity for you to leave this conference with the most complete education possible, preparing you to make a smart VoIP buying decision.

*Our Guarantee:

If you do not feel the sessions you attend made you better prepared to tackle your VoIP project than you were when you arrived, stop by the registration counter at the show and we'll issue you a free pass for any future INTERNET TELEPHONY conference. (No requests honored after the conference ends.)

Benefits of Attending

No other VoIP event offers the combination of a first class conference education, endless networking opportunities with collegaues, vendors, resellers and developers, and visionary keynotes from such wide-ranging viewpoints.

1. Commercial-Free Conference Sessions

Presenters are forbidden from delivering company pitches in sessions. Any who do, are not invited back to future events. You get a purely unbiased VoIP education.

2. Most Knowledgeable Speakers

Each topic and presenter is hand-selected by Greg Galitzine, editorial director of INTERNET TELEPHONY magazine since 1998, and Rich Tehrani, editor-in-chief for TMC, from literally hundreds of submissions. Only the most relevant sessions submitted by seasoned speakers make it on to the program at INTERNET TELEPHONY Conference & EXPO.

3. Invaluable Networking Time

INTERNET TELEPHONY Conference & EXPO Fall 2005 is the perfect opportunity for you to meet and talk with other enterprises/government, service providers, developers and resellers to share ideas, exchange business cards, and discuss the virtues of one solution over another.

4. Over 200 Exhibiting Companies

In between sessions, meet vendors and partners you need to successfully deploy VoIP solutions. The agenda leaves ample time to stop by each booth to discuss how each exhibitors' offerings can help you.

4. Free Learning Centers on the Exhibit Floor

Visit these special areas on the show floor and get more unbiased education about key VoIP topics and see some of the most powerful VoIP products and services in action.

5. Top-Level Keynotes

This year's unprecedented lineup of keynotes includes top executives from equipment manufacturers and service providers, the former chairman of the FCC and one of the most influential executives in technology for the last 20 years. Each has extensive experience in telecom. Learn from them why the VoIP opportunity is far bigger than any in the telecom industry's history.

6. Special Panel Sessions

In addition to the non-commercial conference sessions, you can attend 'The Future of IP Telephony' and 'Battle for the SMB', two panel discussions where vendors and service providers will share their vision of how VoIP can benefit you.

7. Your Conference Fee is Guaranteed

If you do not feel the sessions you attend made you better prepared to tackle your VoIP project than when you arrived, stop by the registration counter at the show and receive a free pass for any future INTERNET TELEPHONY conference. (No requests will be honored after the conference ends.)

8. Convenient, Easily Accessible Location

The conference center is located in downtown Los Angeles, easily accessible by major highways and only 20 miles from LAX.

Who Should Attend?

Corporate Management, CTOs – Ultimately, the vendor you choose for your VoIP deployment will become more of a partner than a supplier. Whether you are an enterprise deploying a solution, or a service provider preparing your VoIP network, INTERNET TELEPHONY Conference & EXPO provides the perfect venue for forging these profitable relationships.

Resellers – You get a full day of free sessions teaching you how to make money selling VoIP service and equipment, and the opportunity to meet with literally hundreds of companies who could become your next partner. Need we say more?

IT/Telecom Management – It's up to you to make sure your VoIP deployment is smooth with minimal disruption. It's also your responsibility to ensure your new system meets all organizational objectives. The days you spend in the conference sessions and the exhibit hall at INTERNET TELEPHONY Conference & EXPO will supply the answers you need to recommend the perfect system for your situation.

Developers – No other conference offers four full days of conferences teaching you how to take advantage of today's most powerful development tools. In between sessions, form partnerships and relationships meet with manufacturers and vendors.

Come Learn as an Unprecedented Line-up Headed by Michael Powell and Carly Fiorina Share Their Vision of Why All Businesses and Government Agencies, Large and Small, Will Soon Be Using VoIP.

Michael K. Powell
Former Chairman of the
Federal Communications Commission

Carly Fiorina
Former Chairman & CEO of
Hewlett-Packard

Niklas Zennström
CEO & Founder
Skype

Brad Garlinghouse
VP, Communications Products
Yahoo! Inc.

Raymond Pennotti, Ph.D.
Managing Vice President,
Lucent Technologies

Tom Burger
President, CEO and Director,
NEC Unified Solutions, Inc.

Jerry Fleming
President,
Vonexus

Craig W. Rauchle
President & COO
Inter-Tel

Thomas Zimmermann
President of Enterprise Systems
Siemens Communications

Larry Meyer
Vice President, Sales & Marketing
Toshiba

Atul Bhatnagar
Vice President & General Manager,
Enterprise Data Networks, Nortel

Mike Donoghue
VP of Sales, Americas Region
Aculab

Opher Kahane
VP, Voice Technologies
Juniper Networks

Philip L. Asmundson
Vice Chairman & National Managing
Partner Technology, Media & Telecom
Deloitte & Touche LLP

Rick Moran
Vice President, Product & Technology
Marketing , Cisco Systems

Dr. Donald Brown
Founder,
Interactive Intelligence

TMC INTERNET TELEPHONY CONFERENCE & EXPO

Conference At A Glance

Day 1 – Monday, October 24, 2005

	TMC University's IP PBX Certification Program First Degree	Large Enterprise VoIP Deployment Workshop	Open Source Summit	Service Provider Summit / Number Porting / ENUM	Conferencing & Collaboration Over IP Summit	IMS Summit
11:00 am - 5:00 pm	**Registration Open**					
12:00 - 12:45 pm	Reaping the Benefits of the IP PBX TMCU-1	Integrating Voice & Data with Business Applications LE-1	The Role of Standards in Open Source OS-1	Creating the Trusted Authenticated Network SP-1	Introduction to Collaboration & Conferencing CC-1	Introduction to IMS IMS-1
1:00 - 1:45 pm	Cost Justifying the Upgrade TMCU-2	QoS & Infrastructure Issues in Deploying Enterprise VoIP LE-2		Number Portability for a Growing VoIP World SP-2	The Future of Unified Conferencing CC-2	IMS: Developing Network Infrastructure IMS-2
2:00 - 2:45 pm	Selecting the Right IP PBX Solution TMCU-3	Delivering Secure & Reliable Enterprise VoIP LE-3	Open Source Round Table OS-2	VoIP Peering Through ENUM Registries SP-3	Selecting Enterprise IP Conferencing Solutions CC-3	Fixed Mobile Convergence – Finding the Killer Applications IMS-3
3:00 - 3:45 pm	Effective Deployment and Migration Strategies of IP PBX TMCU-4	Calculating the True Business Value of VoIP LE-4		ENUM: The Theory is Done, Now It's Time for the Practice SP-4	Adding Virtual Communications to the Mix CC-4	Deploying Services Using IMS IMS-4
4:00 - 4:45 pm	Living With Your New IP PBX – A Case Study TMCU-5	Utilizing Presence to Keep Remote and Branch Offices Connected LE-5	Linux as an Embedded OS and Development Environment OS-3	SIP & ENUM: What's The Story? SP-5	Making Gov't Agencies Effective & Efficient with Real-Time, IP Collaboration CC-5	A Look Ahead: The Future of IMS IMS-5
4:45 - 6:00 pm	**Keynote Session Featuring Yahoo! and Lucent Technologies**					

Day 2 – Tuesday, October 25, 2005

	TMC University's IP PBX Certification Program Advanced Degree	Enterprise / Government Solutions	IP Telephony Development	Service Provider Summit	IPTV Summit
7:00 am - 7:00 pm	**Registration Open**				
7:30 - 8:30 am	**Conference Breakfast**				
8:30 - 9:30 am	**Grand Opening Session Featuring NEC and Vonexus**				
9:30 am	**Special Keynote Presentation by Carly Fiorina, Former Chairman & CEO, Hewlett-Packard**				
10:15 - 11:00 am	Preparing Your Network for an IP PBX TMCU-6	VoIP: Where We're Going, Where We've Been E-1	VoIP Without DSPs: Challenges and Opportunities D-1	On Top Of Their Game?: A Review of Leading VoIP Service Providers SP-6	Introduction to IPTV IPT-1
11:15 - 12:00 pm	Right Sizing Your IP PBX TMCU-7	VoIP From The Trenches: Real Life Successes E-2	Advances in Development Platforms D-2	Triple Play: The Implications of Convergence SP-7	Development Tools For IPTV Deployment IPT-2
12:00 pm	**Conference Luncheon – Featuring Keynote via Videoconference by Niklas Zennström, CEO, Skype**				
1:00 - 1:45 pm	Integrating Your IP PBX With an ITSP TMCU-8	Presence & Unified Communications E-3	DSP Processing for VoIP Applications D-3	Voice Quality Challenges Facing Providers SP-8	Building Networks for Delivering IPTV IPT-3
2:00 - 2:45 pm	Staging, Implementing, and Cutting Over Your IP PBX TMCU-9	VoIP for SMBs E-4	Delivering Wideband Speech Over IP D-4	VoIP Opening Night: Are Service Providers Truly Prepared? SP-9	The Current State of IPTV IPT-4
3:00 - 3:45 pm	An Advanced Case Study TMCU-10	The Future of Enterprise Applications E-5	VoIP Testing & Security Issues D-5	VoIP Performance Management SP-10	An IPTV Case Study IPT-5
3:45 - 4:30 pm	**The Future of IP Telephony Panel Discussion**				
4:30 - 6:00 pm	**Keynote Session Featuring Inter-Tel and Siemens**				
6:00 - 8:00 pm	**Networking Reception in Exhibit Hall**				

Conference At A Glance

INTERNET TELEPHONY CONFERENCE & EXPO

Day 3 – Wednesday, October 26, 2005

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<tr><td></td><td>Wifi Telephony Summit</td><td>Enterprise/ Government Solutions</td><td>SIP Workshop</td><td>Service Provider Summit</td><td>VoIP Peering Summit</td><td>IP Contact Center Summit</td></tr>
<tr><td>8:30 - 9:15 am</td><td>Introduction to WiFi Telephony W-1</td><td>Designing/Developing Enterprise VoIP Security Solutions E-6</td><td>An Introduction to SIP SIP-1</td><td>Consumer VoIP Marketing Summit Part I SP-11</td><td>The State of VoIP Peering VP-1</td><td>Transition Your Contact Center to IP IPC-1</td></tr>
<tr><td>9:15 - 10:15 am</td><td colspan="6">Keynote Session Featuring Toshiba and Nortel</td></tr>
<tr><td>10:15 - 11:00 am</td><td colspan="6">Special Keynote Presentation By Michael K. Powell, former FCC Chairman</td></tr>
<tr><td>11:00 am - 5:00 pm</td><td colspan="6">Exhibit Hall Open</td></tr>
<tr><td>11:45 am</td><td colspan="6">Conference Luncheon – Featuring Presentation by Dr. Donald Brown, Founder, Interactive Intelligence</td></tr>
<tr><td>12:30 - 1:15 pm</td><td rowspan="2">Technical Challenges to WiFi Telephony Deployment W-2</td><td rowspan="2">Security Challenges in IP Telephony E-7</td><td>SIP Interoperability Issues SIP-2</td><td>Consumer VoIP Marketing Summit Part II SP-12</td><td rowspan="2">The Future of Enterprise Network Peering VP-2</td><td rowspan="2">IP Contact Center Shootout IPC-2</td></tr>
<tr><td>1:30 - 2:15 pm</td><td>Overcoming SIP Implementation Challenges SIP-3</td><td>VoIP Regulatory Update: Will You Be Ready? SP-13</td></tr>
<tr><td>2:30 - 3:15 pm</td><td>The Challenges of Citywide WiFi Scalability W-3</td><td rowspan="2">The Challenge of E911 Regulation E-8</td><td>SIP's Role in Open Source SIP-4</td><td>Security in the IP Telephony Network SP-14</td><td>Developing a VoIP Network Exchange Infrastructure VP-3</td><td>VoIP and Offshoring -- Pros & Cons IPC-3</td></tr>
<tr><td>3:30 - 4:15 pm</td><td>Call of the Campus Warrior: A Healthcare Case Study W-4</td><td>Service Provider Issues Relating to SIP SIP-5</td><td>Security/Surveillance Over IP SP-15</td><td>Clearing & Peering Business Models and Technologies VP-4</td><td>It's Up and It's Good! Dallas Cowboys IP Contact Center Case Study IPC-4</td></tr>
<tr><td>4:15 - 6:15 pm</td><td colspan="6">Battle for the SMB Panel Discussion; Keynote Session Featuring Aculab and Juniper Networks</td></tr>
<tr><td>6:15 pm</td><td colspan="6">Networking Reception</td></tr>
<tr><td>7:30 pm</td><td colspan="6">VoIP Service Provider Awards Dinner</td></tr>
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Day 4 – Thursday, October 27, 2005

<table>
<tr><td>7:30 am - 2:00 pm</td><td colspan="5">Registration Open</td></tr>
<tr><td>7:30 - 8:30 am</td><td colspan="5">Conference Breakfast</td></tr>
<tr><td></td><td>WiFi Telephony Summit</td><td>Enterprise / Government Solutions Traffic Management</td><td>SIP Workshop</td><td>Service Provider Summit</td><td>Mobility Summit</td></tr>
<tr><td>8:30 - 9:15 am</td><td>Keys to Successful WiFi Telephony Implementation W-5</td><td>Moving the Market: The True Target for Enterprise Communications E-9</td><td>State of the SIP Union: Best Practices & More SIP-6</td><td>VoIP & Next-Generation OSS SP-16</td><td>VoIP Mobility Applications M-1</td></tr>
<tr><td>9:15 - 11:00 am</td><td colspan="5">Keynote Session Featuring Deloitte & Touche LLP and Cisco Systems</td></tr>
<tr><td>11:00 am - 3:00 pm</td><td colspan="5">Exhibit Hall Open</td></tr>
<tr><td>11:30 am</td><td colspan="5">Conference Luncheon</td></tr>
<tr><td>12:15 - 1:00 pm</td><td>Deploying WiFi Telephony in the Enterprise W-6</td><td>Peer To Peer IP Telephony E-10</td><td>Securing SIP-Based Communications SIP-7</td><td rowspan="2">Migrating to VoIP: A Service Provider Round Table SP-17</td><td>VoIP Mobility Reality Check M-2</td></tr>
<tr><td>1:15 - 2:00 pm</td><td>Developing WiFi Telephony Endpoints W-7</td><td>Enterprise Traffic Management Challenges E-11</td><td>SIP? NAT? Not! Traversing the Firewall SIP-8</td><td rowspan="2">Dual Mode: A Look at the Current State of the Market M-3</td></tr>
<tr><td>2:15 - 3:00 pm</td><td>The Road Ahead: The Future of WiFi Telephony W-8</td><td>Managing Your Network for High-Quality Voice E-12</td><td>SIP in the IP Contact Center SIP-9</td><td>Session Border Control for Hosted VoIP Services SP-18</td></tr>
</table>

Large Enterprise VoIP Deployment Workshop

Monday October 24 - 12:00 - 12:45 pm

Integrating Voice & Data With Business Apps

Organizations are converging their data and voice systems into a single network, providing them with cost savings and increased operational efficiencies in their offices and branch sites. Today, these enterprises are taking convergence to the next level by integrating mission-critical business applications such as CRM, inventory management, and the like into their IP Communications network. The presenters will explain how this integration makes it possible to create customized IP communications solutions that can increase customer service and satisfaction, improve workforce productivity, and further increase operating efficiencies.

Monday October 24 - 1:00 - 1:45 pm

QoS & Infrastructure issues in Deploying Enterprise VoIP

The roll-out of VoIP presents a quandary. The reality of high expectations for voice quality combined with the challenges of running voice on IP networks opens up the risk of serious business failure for the network operations team. How can an enterprise CIO ensure the delivery of QoS that VoIP services demand? This session will look at the specific QoS requirements of VoIP, where those issues arise in the network, and how to measure and deliver high levels of QoS. The presenters will also look at the Ethernet edge and Ethernet core to see how other network elements impact QoS.

Monday October 24 - 2:00 - 2:45 pm

Delivering Secure & Reliable Enterprise VoIP

With most major telecommunications carriers currently in the process of deploying VoIP services for mass deployment, it's clear that IP telephony is finally headed for prime time. However, the promise of mass VoIP consumption also increases the risk for widespread security violations, spawning a new sense of urgency to develop secure VoIP solutions now before hackers wreak havoc on corporate voice networks. This session will examine the special demands of enterprise VoIP, with a special emphasis on delivering secure and reliable IP voice over your enterprise network.

Monday October 24 - 3:00 - 3:45 pm

Calculating the True Business Value of IP

Beyond the "traditional" benefits of VoIP such as reduced infrastructure costs, toll bypass, etc…, VoIP offers more sophisticated value through improved productivity, enhanced customer experience, converged applications and the ability to obtain new sources of revenue growth. To define the true value of an IP solution requires examining more than the short-term benefits of upgrading the architecture; it is also about considering its role as a fundamental business asset in a strategic plan designed to generate financial growth and competitive advantage. This presentation will focus on the business goals and benefits of deploying an IP-based enterprise communications solution.

Monday October 24 - 4:00 - 4:45 pm

Utilizing Presence to Keep Remote and Branch Offices Connected

Presence, once considered merely an underlying technology to instant messaging (IM), is becoming its own killer application, and is posed to go well beyond instant messaging and deep into the enterprise. The future of presence management is the convergence of VoIP, data, video, IM and location tracking on top of wired and wireless networks to display, in an application, the whereabouts and status of a user's colleague. This session will explore the role of presence in keeping remote workers and branch offices connected while meeting the updated performance expectations of today's business environments.

ENTERPRISE/GOVERNMENT SOLUTIONS

Tuesday, October 25 - 10:15 - 11:00 am

VoIP: Where We're Going, Where We've Been

Join this session if you a new to the world of Voice over IP or if you feel you need a refresher course. The speakers will consider the essential technical differences between traditional telephony and IP telephony, the challenges that these differences present and the way that they are overcome in successful real-life deployments. The rules have significantly changed and the way we think about telephony solutions must also change. This is truly a can't-miss session for attendees looking for a thorough overview.

Tuesday, October 25 - 11:15 am - 12:00 pm

VoIP From the Trenches: Real Life Successes

This panel discussion will feature several telecom dealers talking about their experiences deploying IP in a variety of industries. Among the topics they will address:

- Understanding when pure IP telephony is best choice
- Migrating TDM users to IP telephony
- Identifying and overcoming technology hurdles
- Finding creative solutions
- Taking VoIP mobile (wireless phones, softphones, etc.)
- Calculating ROI — cost savings and productivity increases.

Tuesday, October 25 - 1:00 - 1:45 pm

Presence & Unified Communications

Presence has the potential to be the unifying thread between different modes of communication including enterprise telephony, IM, video, and mobile telephony. Presence is the essential element in the migration from device centric to user centric systems, as it integrates user's availability across all their devices and applications. But Presence also has a dark side which may impede its broad acceptance, namely concerns about privacy, time management, enterprise policy and security. Each of these aspects is explored during this session, and solutions in development, standards bodies, and emerging products are discussed.

Tuesday, October 25 - 2:00 - 2:45 pm

VoIP for SMBs

VoIP is not just for large enterprises. In fact, VoIP plays a significant role in impacting performance for small- and medium-size businesses. This session will address the drivers that affect the SMB market — such as the growing reliance on outsourcing and third-party vendors, the challenge to utilize remote knowledge workers, and the increasing pressure to control costs. In addition, this session will cover VoIP-powered applications, like presence management, collaboration, messaging, and mobility, and examine how SME customers can effectively leverage these tools to improve business performance.

Tuesday, October 25 - 3:00 - 3:45 pm

The Future of Enterprise Applications

As more and more customers become accustomed to using VoIP, they'll begin to expect value-added enhancements and additional benefits, aside from cost savings alone. What types of VoIP-enabled services and applications will be developed to satisfy this demand? What will be the "next big thing" in the VoIP market? How will savvy vendors, service providers, and application developers take advantage of this capability? How will they use VoIP technology to roll out previously impossible or unheard-of applications and maximize the benefits for end users? Come to this session to catch a glimpse of the future of enterprise VoIP.

Wednesday, October 26 - 8:30 - 9:15 am

Designing/Developing Enterprise VoIP Security Solutions

While VoIP has many positive benefits to corporations, the potential security risks involved should not be ignored and must be effectively addressed before a loss in money or private data occurs. New products and services must be created to eliminate these security threats. The speakers will address the growing need for the communications industry to address and develop VoIP security solutions now and for the future. The session will cover how implementing security at the "core" of the VoIP system, the platform level, and why ensuring that encryption is synced from the device and network is critical.

Wednesday, October 26 - 12:30 - 2:15 pm DOUBLE SESSION

Security Challenges in IP Telephony

This panel discussion, moderated by the chairman of the recently formed VoIP Security Alliance will strive to identify and reduce VoIP security risks through a number of ways including educating and providing tools for the IT manager. Security is a critical concern for enterprise looking to deploy VoIP. As such this special session stands a must-attend for professionals charged with managing their company's VoIP deployment.

Wednesday, October 26 - 2:30 - 4:15 pm DOUBLE SESSION

The Challenge of E911 Regulation

Nationwide, an estimated 200 million calls are made to 911 each year. In May, the FCC decided that VoIP providers interconnecting with the PSTN must deliver E911 calls — with location and callback information — to local emergency operators. The decision raises a host of issues: What information, specifically, must be delivered with the call? Is network support in place, and at what cost? Who will enforce the E911 requirement? And how will the requirement apply to enterprise VoIP, where callback and location identification pose special challenges? Finally, will the VoIP legislation pending before Congress supplement or preempt the FCC's decision? Join public safety administrators and legal and policy experts to hear the latest E911 news from Washington and around the country.

ENTERPRISE/GOVERNMENT SOLUTIONS

Thursday, October 27 - 8:30 - 9:15 am

Moving the Market: The True Target for Enterprise Communications

This presentation will examine the market from the perspective on how customers want to purchase and consume voice services and map enterprise communications products into those needs. We will also examine the market shift that has been waiting to happen as an application level conversion instead of the more common view of the shift as a more basic transport level conversion. Ultimately, this presentation will identify the true target for VoIP and other enterprise communications tools need to hit to move the market into the mainstream.

Thursday, October 27 - 12:15 - 1:00 pm

Peer to Peer IP Telephony

Peer-to-Peer VoIP is a potentially disruptive force that is taking the IP telephony market by storm. Or perhaps, "incrementally by storm." Little by little, bit by bit, this approach to VoIP is taking hold. Proponents tell us that P2P VoIP addresses issues such as scalability, security, redundancy that are required in enterprise and service provider implementations, while supporting basic as well as innovative features. Come to this session to see what all the fuss is about.

Thursday, October 27 - 1:15 - 2:00 pm

Enterprise Traffic Management Challenges

IT professionals have been examining their networks in critical detail to uncover hidden or latent problems since the dawn of networking, with the ultimate goal of improving delivery of mission-critical business applications. This discipline of troubleshooting has grown in importance in the age of VoIP with multimedia applications, global networks, and new protocols. This session will discuss how gaining greater visibility into the network is leading organizations to better management of their converged networks and optimized delivery of all business and voice services. Other topics will include setting QoS policies, data collection/troubleshooting and how applications impact each other when traversing the same network segments.

Thursday, October 27 - 2:15 - 3:00 pm

Managing Your Network for High-Quality Voice

Throwing voice packets onto a LAN and hoping for the best is easy. Deploying high-quality VoIP is a bit more challenging. This session will address the need for service quality management and how unleashing the network infrastructure, enforcing a centralized policy for end-to-end QoS, and controlling network resources on a proactive basis can deliver high levels of quality for IP communications. The session will also address related issues of bandwidth shaping and equalizing technology that are designed to enable IT professionals to manage their enterprise voice and data network.

OPEN SOURCE SUMMIT

Monday October 24 - 12:00 - 1:45 pm DOUBLE SESSION

The Role of Standards In Open Source

Open Source — software where the source code is available for anyone to improve or modify — has the potential to change the telecom landscape forever. Linux has made huge headway into this market, this OS and other Open Source solutions have caused quite a stir. Developers need to know what is available in the Open Source marketplace and how best to leverage the freedom inherent in a distributed community of programmers to amend and adapt code for their VoIP development needs. This session will address the role of emerging VoIP standards and what they mean to the development of Open Source.

Monday October 24 - 2:00 - 3:45 pm DOUBLE SESSION

Open Source Round Table

This group discussion will zero in on the current state of Open Source telephony; what it means, where it stands, and where this segment of the market is headed. Thought leaders from a number of influential Open Source factions will share their insights with the audience. Attendees can expect a frank conversation, and will leave this session with a thorough understanding of Open Source and its role in the future of IP Telephony.

Monday October 24 - 4:00 - 4:45 pm

Linux as an Embedded Operating System and Development Environment

As with any technology element, subsequent product generations typically yield performance, size, integration, or cost advantages. This is often driven by the silicon devices used to define the product. Key to leveraging the advances in silicon has been the equal advances in embedded software. A leading element is the ongoing growth and maturity of Linux as a fully functional and flexible operating system environment, as well as a configurable environment for embedded systems. The performance and integrated benefits of contemporary silicon devices coupled with the functionality and cost benefits of Linux are providing embedded developers with options never before available. This session will discuss the use of Linux as an embedded operating system and development environment and how it can be applied towards the development of embedded systems.

IP TELEPHONY DEVELOPMENT

Tuesday, October 25 - 10:15 - 11:00 am

VoIP Without DSPs: Challenges & Opportunities

Does moving the media processing function from DSP to the host fundamentally change the architecture? Yes! Host Media Processing (HMP) offers new opportunities for voice-enabled applications to be architected — to create a very broad range of scalability; to partition functionality in very different ways; and to create highly reliable services. This session will also discuss new ways to process media that take greater advantage of the host environment.

Tuesday, October 25 - 11:15 am - 12:00 pm

Advances in Development Platforms

VoIP development platforms are a constantly evolving lot. From PCI and CompactPCI to today's most advanced ATCA and microTCA-based platforms, there are so many changes that it might be tough to stay abreast of all that is happening in this fast-moving space. Attend this session, and you'll hear the presenters talk about powerful, standards-based computing solutions and hot swap and high availability capabilities for communications applications and how it all relates to your development plans.

Tuesday, October 25 - 1:00 - 1:45 pm

DSP Processing for VoIP Applications

DSPs enable equipment manufacturers — and their carrier customers — to launch secure, high-quality, scalable broadband services over IP. Such processors reduce the development time and cost associated with delivering a range of broadband services, beginning with VoIP and extending to full triple-play voice, video, and data offerings. The session will pay special attention to triple play application development and utilizing advanced DSP algorithms to remove noise impairments while enhancing the intelligibility of speech in environments with high ambient noise levels.

Tuesday, October 25 - 2:00 - 2:45 pm

Delivering Wideband Speech Over IP

Wideband speech provides a key quality differentiator from other VoIP offerings and from traditional telephony by more than doubling the encoded signal bandwidth and providing not only richer sound but much better intelligibility that improves communication efficacy and helps speaker recognition. It provides a sense of presence for applications like videoconferencing and teleconferencing as well as any conversation or broadcast. This session focuses on the market drivers, advantages and characteristics of wideband VoIP telephony. In particular, it provides an overview of the key features and functionality of the highly efficient G.722.2/AMR-WB/VMR-WB family of wideband codecs.

Tuesday, October 25 - 3:00 - 3:45 pm

VoIP Testing & Security Issues

Security remains a concern in packet networks. This session will address a number of issues regarding development of VoIP security solutions, including encryption, user authentication, denial of service (DOS) attacks, and the multiple network layers (signaling, media, transport) subject to security testing. The presenters will also look at semiconductor technology, including system-on-a-chip (SoC) architecture and voice processing software, which provides a platform for cost-effectively deploying VoIP services in support of the latest VoIP security technologies including PacketCable Secure Voice, and Voice Over IPSec.

SIP WORKSHOP

Wednesday, October 26 - 8:30 - 9:15 am

An Introduction to SIP

You continually hear about SIP, but do you find yourself bewildered by the buzzwords and terminology? Wish someone would tell you what a SIP Proxy was, and why it might feel the need to fork? What advantages does SIP bring? Why should I move now? This 'back to basics' session will take a high level look at the SIP protocol and the power it puts in the hands of application developers and solution architects. The presenters will discuss the building blocks required to put together a SIP deployment so you can talk with confidence about how the next generation of telephony products will be built!

Wednesday, October 26 - 12:30 - 1:15 pm

SIP Interoperability Issues

SIP is a promising protocol in VoIP due to its simplicity and flexibility. For better or worse, the barriers for developing SIP-compliant devices have dropped significantly. You can find SIP products from developers who work out of their garage. For carriers who operate SIP-based networks that interoperate with this garage-baked equipment, it is crucial that they can compensate and improve the speech quality from the low-budget equipment so that the overall voice experience does not degrade. This presentation examines these SIP interoperability issues in detail and provides practical solutions for carriers who are facing speech quality issues in their networks.

SIP WORKSHOP

Wednesday, October 26 - 1:30 - 2:15 pm

Overcoming SIP Implementation Challenges

As service providers begin real life implementations of SIP networks and products, they are faced with numerous issues including interoperability and quality of service to name just a few. In this session, the speaker will illustrate with examples things that work and areas that still require significant industry effort. In addition, this presentation will identify and address the challenges that vendors and service providers face as they deploy SIP networks.

Wednesday, October 26 - 2:30 - 3:15 pm

SIP's Role in Open Source

There are several IETF RFCs and Internet Drafts that describe mechanisms for securing SIP. Many of these mechanisms are difficult to understand and implement. The reSIProcate project at SIPfoundry has implemented several of these schemes to provide a complete SIP-based security system including end-to-end security using S/MIME and hop-by-hop security using TLS. It also provides an implementation of a mechanism to distribute and store public and private keys using SIP. This presentation will describe the capabilities of the reSIProcate libraries and how companies can take advantage of them in commercial products.

Wednesday, October 26 - 3:30 - 4:15 pm

Service Provider Issues Relating to SIP

As support for SIP continues to grow, creation of SIP-based services becomes a key element for developers to focus on as they scramble for market leadership. This session will provide a close look at examples of promising enterprise, wireless, and wireline applications of SIP, including Push-to-Talk, Infotainment, and the like. The speaker will also focus on the promising wireless applications enabled by SIP as well as the technology and market challenges facing the SIP community.

Thursday, October 27 - 8:30 - 9:15 am

State of the SIP Union: Best Practices & More

SIP and the applications it supports, from management tools and interactive meetings to voice-enriched e-commerce, are continuously reshaping the way today's organization conducts business, by creating increased productivity and enhancing efficiency. This session will address the ins and outs of SIP: how it operates, the advancement of SIP-based solutions, the different diagnostic and support tools that have enabled this protocol to gain such wide acceptance, best practices for deploying SIP solutions, and what the future of innovative SIP-enabled VoIP applications holds. The extension of SIP to provide innovative services and accommodate the requirements of real world deployment, where NATs, service level agreements and regulators exist is also covered.

Thursday, October 27 - 12:15 - 1:00 pm

Securing SIP-Based Communications

Gartner predicts that 90 percent of all new corporate telephone networks will be IP-enabled and based on SIP protocols by 2008. Still, concerns remain around the security of VoIP and the underlying SIP protocol, fearing that they are susceptible to similar types of threats and exploits that plague the Web and e-mail. Today organizations of all sizes need to evaluate and understand the security measures available that allow companies to deploy real-time messaging, voice, data, video and other SIP based applications with confidence. In this session, the speaker will examine specific SIP security problems including voicemail spam, identity theft, impersonation, session eavesdropping, voicemail bombing, hijacking and redirection. Topics will also include the steps that need to be taken to secure and manage the dynamic nature of real-time SIP communications.

Thursday, October 27 - 1:15 - 2:00 pm

SIP? NAT? Not! Traversing The Firewall

Real-time person-to-person communications are fast becoming a critical communications tool for enterprises of all sizes. With the standardization of SIP as the Internet protocol for applications such as VoIP, instant messaging, presence, and video, businesses are eager to adapt their existing hardware to accept SIP quickly, cost-effectively and securely. Traversing the firewall is a tricky proposition when integrating SIP into any enterprise. This session will cover the basics of SIP, its history, evolution, and predictions for the future, with a focus on solutions for traversing the firewall for SIP-based communications.

Thursday, October 27 - 2:15 - 3:00 pm

SIP In the IP Contact Center

This session will take a look at how SIP will dramatically change ingrained enterprise applications such as CRM, ERP, and contact centers. SIP's inherent multimedia capabilities will provide innumerable opportunities to greatly improve productivity, customer interaction, and user experience. Examples will be provided in this exciting look to the future of SIP enabled applications.

SERVICE PROVIDER SUMMIT

Monday October 24 - 12:00 - 12:45 pm

Creating the Trusted, Authenticated Network

We are being hit hard by an increasing identity crisis. A crisis caused by the simple fact that simple, secure, and ubiquitous processes and technologies for user authentication are still lacking. We all are witnesses to ample evidence of this every day. Every Web portal we visit has its own username and password; passwords are inherently insecure and prone to attacks; SPAM is filtered by the millions; phishing has become a serious problem; and as we transition to VoIP many fear a wave of voice SPAM (SPIT). From a regulatory and compliance perspective, the authenticity of documents needs to be established at all times, which is by no means simple to do. This panel of leading venture capital investors will examine the business case for innovative new technology companies as they work to create the trusted, authenticated network.

Monday October 24 - 1:00 - 1:45 pm

Number Portability for a Growing VoIP World

Effective number portability for VoIP is a critical precursor to more widespread VoIP adoption, and if done correctly, it will enable VoIP carriers to generate revenues more quickly. But, in order to bring about the promised benefits, the porting of numbers to VoIP carriers' networks must be simplified. This presentation will explore the current challenges of number porting for VoIP, and discuss recent market developments that are promising to smooth out this process, thus collapsing the time needed to convert a customer's status from "new" to "billable." Attendees will learn the unique characteristics of VoIP number porting as well as new market advances that help to bridge the gap between the traditional landline and IP worlds.

Monday October 24 - 2:00 - 2:45 pm

VoIP Peering Through ENUM Registries

There is a largely misunderstood shift in telecom today: VoIP Peering through ENUM Registries. Peering is the concept of interconnecting networks, allowing IP and subsequently, VoIP traffic to be carried between service providers and companies without the need of the "middle man," or in this case, an additional long distance service provider. By using session border controllers placed between the service providers, or IP PBXs between enterprises and then querying the call via an ENUM database, one can provide a "translation service" between the callers. The speaker will provide insight on both the network and business process implications of the Voice Peering and ENUM.

Monday October 24 - 3:00 - 3:45 pm

ENUM: The Theory is Done, Now It's Time for the Practice

It can be said that ENUM weds traditional phone numbers and the Internet for VoIP, under the ENUM standard, which allows VoIP to use the DNS for signaling. But like any good marriage there are significant adjustments that need to be made. Most current DNS technology suffers from scalability issues. Similarly, the long DNS names of ENUM work badly when run on DNS servers tuned for short names. Customer data privacy is also a concern. But the marriage can be saved, it just requires attention to the needs of both parties. This session will explain to carriers what they should be putting in their ENUM RFPs, describe the traps of practicality, and demonstrate the ability of practice to match theory.

Monday October 24 - 4:00 - 4:45 pm

SIP & ENUM: What's The Story?

ENUM is considered by many to be the key ingredient for the successful convergence between PSTN and Internet. ENUM enables the routing of telephone numbers to IP thus enabling a more efficient method of routing VoIP calls to VoIP recipients (bypassing the PSTN altogether) and also facilitates PSTN to VoIP calling. SIP continues to evolve and is now an ever-present standard in worldwide VoIP deployments. Come to this session to learn all about ENUM and SIP: what they do, how they interrelate, as well as other considerations in these types of deployments.

Tuesday, October 25 - 10:15 - 11:00 am

On Top Of Their Game? A Review of Leading VoIP Service Providers

Keynote Systems recently conducted a study to benchmark the service quality of the leading providers of VoIP phone services, as perceived by end users. The study was designed to assess the market readiness of the leading Internet phone service providers by comparing the call quality of those VoIP providers in San Francisco and New York to traditional PSTN. The study compares the quality of Internet phone service providers based on 10 performance factors to accurately benchmark typical scenarios. Come to this session to hear how the providers fared, and to learn which service providers are truly on top of their game.

SERVICE PROVIDER SUMMIT

Tuesday, October 25 - 11:15 am - 12:00 pm

Triple Play: The Implications of Convergence

This panel discussion will look at how broadband technologies and packet communications come together to form flexible architectures demanded by the triple play services. Attendees will have the opportunity to uncover how triple play services lead to a tighter integration between the "digital pipes," service delivery, and service management layers. This session will also focus on some of the business issues facing providers such as how bundling services will change the way carriers, cable, and wireless providers will manage their expenditures and increase their revenues.

Tuesday, October 25 - 1:00 - 1:45 pm

Voice Quality Challenges Facing Providers

The migration from circuit-switched networks to voice over IP offers a substantial increase in capabilities and features as well as improved economics for both consumers and enterprise customers. In this session, attendees will learn about the complete set of voice quality challenges that voice carriers will face including a discussion about speech impairments as well as packet impairments. The speaker will also discuss some of the latest technologies and solutions for service providers to overcome these voice quality challenges of the 21st century network.

Tuesday, October 25 - 2:00 - 2:45 pm

VoIP Opening Night: Are Service Providers Truly Prepared?

Imagine opening night for a play. The actors are in costume and know their lines, but they have not yet rehearsed together. Unthinkable as the above scenario might be, a very similar risk is in the offing for carriers preparing to launch IP telephony services without testing the performance and security of their VoIP networks end-to-end, before raising the curtain to a paying audience. This session will examine the potentially fatal difference between testing VoIP network components and testing real world VoIP services under real world, end-to-end network conditions. Topics will include: emulating real-world traffic from multiple points on the network; test the full range of IP communications and collaboration applications likely to travel the network; and the full range of network statistics, call measurements statistics, and voice quality metrics.

Tuesday, October 25 - 3:00 - 3:45 pm

VoIP Performance Management

This presentation explains the typical issues affecting the performance of Voice over IP, focusing on some of the more challenging problems such as echo and transient loss/ jitter due to network congestion. The new VoIP performance management framework addresses many of these issues allowing network managers to cost effectively monitor large and diverse networks and quickly identify problem sources. The standards based performance management framework is introduced and practical applications given for both service provider and enterprise networks.

Wednesday, October 26 - 8:30 - 9:15 am & 12:30 - 1:15 pm

TWO-PART SUPER SESSION

Consumer VoIP Marketing Summit, Parts I & II

If you're a VoIP service provider looking for information on how best to target consumers, then you can't miss these sessions! Based on significant research on consumer adoption of VoIP, this session will educate you on what you need to know about consumers, their habits, their expectations regarding service quality and price, and more. Areas of discussion will include how to price a basic service, how to price add-ons such as follow me services, unified messaging, wireless devices, roaming, and much, much more. Come learn how to get customers to your door!

Wednesday, October 26 - 1:30 - 2:15 pm

VoIP Regulatory Update: Will You be Ready?

The situation regarding regulation of VoIP providers is in constant flux. Moderated by Donna Epps of Deloitte & Touche LLP, this panel discussion on VoIP regulatory and compliance issues will address the status of efforts to come into compliance with the FCC's new E911 rules, the FCC's proceeding on the regulatory classification of VoIP service, and the impact of potential regulatory mandates in the future.

Wednesday, October 26 - 2:30 - 3:15 pm

Security In The IP Telephony Network

To address the need for secure, predictable, and efficient services for a wide range of diverse applications on a common network, security must be pervasive and be built into the network, not just an afterthought. This presentation will cover the evolution of the networks viewed from the physical and logical differences, to the role of switches and routers, IDS/IPS, and inherent threat protection to ensure the security of networks. This will cover the infrastructure elements, technologies available in the marketplace and granular policies needed to protect against threats, attacks and unauthorized users. Real-world examples will support the premises of pervasive security and securing the network.

SERVICE PROVIDER SUMMIT

Wednesday, October 26 - 3:30 - 4:15 pm

Security/Surveillance Over IP

Video security and surveillance systems are undergoing a transition, moving away from their analog roots to embrace fully digital, IP-based technology. The benefits of moving to IP are many, including advanced compression/decompression techniques that reduce the need for storage capacity, the ability to view the location under surveillance from any point on the network, and more. This session will serve as an introduction to the world of security/surveillance over IP. If you're looking to deploy this type of solution, this is truly a can't-miss session.

Thursday, October 27 - 8:30 - 9:15 am

VoIP & Next-Generation OSS

Fierce competition in IP services industry demands that providers be able offer new services and create service bundles that draw and keep customers. To offer attractive new services, providers need a solid network infrastructure that can define, enable, and launch these new revenue-producing resources. One key building block in a solid network system is a robust operations support system (OSS). This session will educate providers on how to utilize existing resources to bolster service offerings and increase profit while avoiding networking pitfalls.

Thursday, October 27 - 12:15 - 2:00 pm DOUBLE SESSION

Migrating to VoIP: A Service Provider Round Table

This session will feature the industry's leading experts in a round table discussion of where exactly we stand in regards to Service Provider IP Telephony. Attendees will come away with a true sense of what's happening in the industry, with perspectives from the leading service providers exploring next-generation solutions, including the latest developments, a glimpse into the future, and some real-life implementation tales that you won't want to miss. Attendees are encouraged to bring their questions, and we'll bring the industry leaders who are best positioned to provide the answers.

Thursday, October 27 - 2:15 - 3:00 pm

Session Border Control for Hosted VoIP Services

Hosted IP voice services offer new revenue opportunities and often provide access into new, otherwise unreachable markets. These services, such as subscriber-based IP Centrex or residential broadband VoIP, can be delivered over a variety of access technologies such as leased line, ATM, Frame Relay, DSL, and cable. Deploying a session border controller provides the necessary control functions to enable high-quality, real-time communications across IP networks. This session will address safe, reliable delivery of SIP applications across network borders; protection of a provider's service infrastructure and customer relationships; ability to overcome the NAT barrier to the delivery of high margin multimedia services; and more.

CONFERENCING & COLLABORATION OVER IP SUMMIT

Monday October 24 - 12:00 - 12:45 pm

Introduction to Collaboration & Conferencing

Organizations are waiting for the "killer app" that will help their employees to embrace VoIP — much the same way that the spreadsheet drove adoption of the PC. IP-based collaboration tools just may be that killer application, leveraging the familiar environments of Web, voice, data, and video conferencing while bringing companies the huge cost-savings associated with VoIP. This session will cover examples of how companies are pulling together their ongoing converged network initiatives with the savings opportunities available in IP-based conferencing and collaboration to drive the adoption of VoIP.

Monday October 24 - 1:00 - 1:45 pm

The Future of Unified Conferencing

IP-based conferencing is certainly not your father's conferencing solution. It converges real-time voice, Web, and collaboration tools over a single packet-based network, creating a new, productivity-enhancing user experience. IP is turning conferencing into a one-click, on-demand, desktop-centric paradigm. Users will be able to tap into a rich new range of conferencing services, such as presence- and location-based information about conferences, click-to-call scheduling, instant messaging, and document sharing. This session will discuss the experiential differences between traditional PSTN conferencing and the new world of unified IP conferencing. Discussion will focus on recommendations for better preparing users to take full advantage of the technology's workplace productivity gains.

Monday October 24 - 2:00 - 2:45 pm

Selecting Enterprise IP Conferencing Solutions

The culture of anywhere, anytime connectivity is driving the future of conferencing and collaboration making IP a very powerful tool. Integrated seamlessly into the work environment and linked with business processes and enterprise applications, IP conferencing yields tremendous value to an enterprise. This session covers the issues of choosing proper IP conferencing tool for large and medium sized enterprises.

Monday October 24 - 3:00 - 3:45 pm

Adding Virtual Communications to the Mix

Blended learning is a program designed with the objective of optimizing the learning outcome and cost savings. It is truly a continuous process, and blending Web Conferencing into the mix has various benefits over using traditional learning methods alone including extended reach, optimizing business results, decreasing development cost and time. What tools and media are available to aid this virtual learning? How many companies and institutions are adopting this type of strategy to reduce travel budgets while increasing efficiency and effectiveness? Learn how to achieve strategic learning objectives as they relate to workforce development, sales channel, technical and educational training initiatives. Web conferencing, video conferencing and teleconferencing are among the technologies and solutions that will be discussed.

Monday October 24 - 4:00 - 4:45 pm

Making Government Agencies Effective & Efficient with Real-Time, IP Collaboration

Communications is critical for the success of any organization, but for government agencies attempting to coordinate national security initiatives, secure borders, or adhere to telework initiatives, it's imperative. Fortunately, government entities that need to share critical information, increase productivity, or reduce the need for travel and overhead, have technologies available that can help them be more effective and efficient, while still protecting the confidentiality of information shared through Internet communications. This presentation would highlight why real-time, IP collaboration tools are needed, how they can be used to improve local, state, regional, or federal environments, and provide tips on how various technologies can best be used across related applications.

The INTERNET TELEPHONY Conference program is so thorough, valuable and enriching, your satisfaction is guaranteed. See page 26 for details.

IPTV SUMMIT

Tuesday, October 25 - 10:15 - 11:00 am

Introduction to IPTV

Internet Protocol Television (IPTV) is the delivery of television or video content broadcast over broadband IP networks, such as those provided by fiber deployments to the home, cable, or DSL. Increasingly, IPTV is being bundled with other services delivered over the broadband pipe, such as voice and high-speed data. This session will serve as your introduction to the world of IPTV as well as services such as video on demand — attendees are guaranteed to walk away with a greater understanding of the opportunities afforded by this growing technology.

Tuesday, October 25 - 11:15 am - 12:00 pm

Development Tools for IPTV Deployment

As the telecom industry increasingly turns to offer new services such as IPTV and Triple Play bundles, developers will increasingly be called upon to enhance existing services and create brand-new revenue generating applications to drive a successful bottom line. The product chain extends from the core of the service provider's network out through the set-top box that sits in the customer prem. This presents a tremendous opportunity for developers. Come to this session and hear what tools are available and what you need to know to develop IPTV applications.

Tuesday, October 25 - 1:00 - 1:45 pm

Building Networks for Delivering IPTV

IPTV poses unique challenges for service providers who need to field equipment that allows them to deliver high-quality, time-sensitive video content to hundreds of thousands of subscribers. From Core network elements through edge/aggregation devices and video head-ends all the way through to set-top boxes, service providers need to be able to offer their customers a quality experience. This session looks at the elements that go into creating an end-to-end IPTV network.

Tuesday, October 25 - 2:00 - 2:45 pm

The Current State of IPTV

IPTV is making headlines all over the world with stories of modest deployments and plans to expand service to the masses. This session will take a look at the current state of the industry, through a panel discussion with some of the leading equipment makers and service providers serving this nascent market. We'll hear what's real, and what's not, as well as what we can expect in the near future as IPTV gains traction.

Tuesday, October 25 - 3:00 - 3:45 pm

An IPTV Case Study

Seeing is believing. As IPTV deployments move beyond the early adopter stage, service providers face the challenge of driving new revenue while controlling their operating expenses and effectively managing their network bandwidth. This session will examine the implications of deploying advanced TV services (e.g., HD, PVR, H.264/MPEG4) and value-added applications (e.g., voice services, messaging, gaming), and how existing IPTV service providers are meeting the challenge. Come hear from an industry leader who's "bitten the bullet" and deployed a real-world solution.

VoIP PEERING SUMMIT

Wednesday, October 26 - 8:30 - 9:15 am

The State of VoIP Peering

This session will serve as an overview and an introduction to VoIP Peering. The presenter will review the state of VoIP Peering today. He will give history, analogies, and examples of what is happening in the industry to support the claim that voice peering exists and is having a significant impact on the economics of communications. He will stress that cost savings and not "technology-for-technology's-sake" is driving VoIP implementation.

Wednesday, October 26 - 12:30 - 2:15 pm DOUBLE SESSION

The Future of Enterprise Network Peering

This panel discussion will take a close hard look at what VoIP Peering is and where it is headed. The conversation will center on the future of VoIP peering, and will take into account a number of other discussion topics including:

- VoIP versus VoPI;
- Connecting private networks at Layer 2;
- Connecting public networks at Layer 5;
- Centralized and distributed ENUM registries;
- Equipment deployed and services utilized as of Q1, 2005; and more.

Wednesday, October 26 - 2:30 - 3:15 pm

Developing a VoIP Network Exchange Infrastructure

VoIP services are finally leaving initial trial stages, and reaching production maturity on many fronts. This session takes the next step in the VoIP Peering Summit and looks at the VoIP interface between carrier networks, and the developments needed to displace PSTN hand-off with native VoIP peering. It will review border control architectures and industry initiatives to develop scalable models for VoIP interconnection, and compare this to other interconnection practices such as Internet peering.

Wednesday, October 26 - 3:30 - 4:15 pm

Clearing & Peering: Business Models and Technologies

For the most part, interconnect among VoIP networks is accomplished through PSTN interconnect. There is as yet no clear model for how VoIP networks will interconnect directly in the future. However there are competing business models and technologies, each with their own advantages and disadvantages. The objective of the presentation is to understand the market drivers which determine how networks interconnect and then use this information in a simple framework to understand how the interconnect models compare and which if any models become widely adopted.

MOBILITY SUMMIT

Thursday, October 27 - 8:30 - 9:15 am

VoIP Mobility Applications

Maintaining communications with staff and customers is essential. Connecting the mobile workforce by providing telecommunications solutions that address business continuity, mobility, and telecommunications cost control will empower employees to maintain communications effectiveness under any outage or disaster situation and proceed with "business as usual." Essential to maintaining communications is uninterrupted access to company applications, communications tools, and essential content that's located on the Internet and intranet. This presentation will identify and address business benefits and the expected productivity payback of expanding access to the corporate network using wireless devices and mobility applications.

Thursday, October 27 - 12:15 - 1:00 pm

VoIP Mobility Reality Check

The VoIP explosion is underway. VoIP is now unleashed to be viable solution for wireless and WiFi users. Mobile professionals can incorporate VoIP communications wherever they connect. This panel discussion explores the transitions in telecom, examines VoIP's current trend towards supporting mobile users, and addresses implications to consider when implementing a mobile VoIP solution. Special attention will be paid to the subject of softphones and truly mobile VoIP applications. Come to this session to learn about the true state of the VoIP mobility market.

Thursday, October 27 - 1:15 - 3:00 pm DOUBLE SESSION

Dual Mode: The Current State of the Market

Wireline and wireless carriers alike are currently exploring the market for fixed mobile convergence, which allows subscribers to use special two-radio phones to connect to either cellular or WLAN services, depending on best availability. With this technology, users can enjoy a single set of services regardless of their network, with special mobility technology enabling the seamless transition between networks. Networks must support multiple access types from one common core in order to provide profitable, fast, low-cost, bundled services to consumers. All of these factors are pushing wireless and wireline convergence. These issues must be resolved for wireless and wireline service providers to remain competitive. This panel discussion will examine the issues involved in migrating from separate networks to converged networks.

WIFI TELEPHONY SUMMIT

Wednesday, October 26 - 8:30 - 9:15 am

Introduction to WiFi Telephony

Mobility in the enterprise is quickly becoming a major consideration on a customer's wish list. As more companies investigate the benefits of wireless communications, they face a number of questions specific to deploying 802.11. This session will discuss many of these issues, including network security, delivering enterprise-specific features throughout an 802.11 network, technical considerations when integrating 802.11 into an existing converged infrastructure, and vendor-neutral practical applications.

Wednesday, October 26 - 12:30 - 2:15 pm DOUBLE SESSION

Technical Challenges to WiFi Telephony Deployment

Are wireless networks and technologies mature enough to deliver voice? It's certainly easier to successfully deliver data than voice. Data can be rapidly transmitted without worrying about the effect of network delay, latency, QoS, etc. WLAN providers, equipment manufacturers, network designers and operators must now prove that WLANs and wireless handsets can meet the strict performance requirements required for high-quality delivery of voice. To achieve acceptable voice quality, they must understand relevant performance requirements that affect the network's ability to manage voice and other real-time applications. Come discuss performance requirements and emerging standards being developed to support voice.

Wednesday, October 26 - 2:30 - 3:15 pm

The Challenges of Citywide WiFi Scalability

With all the noise over municipal WiFi zones and free Internet offers, it's hard to separate fact from fiction in the push by cities to deploy WiFi. This session will focus on the reality behind all the hype. What are the challenges of deploying City WiFi zones? What does it take to build, manage and fund these networks? What are the trends in municipal RFPs, and funding tactics? There are several issues to be addressed. Also how does the advancement of WiMax technologies impact the WiFi 'hot zone' deployments and what are the roles of each technology in the future? In this session, you'll learn the real challenges and real opportunities in the race to Municipal wireless networks.

Wednesday, October 26 - 3:30 - 4:15 pm

Call of the Campus Warrior: Healthcare Case Study

Mobile "Campus Warriors" are beginning to utilize PDAs as an efficient replacement for laptops. The newest trend in PDAs is embedding wireless network connections, which allow users to roam across cellular, WiFi and other mobile networks without dropping dynamic voice, video, and data sessions. These new converged devices are enhancing productivity for "Campus Warriors" in healthcare and other fast-paced work environments by allowing professionals remain connected. Today's health care settings are demanding 24/7 connectivity between the patient and the care provider and various hospital support functions like financial services, housekeeping, etc., creating real-time environments. Come to this session to see a real-world example of WiFi in a healthcare environment.

Thursday, October 27 - 8:30 - 9:15 am

Keys to Successful WiFi Telephony Implementation

This presentation will educate you on WiFi telephony: what it is, how to tell if your wireless network is ready for voice, advantages of WiFi, WiFi versus traditional wireless phones, WiFi versus cell phones, and the future of WiFi. If you're new to WiFi telephony, then this is session is a terrific opportunity to learn all you need to know before deploying the technology.

Thursday, October 27 - 12:15 - 1:00 pm

Deploying WiFi Telephony in the Enterprise

Economics and performance are pushing WiFi telephony to the forefront of successful technologies in the workplace. Companies are saving money not only by deploying voice on WLANs, but also increasing revenues by boosting productivity in employees across the workplace. Employees benefiting from WiFi telephony's mobility are collaborating better with coworkers, responding faster to clients and overall improving performance. This session will provide attendees with invaluable tips for successful WiFi telephony deployments in the workplace. With careful planning, WiFi telephony can be used for converged applications that will continue to benefit the workplace well into the future.

Thursday, October 27 - 1:15 - 2:00 pm

Developing WiFi Telephony Endpoints

This session will discuss how VoIP and VoWiFi solve real problems for enterprises and consumers alike. An "inside-view" of VoWiFi phones and the next generation of dual-mode phones (cellular + VoWiFi) will be provided, with a discussion of the key issues that enable, as well as hinder the success of this convergence platform. Topics to be discussed include developer's view of VoWiFi and next-gen Dual-mode phones; benefits of VoIP enabled convergence for mobile communications; and key issues hindering rapid adoption of dual-mode handsets.

Thursday, October 27 - 2:15 - 3:00 pm

The Road Ahead: The Future of WiFi Telephony

WiFi VoIP offers many benefits in its current state, yet it is still a nascent, evolving technology. Today many operators are beginning to implement a converged service for their customers beginning with the development of enabling the transfer of calls between these technologies. While cellular technology automatically hands off calls between towers, the hand off between WiFi and cellular (whether GSM or CDMA) is still a work in progress. As the technology and the supporting marketing plans and packages evolve, carriers can choose to embrace the technology or view it as a competitive threat. If embraced, new applications and competitive pricing structures can help add value, increasing customer loyalty at a lower cost structure.

IMS Summit

Monday October 24 - 12:00 - 12:45 pm

Introduction To IMS

The IP Multimedia Subsystems (IMS) specification has taken the industry by storm, promising access-agnostic multimedia services and fixed/mobile convergence. As IMS gains momentum, service providers are rapidly evaluating and converting network architectures in order to experience the cost savings, time-to-market and added capabilities that IMS promises. This session will take a closer look at IMS service infrastructure and components, providing an overview of the critical components that will deliver on IMS' potential.

Monday October 24 - 1:00 - 1:45 pm

IMS: Developing Network Infrastructure

As the telecommunications industry evolves to embrace IMS, the delivery of unique service combinations requires a horizontal infrastructure that maximizes reusability and minimizes service delivery costs. This session will explore how service providers can develop next generation network convergence architectures that leverage IP and SIP based technologies to bridge mobile and fixed networks while preparing for the future of IMS.

Monday October 24 - 2:00 - 2:45 pm

Fixed Mobile Convergence – Finding the Killer Applications

IMS is becoming the architecture of choice for wireless and wireline convergence. Many service providers are tapping into its ability to offer advanced, revenue generating features, however, making advanced services accessible by broadband wireless users constitutes only half the equation for service provider success. The other half: VoIP and IP multimedia application platforms designed to attract developers and to host the true killer application: continuous, customer-responsive application innovation. The presenter will highlight the benefits of converged services and new applications and provide best-practice examples of how carriers have added voice services and advanced applications to their portfolio.

Monday October 24 - 3:00 - 3:45 pm

Deploying Services Using IMS

This session will educate the audience on IMS architecture and the benefits, challenges, and service deployment implications for this exciting service infrastructure technology. IMS architecture can support a large variety of diverse IP-based services including push-to-talk, color ring-back tones, speech activated dialing, unified messaging, media transcoding, and multimedia conferencing. Speakers will also discuss a pragmatic migration to IMS, including the benefits of integrating VoIP and mobile services, as well as the multiple services involved in true mobile call convergence.

Monday October 24 - 4:00 - 4:45 pm

A Look Ahead: The Future of IMS

It's become clear that IMS will play a major role in the ongoing convergence of wireline and wireless networks. And operators who are keen to build next-generation services on all-IP infrastructures to save costs and increase the speed of service delivery are already taking their first steps to developing an IMS strategy. But what will the future hold? What challenges and opportunities will we face as we move towards convergence? Come to this session and find out.

IP CONTACT CENTER SUMMIT

Wednesday, October 26 - 8:30 - 9:15 am

Transition Your Contact Center to IP

According to a recent Gartner report, "...wide-scale adoption of VoIP by mission-critical mainstream call centers will occur in 2005-2006, and because migration to VoIP takes time, the planning should begin now." IP contact centers have arrived. VoIP solutions are an efficient, effective way to facilitate multi-channel customer interactions across an enterprise-wide network — and if properly assessed, planned, implemented, and operated, converged IP contact centers can yield substantial cost savings and significant architectural flexibility to address strategic business imperatives. In this panel discussion, the presenters will explore the essential steps to creating a converged IP contact center that supports business requirements.

Wednesday, October 26 - 12:30 - 2:15 am DOUBLE SESSION

IP Contact Center Shootout

Come hear a group of industry leaders debate and discuss the relative merits of their IP Contact Center solutions and the overall state of the IP Contact Center industry. This double session promises to be a lively, engaging look at what the industry leaders have to say about their products and their competition. This unique opportunity enables interested parties to get live information directly from the "horses' mouths" as attendees will be given an opportunity to ask the panel their own insightful questions. Past shootouts have covered topics as diverse as the benefits of transitioning to IP in the contact center, offshoring, remote agent strategies, and more.

Wednesday, October 26 - 2:30 - 3:15 pm

VoIP and Offshoring - Pros & Cons

Some suggest that offshoring is the best answer for corporate America's cost reduction. And the call center continues to be a major business unit that is offshored to benefit from labor cost savings available in other countries. However, leading industry analysts estimate that 80% of those companies who chose to offshore call centers to control costs will fail to do so. Others believe that the practice of outsourcing contact centers is a viable one, when considering reducing operating costs, educational qualifications, work ethic, service-oriented workforce, scalability, 24/7 coverage, and more. This session will be an interesting look at both sides of the argument: To outsource or not to outsource? That is indeed the question.

Wednesday, October 26 - 3:30 - 4:15 pm

It's Up and It's Good! Dallas Cowboys IP Contact Center Case Study

This session will present a real-world case study of IP Contact Center deployment by the Dallas Cowboys organization. Come hear how this NFL team is taking advantage of IP contact center technology to facilitate efficiency, cost savings, and more. If you're sitting on the bench in regards to deciding to implement IP in your contact center, you can't miss this opportunity to get in the game and learn from the pros. Bring your questions!

2nd Annual VoIP Service Provider Awards

Sponsored by: INTERNET TELEPHONY, IPCC International Packet Communications Consortium

INTERNET TELEPHONY Magazine, in conjunction with the International Packet Computing Consortium (IPCC), will hand out their 2nd Annual VoIP Service Provider Awards at a special Awards Dinner on Wednesday, October 26th. These awards recognize the best of the best VoIP/IP Telephony Service Providers.

Nominations are being accepted in the following categories:

- Best Overall Service Provider
- Most Popular Mobile VoIP Provider
- Best Marketing
- Best Hosted VoIP Provider
- Best Videophone
- Best Reseller Program
- Best Peer-to-Peer Solution
- Best Customer Service
- Most Popular Cable VoIP Provider
- Most Popular Broadband VoIP Provider
- Best Perceived Voice Quality
- Best Enhanced Services
- Best Prepaid VoIP
- Best Affiliate Program
- Best E-911 Support
- Most Flexible Pricing
- Most Popular ILEC/VoIP Provider
- Best New VoIP Provider
- Best International Provider
- Best Soft Client Support
- Best SMB Service
- Most Innovative
- Best Number Porting Experience

You can nominate a company for this honor by sending your favorite Service Provider's name and Web site, along with the category you wish to nominate them in, to SPAward@tmcnet.com. Please include comments as to why you think your choice deserves to win in their particular category. Also include your name and place of residence (City/State/Country).

First Degree & Advanced Degree Courses

IP PBX CERTIFICATION

THE ONLY INDEPENDENT CERTIFICATION PROGRAM OF ITS KIND VALIDATING YOUR COMPETENCY IN IP PBX SELECTION, DEPLOYMENT, IMPLEMENTATION & MANAGEMENT.

The program is split into First Degree and Advanced Degree Courses

Attendees must complete the First Degree program before gaining admission to the Advanced Degree. Once you've successfully completed the First and Advanced Degree programs, you'll be fully equipped to understand the technologies, pitfalls and solutions you see during an actual installation.

IP PBX First Degree Topics: Monday, October 24

- 12:00 pm Reaping the Benefits of the IP PBX
- 1:00 pm Cost Justifying the Upgrade
- 2:00 pm Selecting The Right IP PBX Solution
- 3:00 pm Effective Deployment and Migration Strategies
- 4:00 pm Living with your New IP PBX - A Case Study

IP PBX Advanced Degree Topics: Tuesday, October 25

- 10:15 am Preparing Your Network for an IP PBX
- 11:15 am Right Sizing Your IP PBX
- 1:00 pm Integrating your IP PBX with an ITSP
- 2:00 pm Staging, Implementing and Cutting Over Your IP PBX
- 3:00 pm An Advanced Case Study

Presenting Companies Include:

Citel • Emerald First, LLC • Epygi • GTP
Inter-Tel • Sphere • Vonexus • Zultys

What You'll Learn:

- Network Traffic Management
- Understanding vendor approaches
- Understanding the business case for IP PBX deployment — including application choices
- Weighing the importance of Interoperability
- "Living with your IP PBX" Day-to-day issues
- Post-implementation management and operational issues (MACs, etc.)
- Understanding underlying technology (Standards, protocols)
- Building fault-resilient systems
- Devising a migration strategy that minimizes business and employee disruption

What You'll Gain:

- Independent accreditation for completing the course
- You'll have independently certified evidence that you possess competencies in IP PBX selection, implementation and management
- Add an impressive certification from a respected source to your resume
- Immediately become the expert called upon to lead your company's IP telephony strategy
- Enhance your chances for a promotion
- Land lucrative consulting opportunities

*At the conclusion of the program, attendees will sit for a 60-minute exam covering the course topics. Those who receive a passing grade on the exam receive TMC University's Certificate of Achievement, certifying that you have successfully completed the course and received a comprehensive education regarding IP PBX selection, deployment and maintenance.

Tuesday, October 25th

VONEXUS Presents... Reseller Solutions Day

A Free Tutorial Seminar Teaching Resellers How To Make Money Selling VoIP Equipment and Services

How To Make Money Selling VoIP

TMC president Rich Tehrani, snom president Robert Messer and other industry experts will draw on over 25 years of experience in the telecom market to help you take advantage of the VoIP market explosion. Learn how to 'talk-the-talk', how to bundle services to create more attractive offerings and how to sell VoIP as an add-on to existing infrastructure.

Reseller Live

The Enterprise Communications Association (ECA, www.encomm.org) is presenting it's very successful panel format, Reseller Live.

This session is structured to maximize reseller participation. Topics include E911, VoIP security, FoIP, Top 5 Reasons to implement VoIP, and other key industry issues. Reseller participation begins now - please submit your suggestions addressing: A) The #1 challenge resellers must overcome to win customer acceptance of VoIP; B) The #1 closer (feature, price, etc.) to clinch the sale; and C) The most important resource a vendor can provide to help you increase VoIP sales.

The panel will review your submissions and select the most frequent, innovative suggestions for full discussion. Participating companies include Iwatsu, Cisco, SecureLogix, FaxCore and other ECA members.

Please submit your content suggestions to: maxschroeder@tmcnet.com

Supported by: ENTERPRISE COMMUNICATIONS ASSOCIATION

Wednesday, October 26th

Service Provider Solutions Day

A Free Seminar Educating Service Providers About Equipment That Can Help You Win Consumer & SMB Business

Any VoIP Service Provider looking to deploy VoIP service to small and medium sized businesses (SMBs) absolutely must attend Service Provider Solutions Day.

This free seminar features equipment vendors specializing in SMB customer premises equipment: the very tools you as Service Providers are looking to deploy in to your customers' place of business.

Handpicked by the editors of INTERNET TELEPHONY Magazine, these companies will demonstrate their products, trying to convince you why they merit your consideration.

Come and see the actual products in action, and meet the companies face to face.

Service Provider Solutions Day is a great venue to see what these vendors have to offer, and a wonderful opportunity to network with the makers of the solutions you will need to attract and keep your customer base.

If you are a Service Provider, this is simply a can't miss event!

If you are a reseller looking to carry the most sought-after VoIP products, you must attend as well.

If you are an equipment manufacturer seeking the chance to present your products and services to the assembled Service Provider audience, please contact Dave Rodriguez for details: drodriguez@tmcnet.com.

FREE FOR ALL ATTENDEES

THE FUTURE OF IP TELEPHONY

Tuesday, October 25th • 3:45 pm

Most Popular

"The Future of IP Telephony" session at INTERNET TELEPHONY Conference & EXPO always draws a 'standing room only' crowd.

A panel of VoIP industry experts offer their perspective on where IP telephony is headed, and try to predict the impact this rapidly developing technology will have on the way businesses communicate in the future.

Panelists as of 7/5/05

MCI, DITECH COMMUNICATIONS, QUINTUM TECHNOLOGIES, INC., IBM, intel, ALTIGEN COMMUNICATIONS

VIEW UP-TO-DATE PANELIST INFORMATION AT WWW.ITEXPO.COM

New For Fall 2005

BATTLE FOR THE ENTERPRISE/SMB

Wednesday, October 26th • 5:15 pm

Choosing the correct IP-PBX for your enterprise is no small challenge. With myriad solutions available, some from new players, other from legacy providers, whose solution is best for your particular installation? One size does not fit all. In some instances strong legacy support will be critical. In others, standards compliance will be crucial. In other situations, branch office support at a low cost or centralized management features will be important to consider. Will the new IP-PBX work well with your current infrastructure? Do you need to rip it out and rebuild? What about support, security and service. This panel will strive to answer important questions from the audience and give you a unique perspective on what items to consider before selecting a solution that is right for your enterprise.

Panelists as of 7/5/05

NEC, AVAYA, TOSHIBA, NORTEL, INTER-TEL

A panel of experts answers tough questions from the audience during "The Battle for the Enterprise/SMB".

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- Internet Telephony ASPs
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- IP Centrex Solutions
- IP Conferencing
- IP Contact Center Solutions
- IP Fax Solutions
- IP PBXs
- IP Phones
- IP Telephony Headsets
- IP Video Conferencing
- LAN-Based Telephony
- Media Servers
- Presence-Based Applications
- QoS/ Network Monitoring
- Session/Border Controllers
- SIP Software
- SMB VoIP Solutions
- Softswitches
- SOHO IP Telephony Solutions
- Speech Recognition/ VXML/SALT
- Unified Communications Apps
- UPS/Power Solutions
- VoIP Development Tools
- VoIP Peering Solutions
- VoIP Security
- VoIP Silicon
- VoIP Testing Hardware
- VoIP-Enabled Handheld Devices
- Web-Based Customer Service
- WiFi Telephony
- Wireless IP Telephony

Exhibit Hall Hours:

Tuesday, October 25 . . . . .6:00 - 8:00 pm
Wednesday, October 26 . . . . . .11:00 am - 5:00 pm
Thursday, October 27 . . . . .11:00 am - 3:00 pm

Exhibit Hall activity captured at a previous INTERNET TELEPHONY® Conference & EXPO

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Participating vendors are prohibited from mentioning their products. Rather, here's yet another opportunity for you to leave this conference with the most complete education possible, preparing you to make a smart VoIP buying decision.

On-Site Registration Hours

Monday - October 24 . . . . .11:00 am - 5:00 pm
Tuesday - October 25 . . . . . .7:00 am - 7:00 pm
Wednesday - October 26 . . .7:00 am - 5:00 pm
Thursday - October 27 . . . . .7:30 am - 1:30 pm

Exhibit Hall Hours

Tuesday - October 25 . . . . .6:00 pm - 8:00 pm
Wednesday - October 26 . .11:00 am - 5:00 pm
Thursday - October 27 . . . .11:00 am - 3:00 pm

Conference Session Times*

Monday - October 24 . . . .12:00 pm - 6:00 pm
Tuesday - October 25 . . . . . .8:30 am - 6:15 pm
Wednesday - October 26 . . .8:15 am - 6:30 pm
Thursday - October 27 . . . . .8:15 am - 3:15 pm

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*Our Guarantee:

If you do not feel the sessions you attend made you better prepared to tackle your VoIP project than you were when you arrived, stop by the registration counter at the show and we'll issue you a free pass for any future INTERNET TELEPHONY conference. (No requests honored after the conference ends.)

Hotel Information

Show Hotels Expected to Sell Out Weeks Before Event Begins

With over 200 exhibitors and more than 7,500 attendees expected, rooms at the official show hotels WILL SELL OUT QUICKLY.

We urge you to contact your hotel of choice right away and reserve rooms at the special INTERNET TELEPHONY® Conference & EXPO rates. Identify yourself as a show attendee to take advantage.
Deadline for these special rates: September 23, 2005

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333 South Figueroa St • Los Angeles, CA 90071
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	Early-Bird Thru 9/26/05	Standard After 9/26/05	Select Days
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2 Attendee Information

Name Title Company

Address

City State Zip Country

Phone Fax E-Mail (required)

3 Payment Information

Faxed Registrations Must Include A Valid Credit Card. Total Amount Due: $______________ (Payable in U.S. dollars drawn on a U.S. bank.)

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INTERNET TELEPHONY

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Full payment is required prior to admittance to the conference. Registrations are transferable and non-refundable. Registrants may have a dollar-for-dollar credit towards another TMC conference. Credit must be used within two years from original registration date. Program and speakers are subject to change without notice. TMC® reserves the right to use attendee company names, titles images and photos for future promotions.

4 Please answer all questions. Incomplete forms cannot be processed.

1. Business Type (Check only one)

COMPUTER INDUSTRY
- ❑ 1. Network/System Integrator
- ❑ 2. Distributor
- ❑ 3. Reseller/Wholesaler/VAR/VAD
- ❑ 4. Consulting
- ❑ 5. Manufacturer/Software Developer
- ❑ 6. Service (non-ISP)
- ❑ 7. Other (Specify) ____________

TELEPHONY INDUSTRY
- ❑ 8. Telecom Developer
- ❑ 9. Manufacturer
- ❑ 10. Distributor
- ❑ 11. Reseller/Interconnect
- ❑ 12. Consulting
- ❑ 13. Other (Specify) ____________

SERVICE PROVIDER/ CARRIER INDUSTRY
- ❑ 14. Next-Gen Telco/ITSP
- ❑ 15. CLEC
- ❑ 16. Integrated Communications Provider (ICP)
- ❑ 17. Telco/RBOC/IXC/Long Distance
- ❑ 18. ISP
- ❑ 19. Wireless/PCS
- ❑ 20. Cable
- ❑ 21. Application Service Provider/ Outsourcing (CASP)
- ❑ 22. PTT
- ❑ 45. BLEC/MDU LEC
- ❑ 46. ILEC
- ❑ 23. Other (Specify) ____________

E-COMMERCE/E-BUSINESS INDUSTRY
- ❑ 42. Business to Business
- ❑ 43. Business to Consumer
- ❑ 44. Both (B to B & B to C)

GENERAL INDUSTRIES
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- ❑ 25. Business Service/Consulting/Consumer Service/ Non-Profit/Trade Assn
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- ❑ 29. Utilities
- ❑ 30. Finance/Banking
- ❑ 31. Insurance
- ❑ 32. Hospitality
- ❑ 33. Healthcare/Medical
- ❑ 34. Real Estate
- ❑ 35. Catalog Marketing/Publishing
- ❑ 36. Direct Marketing
- ❑ 37. Market Research
- ❑ 38. Advertising/Public Relations
- ❑ 39. Teleservice Agency/Outsourcing/ Collections/Call Center
- ❑ 40. College/ University/School
- ❑ 41. Other (Specify) ____________

2. Job Functions (Check only one)
- ❑ 1. IT/IS/MIS/DP Management
- ❑ 2. Telecom/Datacom Management
- ❑ 3. Software/Engineering Management
- ❑ 4. LAN/Network Apps/Systems Management

2. Job Functions (Cont.)
- ❑ 5. Internet/Intranet/Extranet/Web Mgmt
- ❑ 6. Corporate*/General/Financial/Management (*includes: Chairman, CEO, President, Owner, Principal,Partner)
- ❑ 7. Sales/Marketing/Advertising/ Product Management
- ❑ 8. Call Center/Telemarketing/ Credit Collection or Fundraising Mgmt.
- ❑ 9. Customer Svc/Help Desk/Tech Support Mgmt
- ❑ 10. Business Development/Operations/Human Resource/Training/Project/Purchasing Mgmt
- ❑ 11. Consulting
- ❑ 12. Other (Specify) ____________

3. Total employees in your company all locations:
- ❑ A. 10,000+
- ❑ B 5,000-9,999
- ❑ C. 1,000-4,999
- ❑ D. 11-999
- ❑ E. 1-10

4. Would you like to subscribe to Internet Telephony's free eNewsletter?
❑ Yes ❑ No

5. Would you like to occasionally receive free product information and special promotional offers via e-mail from the industry's leading vendors?
❑ Yes ❑ No


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RAV900

ClearOne Communications, Inc.
1825 Research Way
Salt Lake City, UT 84119
Tel: 801-975-7200
Fax: 801-977-0087
Web: http://www.clearone.com

Price: North American retail pricing will be $2,599 for RAV 600 and $3,099 for RAV 900.

ClearOne's RAV 600 and RAV 900 (pronounced "rave") are meeting room conferencing systems designed to provide a higher-quality alternative to audioconferencing systems or standard conference phones. It features Gentner Distributed Echo Cancellation, noise cancellation, microphone gating, and a drag-and-drop graphical user interface for easy system setup, control, and management. In addition to superb audio quality, it can also connect to popular rich-media devices such as video and web conferencing systems. The RAV comes complete with an audio mixer, microphone "pods," an intuitive software interface, RS-232 serial port, VISCA Camera Port, a remote control device, and get this — premium Bose loudspeakers.

In fact, the President of TMC, Rich Tehrani came into the labs area and remarked, "Since when does TMC Labs get its hands on and test Bose speakers?" We too were a bit surprised when we opened the ClearOne box to find high-end wall-mountable Bose speakers. Our first though was that wall-mountable Bose speakers were "overkill" for audioconferencing — but how wrong we were. We'll explain in just a bit.

Installation

We tested the RAV 900, which comes with three microphone pods (the RAV 600 comes with two). Both models include Ethernet control for remote management and configuration capabilities, as well as built-in Web interface for management. Hooking up the RAV900 unit was a no-brainer. We always attempt to hook things up without referring to the manual as an acid test of how easy something is to install. We were able to successfully set up the unit without looking at the manual once thus helping the RAV to earn a perfect 5 installation rating.

We liked the fact that the unit uses standard RT45 cables to connect the microphone pods to the main RAV unit. No proprietary wiring to worry about — you can simply use your existing RJ45 cabling and RJ45 jacks already installed in most conference rooms to connect to one or more microphone pods. Each microphone pod has two RJ45 ports (In and Out) allowing you to daisy-chain multiple microphone pods that terminate to a single RJ45 wall jack. If you connect the wires in the wrong direction a red light turns on; likewise a green light appears if you connect the wires correctly. These microphone pods can be positioned in different areas of the meeting room for maximal acoustic pickup. For our tests we connected three pods positioned in various locations in the labs. Then you go to your networking closet and simply disconnect the meeting room's RJ45 cable from the patch panel currently connecting to a network switch or hub and connect it to the RAV conferencing unit. We should point out that the unit allows single input connection to CD, DVD players, VCRs, etc., for playback over the system's speakers.

Next, we connected the various other components of the RAV unit including a wireless antenna, RJ11 telephone wire, telephone handset, power cord, and of course we spliced some speaker wire to connect to the Bose speakers. Side note: Call us audiophile geeks, but any time you get paid to splice speaker wires at work is definitely a good day!

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

Operational Testing

For our initial tests we decided to use a Teltone Analog Simulator to provide dial tone to the RAV. We connected a 40-foot long RJ11 phone wire to a second telephone handset so we could move this telephone far enough away from the Bose speakers and microphone

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Figure 1. Clear One: The RAV900 Web interface for configuration.

pods to prevent feedback or echo. From this telephone we dialed the RAV unit and it rang. We answered the call using the wireless remote control device, which features a LCD screen for displaying pertinent information such as call length and CallerID, as well as configuring various features. After answering the call we tested the audio quality and were extremely impressed. The audio output from the Bose speakers was very rich and unlike many competing audioconferencing systems we have tested, the audio from the Bose almost made it feel like the remote caller was actually in the room. This fact changes the whole dynamic of holding an audioconference when the sound is "warm" and inviting instead of "cold" or distant like most audioconferencing systems. We now understood why ClearOne chose to use high-end external Bose speakers.

We could already see that the RAV unit was performing flawlessly when it came to echo cancellation. In fact, even though the microphone pods were positioned only two feet away from the Bose speakers, the remote caller did not hear himself in any sort of feedback loop or echo. For our next test, we wanted to be sure of 100 percent full duplex operation since often times echo cancellation can cause full duplex issues. We had the remote caller recite the alphabet and the RAV unit user count to 10. Unfortunately, we noticed that once the RAV unit user started speaking it would clip the remote caller. That is, it would cut off the remote user's voice from coming out the speakers. Even worse, as long as the RAV user was speaking, there was no way the remote caller could interrupt him. We knew that such a high-end conferencing system couldn't possibly be "half duplex." We suspected our Teltone analog simulator was to blame and not the RAV unit, so we did some further investigation. First we called ClearOne and they confirmed that our test results were a bit strange since they have no problems with full duplex on their products. We performed an audio conference with ClearOne with a RAV unit on their end and duplicated our test and it performed flawlessly. We were able to interrupt, talk simultaneously, etc.

Thus, our prime suspect once again pointed to our "trusty" Teltone simulator, which we have used for years in the labs. We disconnected the Teltone simulator and used two real PSTN connections instead and then repeated our tests. This time the unit passed with flying colors. We suspect that due to the fact that the Teltone analog PSTN simulator has almost no latency that the ClearOne RAV unit's echo cancellation mechanism wasn't designed for this. Goes to show you that sometimes the most accurate test is a real test and not a simulated one!

We would be remiss if we did not mention the RAV Web interface (Figure 1). It displays microphone levels, echo cancellation settings, and more. From this user interface you can tweak the audio settings, including the ability to turn off the echo cancellation if you so chose. You can also locate other RAV units on your network using the unique network name or IP address and then manage them via the web interface.

Specifications:

Audio Mixer
- AEC tail time: 128 ms Adaptive noise cancellation: (6 – 18 dB)

Microphone Pod
- Coverage: 360 degrees
- Connection: RJ-45 with Cat. 5, Link In/Out Ports
- Maximum Distance from Base Mixer:50' (RAV 900), 75' (RAV 600)
- Frequency Response:60 Hz – 14 kHz (+/-1 dB) THD+N: <0.08% (–45 dBu input @ 1 kHz)
- Sensitivity: -45 dBu
- Input Level: -6 dBu for 1 kHz 94 dBSPL microphone input
- Dynamic Range: > 65 dB

Room For Improvement

According to their specifications the RAV900 supports up to 50 feet of cable and the RAV600 supports up to 75 feet. We would like to see support for longer length microphone pod RJ45 cables for

PROS and	CONS
Excellent audio quality. Ease of use. Unique ability to tie in video.	Would love to see longer cable length.

situations where the networking closet is far from the meeting room.

Conclusion

Besides the unique multiple microphone pod system and unique utilization of RJ45 cabling for daisy-chaining microphone pods is RAV's truly unique ability to tie in a DVD or videotape player or video from your computer along with the audio. RAV's ability to directly interface with your video- and audio-conferencing systems makes it a powerful solution. TMC Labs was very impressed with RAV900's excellent audio quality, easy-to-use wireless controller, and its innovative Web interface for viewing and configuring the RAV units, making this a great choice for any conference room. IT

UMUX 1500

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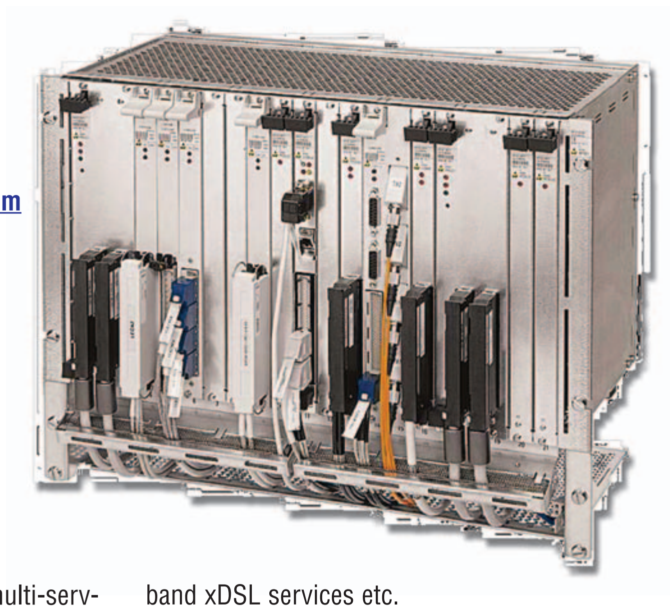

We live in a time of escalating telecom competition, and carriers that take hold of and make the most of technology as a strategic tool will be the industry winners. Their services have to be multifaceted and for that they need a multi-function, multi-service access platform as their customer base could span from the rural to the urban and require a blend of subscriber services and technologies.

At the least a carrier or service provider's wish list for a multi-access platform would run like this — support the old and the new, i.e., PDH, SDH, IP, ATM, POTS and ISDN (define - news - alert) telephony, Ethernet, etc. Elaborating this a bit further you would need a cross connect, SONET/SDH terminal multiplexer, add drop multiplexer, digital subscriber line access multiplexer (DSLAM) functionality coupled with ATM-based concentrated multiplex function for bundled transmission to the core network, a migration path to VoIP, Ethernet over SDH (EoS), broadband xDSL services etc.

For such a platform we did not look far into the horizon as Keymile's UMUX multi service access platform family seemed to meet most of those aspirations. Keymile offers a choice between the UMUX 1500, UMUX 1200 and UMUX 900. But we settled for Keymile's market-proven SDH product platform the UMUX 1500 as it is designed to support very low bit rates and goes all the way up to 155 Mbps. It is a 21-slot modular DC powered 19" chassis thoroughbred. If you have one versatile multifunction platform like that then it certainly makes the finance managers happy for the savings that can be accrued from it.

For the UMUX 1500, Keymile promises SDH standards compliance, support for SDH ECC and overhead bytes, traffic protection functions, a broadband bus backplane, end to end management and interoperability with multi vendor SDH equipment. It is a true carrier class chassis as it implements traffic and equipment protection standards namely 1+1 equipment protection(EQP), Liner trail protection (LTP), Subnetwork connection protection (SNCP), multiplex section protection (MSP), synchronous equipment timing source (SETS) protection, and so on.

UMUX 1500 is very versatile and lends itself to various functions. It is a SDH STM-1 terminal multiplexer and

RATINGS (0–5)
Installation: 5
Documentation: 4
Features: 5
GUI: 4
Overall: A-

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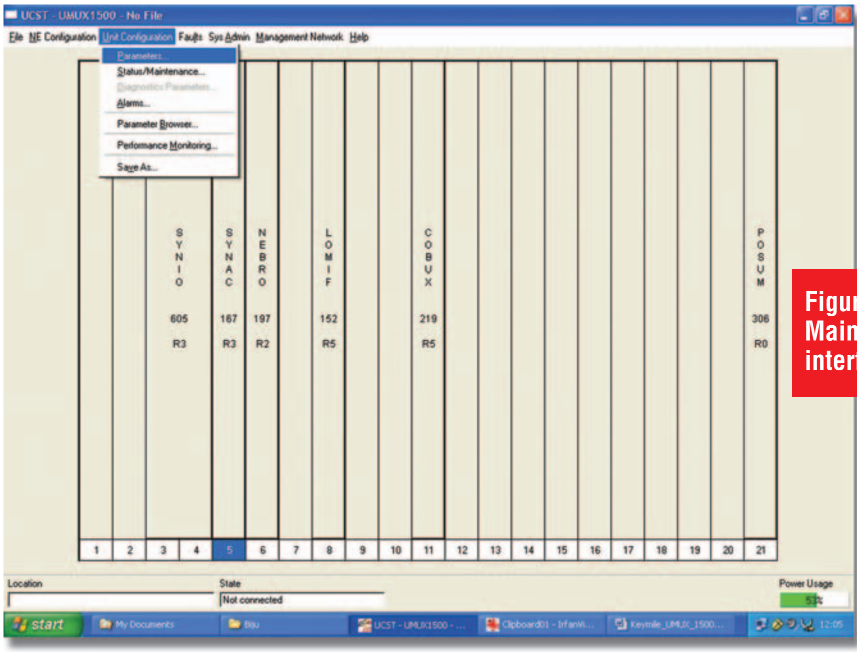

Figure 2. Keymile: Main UCST interface.

add drop multiplexer. It has a 128 x 2 Mbps digital cross connect and we were able to cross connect at the 64Kbps level. Keymile promises support for ATM DSLAM, circuit emulation over ATM and IMA (Inverse multiplexing over ATM). When it comes to telephony we were told that you can do a voice CAS, V.5.1& V.5.2 and if the need dictates you can even get a VoIP subscriber media gateway function. Its EoS function allows you to switch and transport 10/100 Mbps and 1Gbps over electrical and optical interfaces. This means you can now provision new Ethernet services over an existing SDH network.

By integrating a good mix of functions and interfaces it addresses many requirements. Subscribers can get POTS, ISDN-BRA, ISDN-PRA, and other interfaces. When it comes to data subscriber interfaces you are spoiled with many choices and no matter what your requirement is you are covered by a mélange of interfaces including V.24 / V.28, X.21/V.11,V.35,10/100BaseT, 1000Base-X,G.703 2 Mbps, 4x2 Mbps optical, G.703 34/45 Mbps, SDH STM-1 optical, ATM STM-1 optical etc. In the DSL domain you get flavors like ADSL (ADSL, ADSL2, and ADSL2+), HDSL, SHDSL line, etc. Depending on the requirement network designers can choose network interfaces such as SDH or PDH with NxG.703 2Mbps or 4x2 Mbps optical or even one or two SDH STM-1(with support for TDM, IMA, EoS traffic). If it is an IP or ATM requirement (then you get support for Ethernet, optical, ATM STM-1, optical, Nx2 Mbps IMA…) or if it is DSL (then you get support for NxTDM SHDSL 2 Mbps NxTDM HDSL 2 Mbps) and so on.

Given SDH's continued dominant role in telecom markets Keymile's UMUX 1500 is well placed to serve it. With the release of plug in units that support EoS, VoIP media gateway etc Keymile is doing its part in revitalizing SDH's future by quickly supporting new functions and driving down the cost of equipment by integrating more and more functions on to the same platform. Working on the core belief of business continuity Keymile's UMUX 1500 platform ensures carriers can maintain revenue from existing voice services and also move towards next-generation network (NGN) services. This aspect is also reinforced by supporting EoS on the same platform allowing implementation of Gigabit Ethernet overlay network, Ethernet private point-to-point or point-to-multipoint networks. These approaches can help carriers extend the life of the existing SDH infrastructure, which is critical to their success.

Keymile's UCST and UNEM companion management tools are there to help network operators in remote provisioning, management, and diagnostic functions. These tools will help you keep very close tabs on every deployed network element; reduce visits to remote equipment sites thus help in keeping operational costs low. UCST is a Windows-based tool (Figure 2) for small networks with up to say 30 UMUX network elements. More than that you should deploy Keymile's UNEM network management package (Figure 3), which is a UNIX/LINUX based system with a map indication your various network elements. It is a multi-user system that supports multi-layer network topology, error management, configuration management, performance management and the like.

You get started by interfacing the UMUX 1500 21 slot chassis, along with COBUV, which acts as a control unit for configuration and operation of the UMUX and it also includes other functions such as NE synchronization function and it uses one slot of the sub rack. Also, the POSUM power converter unit uses one slot of the sub rack and provides +/- 5V DC to the other units in the rack. If you need high system availability then you can have additional COBUV and POSUM in the sub rack. Next, based on your application and traffic requirements you select the

PROS	and CONS
True multiservice platform. Wide range of interfaces supported.	No secure Web-based or telnet management access support.

aggregate units for example such as SYNIC (STM-1 electrical interface), SYNOT (STM-1 optical short haul interface), or TDM tributary units such as SYNAM (you can map 8x2 Mbps to VC12), SYNAD (you can map 16x2 Mbps to VC12) SYTEL (you can map 34 Mbps or 45 Mbps to VC3).

The NEBRO/NEBRA (Ethernet transport and switching) units will definitely satiate your EoS requirements and it helps carve out Ethernet bandwidth granularly in 2Mbps increments. These units also include support for salient features like built in switching; concatenation with VC12 or VC 3 etc, traffic prioritization, VLAN tagging, rapid spanning tree protocol etc. Depending on application you can deploy either the NEBRO (has four 100BaseT-interfaces and two SFP slots for 100Base-FX or 1000Base-LX optical modules) or the NEBRA (has six SFP slots for 100Base-FX or 1000Base-LX optical modules). The SYNAC, SYNVA units can take care of your 2 Mbps to VC12, TU-12 traffic mapping needs.

Customers developing systems based on the UMUX 1500 solution can flexibly scale their designs depending on the total number of interface ports, required applications, and features.

Thanks to the IP subscriber media gateway (IPSMG) unit the service provider can concurrently offer subscribers both traditional telephony and NGN telephony services. No more fork lift upgrades — instead you can migrate towards NGN telephony at your own pace.

Operational Testing

For our testing we used UMUX 1500's each of them equipped with a COBUV, POSUM , SYNIO (STM-1 aggregate interface unit with two optical ports and two slots in the sub rack), SYNAC (STM-1 tributary interface unit providing tributary signals access to STM-1 aggregate) and a LOMIF (8x2 Mbps interface unit uses one slot in the sub rack). For the test and to pass the traffic between sites we connected third-party external routers equipped with E1 WAN interfaces to the 2 Mbps interface of the LOMIF card.

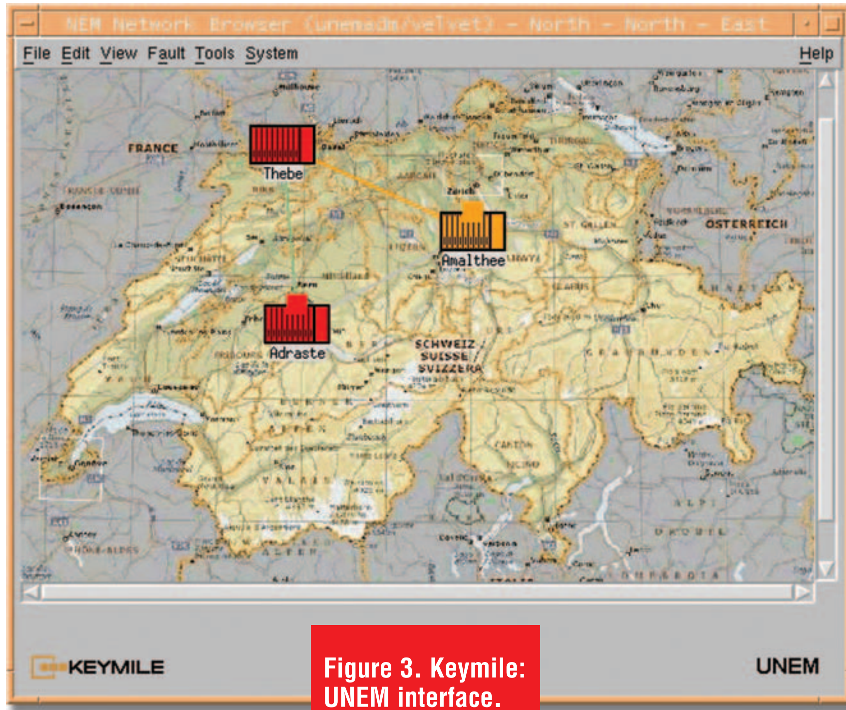

Figure 3. Keymile: UNEM interface.

Using UCST I connected to the UMUX. For that we had to get the management network agent configured and running and then added the appropriate network element which in this case is the UMUX 1500. One good thing with the UCST is that you can prepare a configuration offline even without connecting to the UMUX. We configured the LOMIF, SYNAC, and SYNIO with the required parameters. Then we set about creating the relevant cross connects. Each time we made any configuration changes I had to do a partial download to the network element. The UCST serves many useful functions — allowed me to do a software delivery (between the network manager and network element), software installation (on to various units on the UMUX), set time on the UMUX, do an upload or download form the UMUX, create, modify, and delete cross connections on the UMUX, add, delete, or convert units, perform status and maintenance functions, configure and check timing sources, monitor alarms on a particular unit, alarm listing, etc. All of this will enable system designers to architect sophisticated systems providing enhanced provisioning, grooming and performance monitoring capabilities.

We found the UCST to be an ideal companion of the UMUX 1500 for network provisioning, management, and diagnostics and we found the documentation to be very comprehensive.

As I concluded testing the UMUX 1500, to me it emerged as a very versatile platform with diverse feature sets providing substantial design and cost benefits over similar solutions.

Room For Improvement

This product meets the necessary functional and application requirements so nothing much to suggest except that we would like to optionally see secure Web-based management and secure telnet management access supported.

Conclusion

UMUX 1500 is a true multiservices platform for network operators and we highly recommend this carrier class modular platform as it strikes an optimal balance between modularity, functionality and cost in an SDH access environment. It's EoS and NGN access gateway functionality support is a prescription to success. **IT**

This review was prepared by Biju Oommen, a Telecommunications and Networking Solutions Consultant with a special focus on enterprise products and solutions.

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The Rise And Fall Of Centrex Services

Centrex services are not a new concept in the world of telephony. Whereas the concept of outsourcing your business telephony requirements to a service provider existed in the TDM world, it is now rejuvenated with the introduction of "IP Centrex," also known as "hosted IP services."

This paper discusses the question of why IP Centrex is succeeding in places where TDM-based Centrex has failed. We will demonstrate that the key to IP Centrex's success lies in a combination of several factors, which differentiate IP Centrex services from the old TDM-based Centrex.

What Is Centrex Service?

Generally speaking, Centrex service is the outsourcing of internal enterprise telephony services to an external service provider. The telephony services included in most cases are the following:

- Toll-free calls within the organization;
- Extension dialing (e.g., four digits);
- Networked, toll-free calls between different branches of the organization;
- Connection to the PSTN;
- IVR/Auto Attendant;
- Voice mail and/or unified messaging;
- Basic call center; and
- Internal/External conferencing.

When Centrex services are not employed, enterprises need to purchase, install, and maintain all the equipment required to provide the services mentioned above: PBXs, IVR servers, voice mail servers, etc.

With the Centrex model, the service provider is responsible for providing the services using its own equipment which is usually purchased for a group of customers, and installed in the central office of the service provider or at the customer premises.

Why Use Centrex Services?

Most enterprises today tend to outsource a lot of the services and processes that are not a part of the core business of the enterprise. Despite the fact that is not the core business, most enterprises regard telephony services as a strategic business asset required for normal operation of its organization. Outsourcing telephony services using a Centrex model saves the investment in capital equipment required for the telephony services, and the skilled manpower and operation costs involved in maintaining the telephony services.

On the other hand, providing Centrex solutions is a key differentiator for the service provider, creates new revenue streams, and reduces customer churn for the service provider.

The TDM Centrex Model

Centrex is not a new idea or service. It has been an option since the early days when service providers supplied digital telephony services over TDM (define - news - alert) networks. Originally, Centrex service meant grouping several regular PSTN (define -news - alert) lines under the same umbrella, providing toll-free calls inside the organization, internal dial plans for various Centrex customers, and basic value-added services via the PSTN TDM network. In addition, Centrex customers were assured of the following benefits (amongst others): reliable and worry-free service, maintenance, and upgrades, and no wasted floor-space due to on-site PBXs.

Actually, what happened was trying to use the classic PSTN Class 5 switches, which were designed for scalable PSTN services, to provide partitioned

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By Haim Melamed

IP Centrex actually creates a brand new business model and a new type of service provider.

enterprise features and applications. From the service provider's point of view, it was a great solution. Almost no investment was required by the service provider except for upgrading the PSTN switch software and new revenue streams were supposed to be generated. Among the challenges that made this concept a partial failure in the TDM world was that it didn't offer the customers a service that was substantially more attractive than on-site PBXs. In addition, low-cost and feature-rich entry-level PBXs were widely available, making Centrex less attractive for small to medium-size enterprises. Such PBXs offered the customer more flexible control of the lines (e.g., changing or extending phone numbers required a bureaucratic ordering process) and more subscriber features (without subscription fees).

The IP Centrex Model

In the IP world of location independence, an IP Centrex solution is not different in nature from any regular IP-PBX implementation. In the IP world, the IP-PBX / Softswitch, the phones, the PSTN connections and the value-adding application servers can be located anywhere on the IP network. Therefore, IP Centrex service can be implemented in many different ways. The IP-PBX can be located in the customer site, co-located in the service provider site, or implemented as a partition on the service provider IP Centrex server. The same goes with the value-adding application servers. In all cases, independent of the location of the systems, the service provider can manage the service remotely for the enterprise.

The Biggest Difference

In most cases, Centrex services will replace an aging PBX (define -news -alert) installed in the customer premises without the related depreciation costs. The business model behind moving to Centrex is saving the capital and operation costs of owning a telephony network, without compromising the level of service and the features provided. One of the biggest drawbacks of the TDM-based Centrex solution was that because of its implementation over the existing old TDM switches, it provided similar (basic) telephony features as was offered to normal Class 5 subscribers (although the feature list of TDM-based Centrex did improve and become more competitive with on-site PBXs over time). On the other hand, the IP-based Centrex solutions provide many more features than the regular corporate PBX. The IP Centrex PBX is actually a fully-featured enterprise IP-PBX managed for the enterprise by the service provider, while the TDM Centrex service was based on carrier grade public PSTN switches that did not provide the required enterprise features.

In addition, the TDM-based PSTN switch was not created with partitioning in mind. Therefore, handling basic features required for Centrex such as overlapping dial plans, security, and billing was a nightmare for the service provider. However, today's IP Centrex solutions are built from the ground up to support partitioning and multi-tenant implementations. Managing multiple customers is fast and easy for the service provider. In addition, it is far easier for customers to manage their lines or extend them (e.g., via the service providers' Web site).

Value Added Services

An enterprise PBX solution is not complete without some value added services such as IVR, ACD, voice mail, and more. The Centrex model provides the enterprise a great advantage in this space. Instead of purchasing, configuring, and maintaining highly complicated systems on their own, customers can enjoy the economies of scale of a service provider, providing a service as a part of a system supporting many customers.

Actually, this general model works much better with the IP Centrex than with the TDM-based Centrex. Value added services today are implemented by using highly scalable IP-based application servers, centrally located in the service provider network, supporting hundreds of customers. In the TDM world, the servers had to be installed locally, the CTI support was very challenging, and the time-to-market simply did not justify the model.

End User Terminals

Replacing the phone located on the corporate CEO desk is not a trivial thing. The phone is one of the most useful pieces of technology in today's office environment, and the features provided by the phone are very important for personal productivity.

Replacing a digital smart phone with a TDM-based Centrex service does not allow the use of the same features any more. On the other hand, in the IP world, the same IP phone can be configured to work with a local IP-PBX or an IP-based Centrex application. The move is transparent to the end user, and today's IP phones provide features that are richer than that of the smart TDM phones. Features, such as Internet browsing, touch screens, visual unified messaging on the phone, conference call management, and CTI are common in today's IP phones.

Moves, Adds, And Changes

One of the main costs of maintaining a telephony network for the enterprise is managing moves, adds, and changes. The average cost of moving a person from one seat to another in the TDM telephony world is $100. This number is based on an enterprise using its own PBX, and it is even higher using the TDM Centrex model, since in most cases, changes can not be done by the enterprise and adds require new PSTN

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lines from the service provider. Actually, the time required to add a new phone into the system in the TDM Centrex world can be a show stopper for the model. On the other hand, in the IP Centrex model, moves, adds, and changes are much simpler.

Such changes involve almost no time or money. Adding a new phone can be done by the customers themselves using a Web interface. Moving requires only taking a phone and connecting it somewhere else, or even logging into another phone.

Remote Management

Another day-to-day task that has changed as a result of the IP technology implementation is the management and configuration of the PBX and the value added services. Using your own PBX and value added services require extensive training and operation costs.

With the Centrex model, you outsource the maintenance and operation to the service provider, but there is always a delicate balance between outsourcing the service and maintaining control in the enterprise and in the user community.

A TDM-based Centrex solution does not allow enterprise customers to do anything on their own. In addition, it requires direct connectivity into the PBX network to monitor and control the service. Any configuration or changes done by the end user require the use of a cumbersome DTMF-based interface.

On the other hand, the IP Centrex solution allows the sharing of management, configuration and monitoring tasks between the service provider, the enterprise administrator, and the end users, who all use a user-friendly web-based GUI.

The New Business Model

IP Centrex actually creates a brand new business model and a new type of service provider.

In the world of TDM-based Centrex, the only service provider that could supply the Centrex service was the Telco (or “PTT”). Centrex service was implemented on the existing Class 5 switches as an extra software package. The service provider had to own the equipment, as well as the last mile in order to supply the service.

In the world of IP Centrex, all you need in order to supply the service is a fast IP connection to the customer. This can be done on another service provider’s network, using an IP-VPN service or even the public Internet. The service provider can be a Telco, an MSO, an ISP, or even a PBX reseller that wants to transform his business into a Centrex model.

Summary

Here is another case where IP changes the business model in the telecom world. Centrex services, which fell short in the TDM world, are now becoming very successful using IP technology. The benefit of moving to IP Centrex is clear both to the service providers and to enterprise customers since the new IP Centrex technology allows the surmounting of most of the obstacles that occur with the TDM Centrex model. **IT**

Haim Melamed is director, channel marketing at AudioCodes. For more information, please visit the company’s Web site at http://www.audiocodes.com.

Best Practices For Providing VoIP Quality Of Service

The biggest technical challenge in transitioning from traditional circuit-switched voice and video systems to the new, more economical voice and video over IP packet-switched technologies is obtaining adequate quality of service (QoS) over the wide-area network. Quality of service is the capability built into the network to guarantee that information traverses the network without interruption, in a timely manner. Most existing data networks were designed for bursty applications that are not delay-sensitive, meaning that if a data packet arrives within a reasonable amount of time, both the application and the user are satisfied.

Voice and video data, on the other hand, are very sensitive to delay; if a packet arrives more than approximately 200 milliseconds (ms) after it is transmitted, the packet is worthless as a carrier of real-time communication because it will arrive too late to be used in the conversation or video image. Consequently, networks carrying IP voice and video must be designed and configured properly to ensure that real-time packets traverse the network efficiently.

The Battle For Bandwidth

The challenge of achieving adequate quality of service is exacerbated when a data packet must traverse the WAN. Typical local-area networks (LANs) run at 10 Mbps, 100 Mbps, and some even have bandwidth of a gigabit per second (1,000 Mbps) and higher. However, because bandwidth over the WAN is significantly more expensive than over the LAN, many wide-area networks operate at T1 speeds (1.45 Mbps) and slower, creating a huge bottleneck at the LAN/WAN interface. For normal data packets like e-mail, Web browsing, client-server programs, and a host of other applications, this LAN/WAN bottleneck is a nuisance, but not a performance killer because these applications can withstand delay and still function satisfactorily. However, when voice packets must compete with regular data packets for transmission over a bandwidth-constrained WAN, the voice and video applications may be rendered useless unless steps are taken to insure voice and video QoS.

All VoIP Traffic Is Not Created Equal

But how do you initiate the process of establishing a coherent QoS strategy for VoIP? The adage, "you can only control what you can see" is entirely applicable to this challenge. The prudent enterprise needs to identify which applications are currently running on their network, when they run, how much bandwidth they use, and if they are performing satisfactorily. A variety of tools, including packet sniffers and network probes, can be used to initiate a "VoIP readiness" assessment. However, relying only on low-level protocol analyzers, such as those that look at IP packet headers to classify traffic, gives no indication of which applications are running across the network and may preclude the discovery of significant traffic trends

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that can impact the performance and integrity of VoIP services dramatically. By deploying monitoring systems that analyze traffic at the application level (Layer 7), IT can determine the behavior characteristics of individual applications and how that behavior influences WAN application performance.

Rich traffic classification is crucial — you can't assess or control an application's performance if you can't distinguish its traffic. For example, P2P, viruses, and worms hide in an HTTP tunnel using port 80 to traverse the firewall. The growing complexities associated with network traffic make sophisticated classification techniques a necessity. Simple IP address or static port schemes fall short. Classification must detect dynamic and migrating port assignments, differentiate applications using the same port, and use Layer 7 application indicators to identify applications.

Armed with this understanding, enterprises must protect VoIP, not only from data traffic, but from unsanctioned VoIP applications arising from instant messaging (IM) programs such as AOL IM, Yahoo!, and MSN Messenger. Furthermore, there is a distinct possibility that future P2P applications may start masquerading as VoIP traffic in order to gain passage through NATs and firewalls and to receive priority treatment when traversing the WAN.

Controlling VoIP Performance

Once VoIP business applications are accurately identified, the next step in optimizing VoIP is to create a reporting methodology that examines three primary variables:

• **Performance**: how long application traffic spends in different portions of the network.

• **Utilization**: bandwidth usage by applications, locations, and users.

• **Diagnostic Aids**: Information that helps when trying to analyze and pinpoint a problem.

IT must ensure that it does not succumb to simply collecting data. To manage VoIP applications effectively, comprehensive reports that succinctly analyze bandwidth usage, response times, the impact of configuration changes, and sources of delay, broken down by time spent on the network and server, are crucial. Since each network is unique, reporting tools must have the flexibility to measure, graph, and/or export numerous metrics (in some cases exceeding 100), in order to accurately capture usage, availability, efficiency, response times, errors, and diagnostics.

Armed with the right reporting tools and data, IT is better prepared to address three critical performance issues that must be controlled to deliver VoIP QoS:

1. Latency — the end-to-end delay in delivering the voice stream from the speaker's mouth to the listener's ear;
2. Jitter — the unpredictable, variable delays in the delivery of each voice packet; and
3. Packet Loss — the dropping of individual packets caused by network congestion.

Each of these three issues can cause significant degradation in voice quality and overall system reliability. Because VoIP is real-time, two-way communication, it is very sensitive to delays in the network.

Acceptable VoIP quality requires a bi-directional latency or delay of not more than 80ms for true toll-quality voice communication. Voice quality degrades as latency increases, but even with a delay of 150-180ms each way, voice quality is still in the "acceptable" range. In addition to the voice stream itself, latency issues must also be addressed with other VoIP protocols (SIP, H.323, MGCP, etc.) that handle the call control functions between two systems or users. In fact, these signaling protocols are often even more sensitive to delays in the network.

Some network devices attempt to overcome this problem by employing various queuing techniques (DiffServ, 802.1 p/q, IP ToS, etc.) to ensure voice packets take priority over other traffic waiting to get on the network. This is helpful to a certain extent, but being first in line to get on a crowded freeway doesn't mean you'll get to your destination quickly. What's required is more stringent, intelligent bandwidth management/QoS that can classify all VoIP-related protocols, allocate a guaranteed amount of bandwidth to each traffic type, appropriately prioritize VoIP traffic as it traverses the WAN, and provide granular assurance of VoIP on a per-call basis.

If jitter causes consistent delays in excess of 20–30ms, voice quality will suffer. Some VoIP vendors have tried to solve this problem by introducing their own jitter buffers or queues to temporarily store and "smooth out" the delivery of voice packets. Likewise, routers also offer queuing mechanisms for the same purpose. Both options, however, can exacerbate the problem by actually contributing to delays. Therefore, minimizing and controlling network jitter is required to prevent the disruption of VoIP traffic. And this calls for the ability to assign and maintain a guaranteed rate and quality of all voice and data traffic across the WAN, which are prerequisites to delivering VoIP QoS. These technologies are functionally similar to conventional queuing, in that they "smooth" delivery of the traffic, but do so in the context of a more intelligent, policy-based bandwidth management/QoS strategy.

Because IP is a "best effort" protocol, if left unattended it will always be subject to unpredictable performance, including packet loss. Like jitter and latency, packet loss can be very disruptive to VoIP's performance. This is usually not an issue in the corporate LAN and should not be a problem if an enterprise relies on an MPLS service across the carrier's backbone network. But at the bandwidth-constrained LAN/WAN boundary, where there is much greater contention for space in a smaller pipe, congestion and packet loss can become a serious problem. Although packet loss of one percent or less is within the bounds of toll quality voice, once packet loss reaches three percent or more the listener will notice the conversation "breaking up." Unless this problem is controlled, packet loss can lead to dropped calls and the possibility of VoIP system failure. The key to addressing packet loss is to apply more controls to the IP network, moving it from "best effort" to predictable, optimal performance for all business-critical applications — including VoIP. The value is delivered through a set of QoS and bandwidth management tools that minimize the congestion and unpredictability of IP and maximize application performance over the existing WAN, often without the need for costly bandwidth upgrades.

Only by applying a complete suite of technologies that include deep classification, detailed reporting, control mechanisms to proactively protect and ensure performance and the ability to apply compression gains to the benefit of VoIP applications is it possible to deliver VoIP QoS. IT

VoIP QoS Best Practices

• Deep classification of VoIP applications and network traffic. Identify what is important so that you can manage, protect and control application performance.

• Detailed VoIP analysis and reporting. Gather, sort, present, analyze, and share information about bandwidth utilization, response times, network efficiency, and service-level compliance, hosts, top applications, and new applications across a system.

• Protection of business-critical VoIP applications. Contain unsanctioned traffic and allocate resources where they're needed most.

• Free up additional capacity within the existing network, when possible, and allocate the bandwidth to VoIP applications.

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Maintaining Quality Of Service In A Demanding World

With the emphasis on rapid service delivery to customers, carriers must be proactive in maintaining high service quality.

The higher expectations and satisfaction levels demanded by today's customers are putting tremendous pressure on carriers to improve the quality of their services. Since a satisfied customer equates to higher Average Revenue Per User (ARPU), reduced churn, and is paramount to driving business, carriers must find a way to deliver new services quickly while maintaining the same level of quality their customers have come to expect.

The evolution and constant transformation taking place within the telecommunications industry — particularly with the need for carriers to offer so many new services so frequently just to remain competitive — is causing a shift in the way services have traditionally been provided. Carriers are now offering and deploying many services first, and then trying to piece together the technology puzzle that will enable the successful operation of those services across the network — along with how to ensure the quality of those services.

This is creating a distinct gap between services and operations. Where technology was normally first and services second, carriers are finding themselves caught in the uncomfortable position of having to offer services quickly to maintain market share, and then figure out the technology necessary to ensure the quality. As that gap continues to widen, the ability of carriers to continue providing quality services and support becomes increasingly difficult.

Bridging The Gap

Market drivers have traditionally taken the form of one particular "killer application." For wireless carriers, it was voice; for wireline carriers, it has recently centered on data service. Today, however, customers are demanding multiple services in a "mix and match" matrix of available, cutting-edge offerings. Any operator or carrier not prepared to offer any mix of services will lose customers and market share, which is why getting new services out as quickly as possible, often ahead of the technologies and operations that will support them is becoming more the norm than the exception.

Service Quality Management (SQM) has evolved from a mere industry buzzword in the last eight years to a stark realization for carriers that they need to transform the way they operate their business. The traditional network-centric view of managing the network, customers, and operations is no longer an option due to the diversity of services, sharing of resources, and multiple personnel responsibilities.

Today, there is a need to bring all network management, service quality, and customer care initiatives together — transforming the way carriers operate their business from a network-centric point of view to a customer-centric point of view. This is the intent of SQM — to bring everything related to the customer's experience together to enable carriers to look at services as they relate to the customer.

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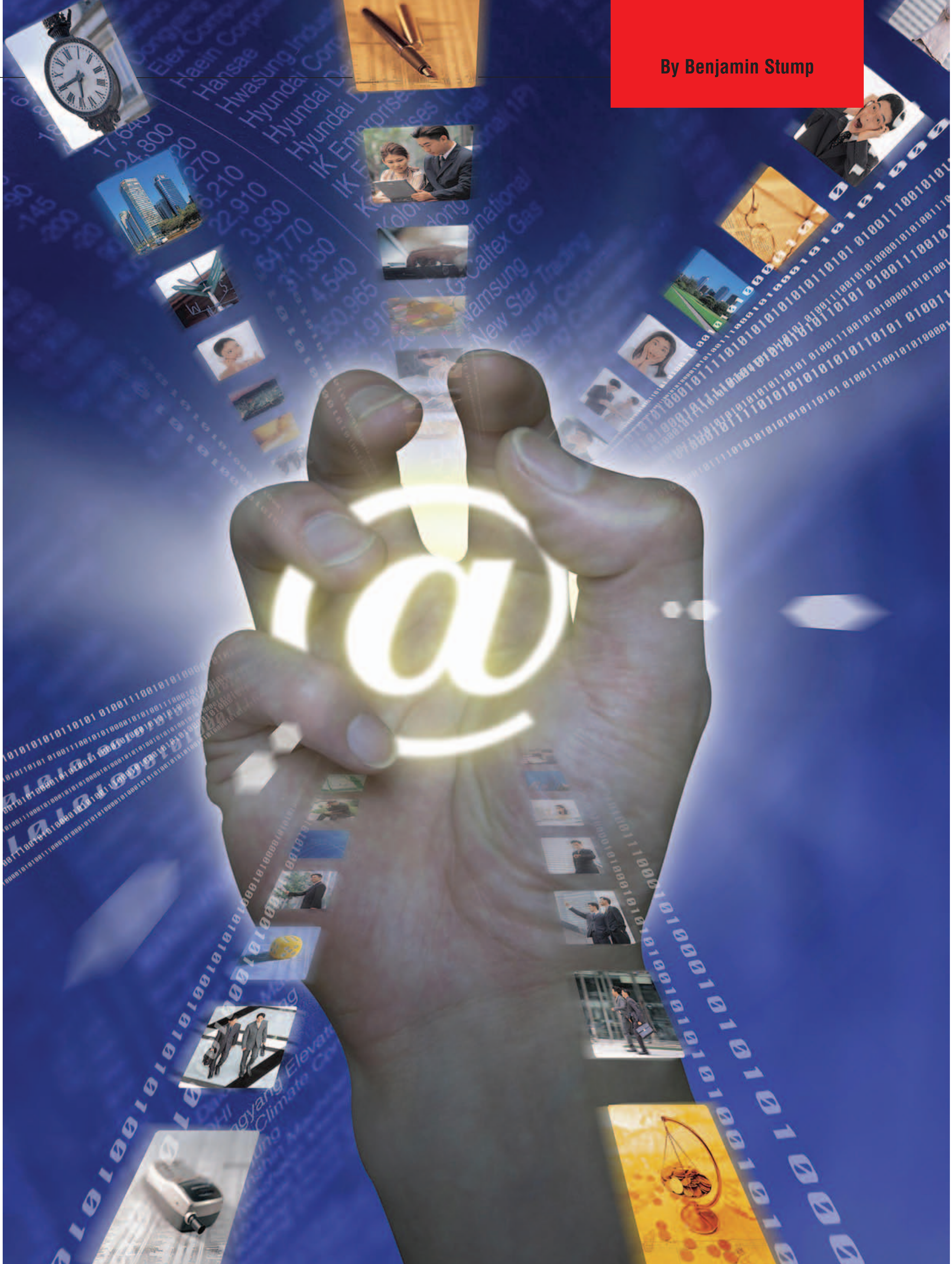

By Benjamin Stump

Carriers are quickly realizing they cannot achieve SQM without outside help from software vendors with proven experience in providing SQM and quality-of-service (QoS) solutions as well as business process management. These vendors must create close partnerships with the carriers to leverage existing infrastructure and expertise while overlaying an application that can automate the process of bringing all the disparate data together. The second phase of that application is to provide each different internal organization — everyone from customer care to marketing to sales representatives to network technicians — the same data in a view that each can comprehend. To be successful, this must be accomplished while minimizing the business process impacts to the carrier.

For instance, a network technician needs information that will quickly point out any fault or problem and prioritize it. A customer care representative wants to easily figure out if the problem is related to network equipment or the customer not knowing how to use a device or service. Marketing and sales people want to take the same data and apply it to proof points for selling SLAs or to project revenues.

Successful SQM

In order to successfully implement SQM, the software must first model the service mix to ensure it is ready when service roll-out occurs. The carrier wants to know the research has already been conducted that enables the SQM package to work with the unique characteristics of the service and service bundle. A first-class SQM provider works with the carrier to recognize and mitigate the gap between offering services and being technologically prepared to operate them. Therefore, the service modeling capability and the flexibility of the SQM solution are keys to a successful application.

A viable SQM solution also solves the disparate data problem by leveraging the provider's existing resources. Carriers may have many traditional

VoIP Performance Assurance Leads To Business Revenue And Productivity Assurance

By Yves Cognet

VoIP appears to now be hitting its stride. However, VoIP still doesn't deliver the same quality and reliability — with the same consistency — that traditional voice delivers. In addition, with other services often being converged on the same network, QoS monitoring becomes paramount. Those that fail to stay on top of quality, service assurance, and disruptive issues are also most likely to feel the wrath of unsatisfied customers or users.

There are strong cost-reduction benefits to businesses expected in the migration from TDM voice dedicated networks to converged IP networks. There is also a concern about the capabilities of IP centric data networks being able to handle voice call loads with customer-acceptable quality. Often, network administrators are asked, "Why don't we leverage our high-quality network to run our voice calls?" With cost reduction being paramount in today's business world, this situation has given rise to VoIP. But adding VoIP to existing data networks is not so easy, unless such a request was anticipated, and it's rarely the case.

First, one needs to distinguish the difference between Telephony over IP, where a business discards their local PBX and POTS phones and move to an IP PBX solution using IP phones. This can be a costly decision if you want to switch later. Others may prefer to keep their local PBX to manage and route calls between POTS or IP phones and route traffic over IP. This is a "soft" migration, which still allows the capability to use the public network, just in case.

Most people take traditional phone service for granted because businesses rarely suffer voice interruptions. In contrast, a lot of attention is being paid to the cost of disrupted data services, such as CRM, SAP, etc. Voice is just as critical a business application that is often taken for granted. Telephones, after all, are still a highly-deployed and widely used communications medium in the business world.

One may believe that to deploy VoIP a business can essentially just borrow bandwidth. But deploying VoIP only under this simple assumption can be a disaster to business goals, revenues, and customer/user retention. Trying to guess the limits and behavior that borrowed bandwidth will have on an IP network will soon lead to a lack of acceptable voice quality altogether.

Most TCP-based applications, such as e-mail, can rely on TCP to re-order or retransmit packets when one goes missing. VoIP transport does not have this luxury, because of the real time nature of voice. With VoIP, there is no retransmission, no packet reordering, and even worse, if a VoIP packet arrives too late because of congestion — a very likely scenario — it will be dropped, leading to choppy, cut-out, or dropped-altogether voice calls.

The Technical Due Diligence Needed for Successful VoIP

To assure that an IP network can deliver VoIP services that meet business and customer/user needs of expected quality, two fundamental questions need answering:

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How much VoIP traffic will be routed? In order to get this answer you need to translate Erlangs into bandwidth, according to the Voice Codec you are going to use.

Is my overall end-to-end network delay below 100 milliseconds? Acceptable voice quality, as perceived by an end user, drops dramatically when the end to end delay exceeds 150 milliseconds.

Traffic in terms of bandwidth can be dedicated from Erlangs by making some assumptions from the following table.

	Serialization Delay										Erlang (100% line utilization)		
	Packet transmission for different link speeds										Codec (best case)		
Link Speed (kbps)	Packet Size										G711	G723.1	G729
	40 bytes	80 bytes	88 bytes	136 bytes	184 bytes	232 bytes	280 bytes	520 bytes	1K bytes	1.48K bytes			
56	5.714 ms	11.429 ms	12.571 ms	19.429 ms	26.286 ms	33.143 ms	40.000 ms	74.286 ms	146.286 ms	211.429 ms		1	
64	5.000 ms	10.000 ms	11.000 ms	17.000 ms	23.000 ms	29.000 ms	35.000 ms	65.000 ms	128.000 ms	185.000 ms		1	1
128	2.500 ms	5.000 ms	5.500 ms	8.500 ms	11.500 ms	14.500 ms	17.500 ms	32.500 ms	64.000 ms	92.500 ms		2	2
256	1.250 ms	2.500 ms	2.750 ms	4.250 ms	5.750 ms	7.250 ms	8.750 ms	16.250 ms	32.000 ms	46.250 ms	1	4	4
384	0.833 ms	1.667 ms	1.833 ms	2.833 ms	3.833 ms	4.833 ms	5.833 ms	10.833 ms	21.333 ms	30.833 ms	2	7	6
1,000	0.320 ms	0.640 ms	0.704 ms	1.088 ms	1.472 ms	1.856 ms	2.240 ms	4.160 ms	8.192 ms	11.840 ms	5	19	17
1,540	0.208 ms	0.416 ms	0.457 ms	0.706 ms	0.956 ms	1.205 ms	1.455 ms	2.701 ms	5.319 ms	7.688 ms	9	29	26
2,048	0.156 ms	0.313 ms	0.344 ms	0.531 ms	0.719 ms	0.906 ms	1.094 ms	2.031 ms	4.000 ms	5.781 ms	12	39	35
10,000	0.032 ms	0.064 ms	0.070 ms	0.109 ms	0.147 ms	0.186 ms	0.224 ms	0.416 ms	0.819 ms	1.184 ms	59	191	173
100,000	0.003 ms	0.006 ms	0.007 ms	0.011 ms	0.015 ms	0.019 ms	0.022 ms	0.042 ms	0.082 ms	0.118 ms	590	1,915	1,737
1,000,000	0.000 ms	0.001 ms	0.001 ms	0.001 ms	0.001 ms	0.002 ms	0.002 ms	0.004 ms	0.008 ms	0.012 ms	5,902	19,151	17,378

The link speed is in kbps, according to the codec capabilities and the product planned for use. There might be several options to pack numerous VoIP payloads in a bigger IP frame to minimize traffic. This can come at the expense of more delay and most codecs do not support large payloads.

Getting Serious About VoIP

Businesses that are serious about what VoIP can do for their customers, efficiency, cost reductions, and revenue must be as serious about the benchmarks and tools used to plan, deploy, and continually evaluate VoIP services. There are a variety of challenges including deciding on the tools for deployment and support of VoIP services; assuring quality through service availability monitoring and service performance monitoring; and even billing considerations. There is also a complexity and diversity problem when it comes to selecting VoIP products to deploy.

Failure to tread these waters carefully and without due diligence can have a profound impact on voice communications for a business, and even an impact on the data flowing through the network. All this can lead to damaged customer loyalty for VoIP service providers, to decreased productivity for enterprises, and in the end to lost revenues and time spent dealing with VoIP problems. With careful planning, benchmarking and the right tools, business can traverse rough waters to realize new revenue streams or operational cost reductions. Businesses that try to "wing it" with what they have, such as by using ping-like tools or simply adding bandwidth, are destined to have VoIP problems. In the end they will have to turn to careful planning and good tools and benchmarking to get it right. So, get it right from the start and avoid the pitfalls of mismanaging VoIP services. **IT**

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fault and performance management applications already in place in addition to possible probe and test and measurement solutions — often with some having their own customer care and trouble ticket management applications overlaid. For SQM to be successful, it must easily integrate with all those disparate data sources. The better the SQM solution adapts to new and unique data sources, the faster new efficiencies can be realized.

Another successful SQM characteristic is the visualization of the results and service degradations. Everyone knows the importance of a flashy and user-friendly graphical user interface (GUI). However, the intent of SQM is not to be flashy but to ensure that different organizations of vastly different technical expertise levels can see their own uniquely presented data. SQM must have the flexibility to provide different visualization of the same data in real time.

As SQM gains notoriety among carriers, the next logical evolution will be its "transferability." Once SQM is implemented in current applications, it will eventually begin making its way more directly to the customer, including setting up customer Web portals to visualize their own data. This transferability could also provide QoS information via PDAs or tablet PCs for use as a marketing tool that quickly points out the advantages of SQM to potential customers.

The true success of any SQM implementation, however, will not be dictated by its features alone. More importantly, its success will be determined by how quickly and easily the solution can be deployed while minimizing the impact to the operator's current business processes.

Real-World Cases

SQM solutions have already been successfully implemented by several large telecommunications carriers, including O2 (Germany) GmbH & Co. OHG, a leading European mobile communications provider. Back in June

2003, O2 recognized that users of data services generated higher ARPU than traditional voice-only customers, yet the ability to provide outstanding QoS would require monitoring a whole new range of additional parameters. O2 soon realized it could not manage QoS by monitoring the various, individual components that drive each service and that outside help from an operational support system (OSS) vendor would be necessary.

O2 required an SQM product that could operate independently of network technology, provide an end-to-end view of the services and network to maximize QoS, and included an extensive service library that would enable them to quickly offer new content, services, and applications.

The solution they selected enables the O2 network operations center to monitor the GPRS (define - news - alert) network, data services (e.g., MMS), and the end-to-end quality of individual services quickly and easily. The true test and proof that the SQM solution was a resounding success for O2 was evident during the Oktoberfest 2004 festival in Munich. With a significant spike in network traffic, it was critical that the SQM solution successfully enabled the O2 network to seamlessly monitor the increase and manage the network to support traffic increases without impacting customers or customer service levels.

Several other SQM deployments, including T-Mobile USA, have met with similar success in implementing management systems that solve disparate data problems and help close the gap between deploying new services and the operational transformation to manage them.

In one particular case, an operator was monitoring, on average, 4,000 core IP network alarms per hour. In evaluating and implementing an SQM solution, it was discovered that only 6 percent of the alarms were either service or customer impacting. By reducing the number of alarms monitored, increased efficiency was immediately realized.

VoIP Standards For Real-Time Performance Management

By Kaynam Hedayat

After many years of hype and promises, VoIP is finally here. For providers and enterprises alike the question of "should we consider VoIP?" has been replaced by "how quickly and reliably can we deploy VoIP?" Because we are still in the evolutionary stages of VoIP technology, this inevitable move has created new challenges in VoIP network operation and management.

Many initial VoIP deployments are often part of hybrid VoIP-legacy networks. However, VoIP networks are becoming the cornerstone of today's telephony networks, and the overall infrastructure relies more than before on the performance and availability of IP-based networks and VoIP services.

This reliance, along with the uncertainty about VoIP network availability and performance, is behind the appearance of new performance and monitoring standards. These emerging standards range from new ways to collect and report performance-related information, to methods that extend this information to network management applications.

RTCP-XR

The building-block standard that addresses VoIP performance management is RTP Control Protocol Extended Reports (RTCP-XR). RTCP-XR provides extensions to the RTCP protocol for reporting detailed RTP and VoIP-related metrics. RTCP-XR is a mechanism for reporting detailed RTP and VoIP metrics.

There are new standards that take advantage of RTCP-XR to enable operations and management applications. These range from methods for reporting RTP and VoIP metrics in the context of a user call by the endpoints involved in the call, to methods for proactively measuring network performance to the user edge of VoIP networks. In both methods, RTCP-XR collects and reports the metrics.

Reporting performance metrics in the context of user calls provides network operators with visibility into the user experience. This is a common method for gaining visibility into end-to-end network performance, and complements monitoring agents that provide visibility at different segments of the network. However, this approach is reactive, not proactive. The user knows about the problem before the network operator. Reactive monitoring has partially worked in more mature TDM networks. However, VoIP is still a young technology. IP networks are multi-vendor and complex, and the network is no longer isolated. Proactively addressing performance and availability problems is mandatory to ensure successful VoIP networks and services deployment.

Media Loopback

One method of meeting the challenges mentioned above is monitoring the media delivery performance through "test and monitoring" calls where media is "looped back" from the receiver to the caller. Media loopback is especially popular in ensuring transport quality to the networks' edge. Today, in networks that deliver real-time media, and short of running "ping" and "traceroute" to the edge, service providers are left without the necessary tools to actively monitor, manage, and diagnose service quality issues.

This monitoring method has been used in TDM and analog networks. Special intelligent endpoints, deployed at strategic locations, loop back media and cooperate with the caller to measure call performance. This method, however, can be costly and does not scale to support large networks. Furthermore, it does not offer the granularity of testing to a specific endpoint or location that may be exhibiting problems.

Proactively addressing performance and availability problems is mandatory to ensure successful VoIP deployment.

VoIP offers an interesting opportunity for deploying this capability. What if the intelligence of looping back media and cooperating for performance management was embedded in endpoints, such as SIP-based phones and IMS-based mobile handsets? This is the goal of the An Extension to the Session Description Protocol (SDP) for Media Loopback draft within the MMUSIC group of the IETF.

The concept of media loopback is very simple: the caller asks the receiver to loop back every packet it receives. The two endpoints exchange performance metrics via RTCP-XR and provide the data to higher-level applications, such as performance management and monitoring systems. Typically, policies such as not ringing the user for loopback sessions and rejecting loopback calls if the user is busy are put in place to enable non-intrusive monitoring.

The An Extension to the Session Description Protocol (SDP) for Media Loopback takes the concept further by introducing different loopback types for measuring service performance and availability services at different end point layers. The extensions are limited and based on the current standards for lightweight implementations.

Conclusion

IP telephony promises more than just having another type of phone on your desktop. The potential is for more advanced and better performing services. As VoIP evolves, its success relies on proactive operation and management strategies that scale to large networks. Multiple emerging standards are attempting to address large-scale network monitoring, and they have RTCP-XR as their foundation. The An Extension to the Session Description Protocol (SDP) for Media Loopback draft from the MMUSIC group of IETF is one of a series of standards that brings the value of RTCP-XR to operation and management applications. **IT**

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Further review showed that more than 50 percent of those same alarms were also predictable and repeatable. By tying multiple data sources together, the identification of degradation trends leading to those failures was automated. By identifying the trends, the operator can now manage predictive problems to solve the issues before the alarms occur — getting in front of customers calling in with service problems — and thereby keeping customers happy.

Finally, the speed at which these solutions have been implemented is notably important. Successful SQM deployments must minimize business process impact while maximizing speed and efficiency.

As customer expectations increase and higher satisfaction levels are demanded, network operators must meet the challenge by seeking out SQM solutions that enable them to remain competitive. OSS vendors must act as a vehicle by providing high-quality, highly reliable solutions that leverage the infrastructure and service mix of each network operator.

The gap between operations and complex service offerings is not always evident within the carrier. In fact, carrier operations have progressed and with experienced technicians they can often appear to be keeping pace. The difficulties will continue to grow as the gap widens and the carriers that are able to get ahead of the service curve and augment their operations to close that gap transparently will ultimately become the market leaders in this new and complex world lacking the killer application of the past. **IT**

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The Role Of SIP In Advancing A Secure IP World

Internet telephony is poised for widespread adoption. Already, there are countless Voice over IP (VoIP) services that collectively boast millions of users worldwide, with a compelling new set of telephony services and features appearing on desktops and mobile telephones everywhere.

Today, the Internet allows users to readily send information to many people with great ease and speed — a significant departure from the managed service of the traditional telephone network, where service providers oversee every call. However, with these innovative new capabilities, how will the Internet live up to consumer expectations for superior security and privacy, two hallmarks of traditional voice service?

The Session Initiation Protocol (SIP) identity standards have evolved as a means of addressing VoIP (define - news - alert) security issues. SIP (define - news - alert) provides VoIP services with a critical "rendezvous" function by connecting users to one another. More specifically, SIP allows a user to discover how best to reach the person they want to contact — whether on their mobile phone, home phone or PC. SIP identifies the right device to accept a call, then allows the endpoints to "negotiate" what sort of session is optimal based on their device capabilities and preferences (VoIP, instant messaging, videoconferencing, etc.) Finally, once the endpoints have "agreed" on what kind of session is optimal, SIP manages the session for its lifetime, controlling when media will and will not be sent, allowing endpoints to add new media to a session as necessary, and closing down the session when the communication is finished.

SIP leaves the work of describing sessions to a different protocol: the Session Description Protocol (SDP). SDP is tunneled within certain SIP messages as a MIME body, the same way that a plain-text MIME body might be carried in an e-mail message. These SDP messages are not intended to be written and read by users, however. Instead, SIP devices and applications use them. A SIP device creates an SDP message that describes its own capabilities (voice, instant messaging, etc.) and combines them with priorities set by users. When a device receives a SIP request containing SDP, it compares the capabilities and preferences described in the SDP with its own, and creates a response that determines the best way for these two devices to communicate.

Once SIP has provided the "rendezvous" function, and SDP has figured out the right way for devices to communicate, the actual session media can begin to flow between devices. That media can take any number of forms, depending on what kind of session has been established. For VoIP sessions, the most common type of media is the Realtime Transport Protocol (RTP), a lightweight protocol optimized for car-

By Jon Peterson

rying real-time information on the Internet. RTP carries the actual packets of audio that are gathered by the devices participating in a VoIP session. Also, RTP starts and stops when SIP tells it to do so. The precise IP address to which audio packets are sent is specified within SDP, as are the codecs supported by the endpoints.

These three protocols (SIP, SDP and RTP) operate in concert to provide a VoIP service for the Internet, making the security capabilities of these protocols quite interdependent.

When we start to investigate the security needs of VoIP, the core questions are fairly obvious: How can I know who is calling me? How can I be sure that the audio I am hearing is coming from the right place? How can I prevent unwanted calls? How can I prevent eavesdropping?

Because SIP draws so much of its protocol design from e-mail and the Web, it is able both to rely on the strengths of e-mail and Web security and to learn from their shortcomings.

So what are the strengths of email and Web security? Most Web users are familiar with the Secure Sockets Layer (SSL) that is used to secure e-commerce connections, though few may know that SSL changed its name a while back to Transport Layer Security (TLS). TLS is most famous for providing encryption that prevents eavesdroppers from learning credit card numbers sent to Web pages. Perhaps even more importantly, though, TLS passes a certificate from a Web server to a browser, which identifies the Web site to which the browser is connected. The browser can then compare the URL entered by the user (e.g., http://www.example.com) with the URL contained in the certificate. If the two match, then the user knows he or she has connected to the right Web site, and that a wrongdoer isn't trying to collect their credit card number through a phony Web site.

Another security mechanism familiar to Web users is passwords. The manner in which the Hypertext Transport Protocol (HTTP) handles passwords is called Digest authentication. Digest only sends an encoded password over the Internet, which helps protect the password in the absence of a mechanism like TLS. Even with TLS, if a Web request goes through a Web proxy, users still might not want that proxy to know their passwords.

The designers of SIP saw how well TLS and Digest work in the Web space, and decided to incorporate these mechanisms into SIP. For example, when a SIP user logs onto to a SIP server and

SIP may very well enable VoIP services that are more secure than what users can expect from the PSTN.

becomes available for calls, the manner in which they register and prove their identity is via Digest. For example, if someone were registering at a SIP service at 'neustar.biz' under the name 'john.doe,' he would send a SIP registration request to 'neustar.biz' with the password for 'john.doe' in a Digest header. In order to be sure that he was actually connecting to 'neustar.biz', and not another server pretending to be 'neustar.biz', he would use TLS and verify that the certificate passed by the server matched 'neustar.biz.'

But SIP security can't stop there. In terms of its architecture, and the way SIP messages travel around the Internet, SIP has quite a bit in common with e-mail. Most importantly, SIP messages travel from one endpoint to another endpoint, just like e-mail messages, and they tend to pass through one or more servers on the way. While someone can use a password to identify himself or herself to his local SIP server, sharing a password with all people who might be included in a future SIP session is unrealistic. There may be plenty of people he or she wants to call, with whom he or she has no previous association.

One of the biggest challenges facing e-mail security is the way one knows who sent an e-mail message. There is a "From" header that indicates to the recipient the identity of the sender. Unfortunately, users can usually put whatever identity they want in the "From" header. Spammers leverage this tactic constantly to "spoof" mail from various sources; clever spammers even try to spoof mail from people the recipient may know, increasing the likelihood that the recipient will open and read it. The SIP "From" header works the same way as the e-mail "From" header. That is, endpoints populate it. So, the question becomes: how can one ever truly know who is trying to make contact?

The answer is that the security of TLS and Digest can be leveraged to provide an assurance of the identity of the caller. In SIP, this assurance is reflected by the presence of the Identity header. When a SIP call is placed, the calling endpoint connects to a local server and provides a password with Digest — just as it does when it logs on. This server then checks the "From" header field and compares it to the password to make sure the user is authorized to claim the identity listed in the "From" header. If this is actually the case, an Identity header is added to the request, and signed with the certificate that the server uses for TLS. In other words, the identity header says that the calling domain (e.g., 'neustar.biz') vouches that this user (e.g., 'john.doe') authenticated himself properly and that the "From" header field is accurate. Who better than 'neustar.biz' to tell the recipient that a given endpoint is 'john.doe@neustar.biz'? This Identity header stays with the SIP message as it makes its way through proxy servers all the way to the endpoint. Any of these recipients, intermediaries or endpoints can look at the Identity header and see who the calling domain vouches for in the "From" header.

Identity doesn't just vouch for the "From" header; it also contains a signature over the body of a SIP message. That secures SDP, and lets any recipient make sure that no one has tampered with SDP. If SDP is secure, then it can contain keys that allow RTP to be encrypted, which prevents eavesdropping. The result is a chain of trust starting with SIP that cascades all the way down to the session media, letting callers know to whom they are talking and making sure that their conversation is private.

Identity itself does not innately prevent unwanted calls or spam, but it does prevent impersonation, which is a critical enabler for both spam and unwanted calls. Without an assurance of identity, spammers can claim to be anyone — and thus, there is no way for endpoints to create policy that can block particular senders.

Furthermore, identity provides accountability. The Identity header tells one what domain vouches for a caller, so that domain can be contacted (or blacklisted, as necessary, in cases of abuse). This general idea has applicability far beyond SIP; there is even an initiative for e-mail called Domain Keys/Identified Mail (DKIM) that also uses a domain-based signature to assurance the identity of an e-mail sender.

Through mechanisms like TLS, Digest, and Identity, SIP may very well enable VoIP services that are more secure than what users can expect from the PSTN — and just as easy to use. All a user needs to get started is a service and a password, just as e-mail and the Web operate today. The true advantage of a standard approach like SIP is that these services, including security features, will be interoperable across the disparate providers and devices using VoIP on the Internet. If the marketplace brings these standards into their offerings, consumers will rarely have to give VoIP security a second thought. IT

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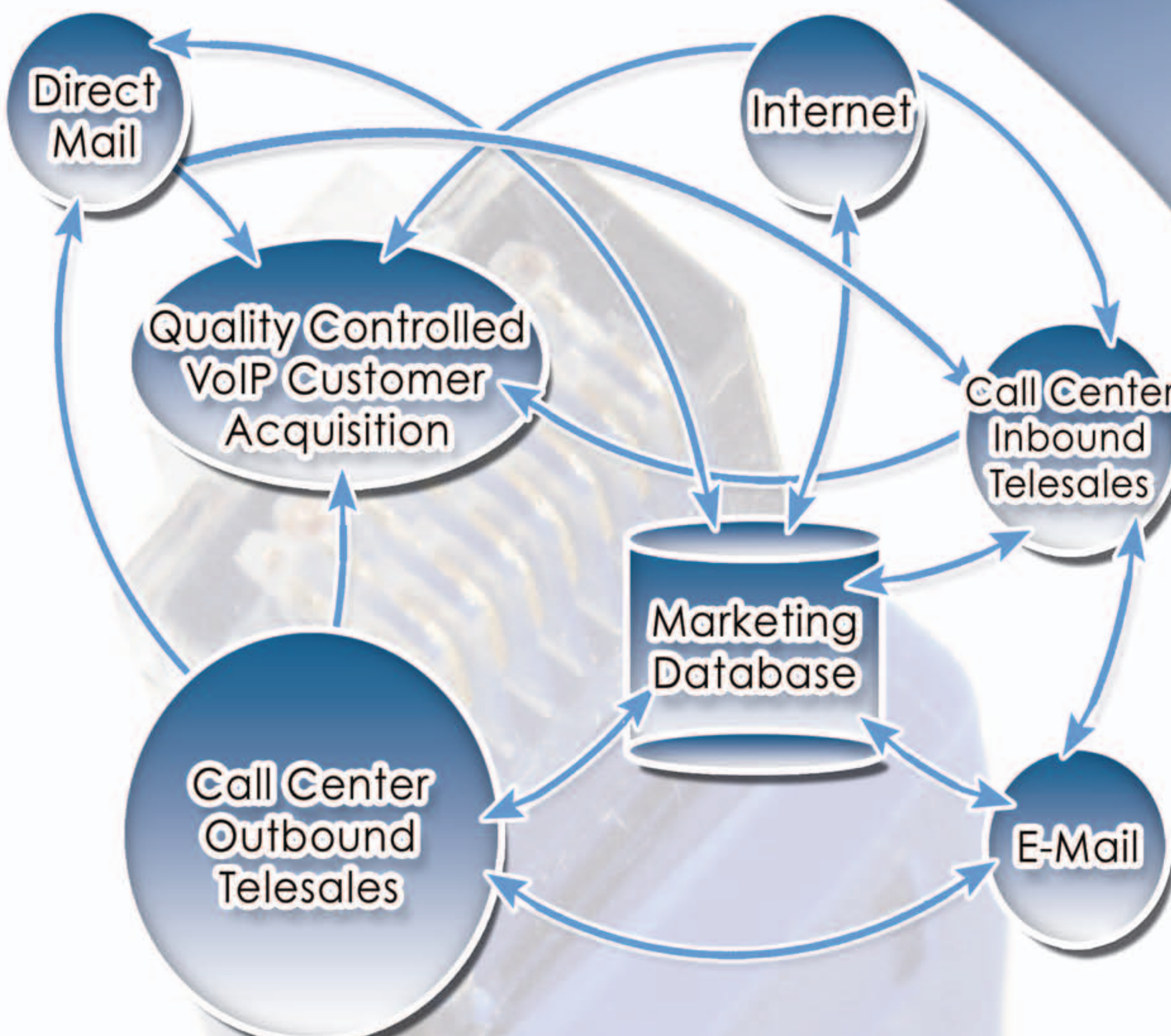

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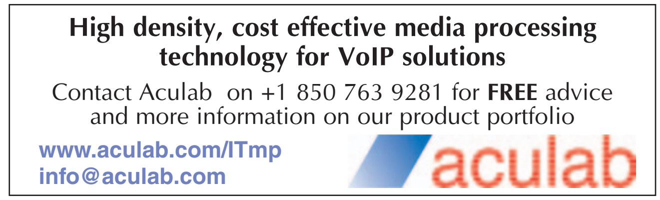

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VoIP MARKETPLACE

CEO Spotlight

Robert Mimeault
CEO
Versatel Networks

In the CEO Spotlight section in *Internet Telephony*® magazine, we recognize the outstanding work performed by exemplary companies. Each month we bring you the opinions of the heads of companies leading the Internet telephony industry now and helping to shape the future of the industry. This month, we spoke with Robert Mimeault, CEO of Versatel Networks. (news - alert)

GG: What is Versatel's Mission?

RM: Our mission is very clear, to be the leading supplier of value added services platforms that offer personalized voice services for legacy and next generation networks. We enable carriers to offer differentiated and personalized telephony services to their customers. With the decline in long-distance margins and local services, providers need the tools to create unique offerings that people will pay for. In addition, most service providers can't afford to throw out their legacy equipment and replace it with a pure next-generation architecture. Our products extend legacy equipment and give it new life, providing an excellent migration strategy for carriers of all sizes.

As an example, any traditional TDM (define - news - alert) carrier can take our equipment, install it adjacent to their legacy equipment and use VoIP to terminate their long-distance traffic. This alone is a compelling reason to use our technology, but that's only a start. Without adding any new hardware, our customers can offer a unique service like CallPods to their customers. CallPods is a party line for a group of people. You can set up the service from your computer by providing a list of phone numbers. Maybe they are your children's cell phones, maybe it's the list of people in a volunteer fire department. When you call the number, every phone on the list rings, and when they are answered they are placed into a conference. Imagine how easy it will be to find your family when they have scattered about the shopping mall or for a Sunday night family call. CallPods is just one example. Service providers of tomorrow can develop more of these value added services easily on our platform, creating differentiation that will ensure their success.

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

RM: To drive the implementation of specialized hardware for signaling, media manipulation, and call control into software, which enables rapid prototyping and deployment of new services on a one-to-one basis using industry standard open components. Nearly all telephony services require some sort of media processing, such as voice mixing, prompt playing, or DTMF (define - news - alert) digit detection. Our products are optimized to provide cost-effective hardware support for these sorts of applications. Our ability to deliver these applications provisioned to a single customer makes new service delivery cost-effective. As more applications are written for our platform, the value proposition for our customers will become even stronger. This positions us well to be at the heart of what drives the next generation telecom market: new revenue streams from new service offerings.

> Providers need the tools to create unique offerings that people will pay for.

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?

RM: We think that there is a strong correlation between the size of an organization and its ability to innovate. Our market is a perfect example of this — most of the innovation has come from the smaller service providers and vendors. This innovation fuels the growth and opportunity we are enjoying. If, through market consolidation, smaller players start to disappear, I'm afraid that the current pace of innovation will suffer. This can't help but reduce momentum in the market. As an example, directly from our market, look at the pace of innovation before and after telecom deregulation. After deregulation, many smaller companies entered the market, radically increasing the amount of innovation.

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

RM: We are focusing in two areas right now: scalability and applications. Our customers have real concerns regarding both ends of scalability. They need a cost effective, yet full featured, starter system. For many, this is a new business area, and they have a reasonable caution surrounding large capital purchases. We have an excellent starter system, but we think we can provide an even lower cost option. Once these customers begin to grow, they need a system that will scale upwards cost effectively. We already handle four thousand subscribers in a single chassis (and you can chain them together). We're going to support tens of thousands of subscribers in a single chassis, giving our customers plenty of room to grow. We think this was a major issue in computer telephony integration in the nineties. Developers with Dialogic cards created incredible applications, but couldn't scale them for true carrier deployments.

> The essence of the Internet is democratization. IP telephony won't be any different.

GG: Describe you view of the future of the IP Telephony industry.

RM: Our feeling is that the essence of the Internet is democratization. IP telephony won't be any different. IP Telephony allows even the smallest company to offer something unique to the marketplace. That's the future of IP Telephony. IT

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