Why Is the FCC Protecting Consumers?

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EDITOR'S NOTE

Lots of Action in a "Slow" News Month

by
Greg Galitzine

August is so often a "slow news month" by mainstream news standards. But based upon the craziness in the IP Communications industry, this past month has proven to be anything but slow. Typically, August may feature one overarching story, like, say, an invasion of a neighboring country, or a major environmental event. In early August, the news of Ottawa-based Eicon Networks buying Intel's Media and Signaling business might just have met both of those criteria. But the rest of the month was just as brisk with SIP-related announcements, and certainly the pipeline looks to be filled for this autumn.

For example, Bandwidth.com (news - alert) launched its SIP Trunking VoIP (define - news - alert) product. Bandwidth.com's SIP Trunking product is designed to eliminate the need for additional hardware to convert TDM traffic to VoIP, providing a simple end-to-end SIP VoIP connection to their network of carrier gateways. The product's unique selling point is that businesses can oversubscribe each VoIP trunk, enabling them to purchase trunks only for the number of concurrent calls they support, rather than buying one for each individual employee. Bandwidth.com has chosen to entice customers by offering the product with unlimited incoming and local calls and very competitive long-distance rates, for under $.02 cents per minute.

CommuniGate Systems (news - alert) announced a new customer: Versature has chosen the CommuniGate Pro platform as the foundation for its managed IP PBX offering for small- to medium-sized businesses (SMBs). Versature, an established service provider of business-class VoIP, initially developed a premise-based system, which proved to be complex and time-consuming. The system also prevented Versature from standardizing its offering to generate value-added revenue for enhanced services. Versature was in the market for a scalable solution to enable the move to a scalable, fully hosted, and virtualized PBX model.

After evaluating multiple solutions, Versature selected CommuniGate Pro, which enables them to offer services beyond SIP-based VoIP, such as e-mail, instant messaging, and collaboration packages. Deploying the CommuniGate product has enabled Versature to offer their customers voicemail and auto-attendant, self-service, Web mail access to voicemail, multiparty conference calling, and secure instant messaging.

Victor, NY-based REDCOM (news - alert) introduced the REDCOM TRANSip call controller at their annual government users meeting. TRANSip provides flexible voice and network capabilities across both VoIP and legacy TDM networks and provides network operators and users with integrated call management, gateways, and controller services and multiple other communications-related features.

AVST (news - alert) made news in August with new SIP integrations for its CallXpress unified communications platform. The company announced they were expanding CallXpress' compatibility by focusing on SIP, with plans to provide additional SIP integrations with other communications products, including Microsoft's Live Communications Server.

Oh and one last piece of news from the "slow news month:" We've recently hired Richard "Zippy" Grigonis as Executive Editor of *SIP Magazine*. Richard will be taking over the reins beginning with the November issue of this bi-monthly publication, thus freeing me up to devote more time and energy to our Web resource TMCnet. It's been an honor and a privilege serving as the founding editor of this magazine. I am confident that under Richard's leadership, *SIP magazine* will continue to deliver news and analysis of this exciting market space, and will continue to be viewed as the #1 resource serving the SIP community.

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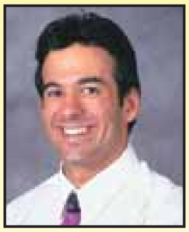
CONTENTS

feature articles

PUBLISHER'S OUTLOOK

Phone Companies Get an FCC Spanking



by

Rich Tehrani

A number of consumer advocacy groups and others have been trying to encourage the FCC and government to put a stop to the consolidation of service providers that is taking place. They argue that the old AT&T is being rebuilt and the telecom industry is heading in the wrong direction.

In response, the FCC has publicly commented that there is more competition that ever from cable companies, as well as wireless and satellite providers.

But the FCC's comments about an abundance of competition seem to have been put to the test recently as Verizon has announced a new broadband surcharge. According to a recent article in *The Wall Street Journal*, Verizon began e-mailing its roughly six million high-speed Internet subscribers, informing them they would no longer be charged the Universal Service Fund fee — which was $1.25 or $2.83 a month, depending on speed of service. But it went on to say that it was instituting a "supplier surcharge" of $1.20 or $2.70 a month beginning August 26.

Many point to this new fee as evidence that there is a lack of real broadband competition in the market and they further argue the timing of this increase — just as the FCC declared broadband subscribers no longer had to pay into the Universal Service Fund, is especially sneaky.

It is for this reason that the FCC decided to send a "letter of inquiry" to Verizon in order to ascertain the reasoning behind the surcharge. The letter is the first step toward a formal investigation. Verizon said it decided to impose the new fee on all Internet subscribers because of increased costs of providing service to customers who only buy high-speed Internet. Verizon strongly disputed the idea that it hadn't been upfront with consumers about the new charge and said its timing was designed to minimize the impact on consumers who won't see their bills change significantly.

From my vantage point, it is bizarre that the FCC needs to step in and protect consumers.

The following statement perhaps explains the point best:

"The commission takes its obligation to protect consumers very seriously," said FCC spokesman David Fiske. "Consumers must be provided with clear and non-misleading information so they may accurately access the services for which they are being charged and the costs associated with those services."

Let's put aside the fact that the term non-misleading is something most people have never heard before — I imagine when most people encounter the term, they get the same confused feeling they get when trying to decipher the fees and taxes on their current phone bills.

The FCC's actions are admirable. The question is why are they necessary?

If there truly is competition, why would Verizon and SBC — who just recently dropped their $2.97 "regulatory cost recover fee" — not be allowed to charge fees to their heart's content? After all Federal Express and other shippers routinely add fuel surcharges to packages their customer's ship.

If all of the above-mentioned companies are publicly traded and they have an obligation to shareholders to increase profit, what business is it of the FCC to step in and police the increased fees being charged? After all, isn't it obvious that Skype, cable companies, Vonage, and other VoIP companies are wreaking havoc on the bottom line of Verizon, AT&T, and others?

This is the exact irony the public should be thinking about, as on the one hand, the FCC is allowing large-scale mergers to decrease broadband competition, but on the other hand is trying to make sure the companies in the new telecom landscape don't take advantage of their newfound power.

One has to wonder if the problem is to allow all the mergers in the first place. It would seem that as fewer and fewer real alternatives exist in the market we will need the FCC to step and protect consumers ever more often. It is worth pointing out that the above scenario is exactly what consumer groups have been afraid of. Perhaps it is worth taking a breath and listening to what these groups and others are saying before we proceed to eliminate more broadband competitors from the market.

AVST Adds New SIP Integrations to CallXpress

Applied Voice & Speech Technologies (AVST) (news - alert) announced it has added new SIP integrations to its flagship CallXpress unified communications platform. As part of its ongoing goal to support interoperability and offer the most integrations in the market, AVST's CallXpress will support SIP integration to the Nortel Communication Server 1000 and Avaya Communications Manager PBX products. AVST plans plans to provide additional SIP integrations with leading communication products, including the Microsoft Live Communications Server.

"Because SIP is a popular protocol for IP telephony, video, instant messaging, and presence engines, it is the ideal protocol to be used by unified communications solutions like CallXpress," said AVST's Vice President of Product Management, Tom Minifie. "For companies that are interested in migrating to VoIP, either now or in the future, CallXpress can be easily integrated via SIP to IP telephony systems, and at the same time, continue to support legacy PBX systems."

In addition, AVST now provides a CallXpress Turnkey option for all of its servers, which enables customers to purchase the T2000, I4000, I6000 and the new V2000 models pre-loaded with CallXpress software. This option was developed to make deployment easy for system integrators and enable resellers to quickly fulfill orders by delivering the server with all the software pre-installed.

"This is just another example of AVST's commitment to providing best-of-breed technology built to ensure ease of use, simple installation and administration, and low cost of ownership," added Minifie. "Using our CallXpress Turnkey option, final onsite installation is reduced to site specific configuration tasks. System integrators save valuable time by avoiding hardware integration, operating system configuration, and basic software installation tasks."

http://www.avst.com

Telrex Extends SIP IP PBX Support

Telrex, (news - alert) developer of VoIP call recording and monitoring software for SMBs using IP PBXs or hosted PBX services, announced support for IP PBXs from Pingtel, Fonality, TalkSwitch, and Switchvox.

Pingtel, Fonality, TalkSwitch, and Switchvox are all innovative providers of SIP-based IP PBXs. Support for these IP PBXs strengthens CallRex's leadership position in the VoIP call recording market for small and medium businesses.

With hundreds of deployments on SIP-based IP PBXs and softswitches from multiple manufacturers, CallRex supports more SIP-based IP PBXs than any other call recording solution. In addition to Pingtel, Fonality, TalkSwitch, and Switchvox, CallRex supports SIP-based IP PBXs from Cisco, Asterisk, Avaya, Nortel, Mitel, ShoreTel, 3Com, NEC, Siemens, Mitel, Zultys, and Vertical, and SIP-based softswitches from BroadSoft, Sylantro, and Tekelec.

SIP is growing rapidly as an IP PBX standard and makes it easy for IP PBX resellers to quickly sell and deploy VoIP in businesses of all sizes, enabling small and medium businesses in particular to access a wide variety of previously unaffordable applications such as call recording.

"Telrex is pleased to bring affordable call recording to those businesses that have invested in SIP IP PBXs, including those from Pingtel, Fonality, TalkSwitch, and Switchvox," said Robert Kapela, president of Telrex. "For many of these businesses, it is the promise of applications like call recording that makes standards-based IP PBXs so attractive."

http://www.telrex.com

TelePlus Group to Deploy SIP Platform

By Cindy Waxer

Thanks to a two-year strategic partnership with Digitrad France, TelePlus Group (news - alert) is gearing up to offer VoIP services using a turnkey SIP network - the PsipTN (Public SIP Telephony Network). The deployment of PsipTN, set to go live later this year, will allow TelePlus to offer customers a telecommunications and billing system, using the Vocalyz product for on-the-road travel and at home using the VoIP enabled 1TC product.

"By allowing customers to continue making and receiving long distance calls and to access the TelePlus proprietary 'in-language' interpreter/concierge services beyond the travel experience, we are not only creating an efficient form of seamless communications, but also a continuous revenue stream by unifying access components (GSM/VoIP/WiFi) and consolidated billing applications," said Jim Gibson, vice president of business development for TelePlus Group.

PsipTN is developed by TelTel, (news - alert) a developer of VoIP SVNO platforms. Together with PsipTN comes a full-featured VoIP softphone client, which allows the customer to seamlessly continue using TelePlus communication services well beyond the travel experience.

The process is universal and allows users to make low cost calls, continue receiving calls from their local number, manage personal buddy lists and access the TelePlus international services and more from anywhere around the world. PsipTN is integrated with the existing Digitrad (news - alert) GSM billing system, Stand4U, which is a Web-based IVR platform that allows for the efficient immediacy and accuracy of account provisioning, data access and billing transfers.

http://www.teleplusgroup.com
http://www.digitrad.com
http://www.teltel.com

JPS Rolls Out Analog Radio Adapter for SIP Networks

By Laura Stotler

Raytheon JPS Communications (news - alert) has announced a new Analog Radio Adapter, the ARA-1. The new solution is comparable to an analog telephone adapter and enables a standard radio to operate on a SIP network.

The ARA-1 extends the coverage and capabilities of existing SIP PBXs by enabling the interface of land mobile radios to the system. Radios are connected through the ARA-1 and are assigned unique extensions, so they can easily be dialed via any IP phone, softphone, or voice device associated with the SIP PBX. A number of services are also available, including call logging, call recording, and call forwarding. The SIP PBX can also enable video conferencing, document sharing, and text messaging between compatible devices.

The new solution brings radio networks into the SIP arena and also brings SIP-based communications to areas not serviced or reachable using a SIP network. For instance, an ARA-1 could be used to extend SIP communications into tunnels, across bodies of water or through rugged terrain.

"The ARA-1 is a perfect marriage between land mobile radios and IP-based networks," said Mike Cox, vice president of engineering for JPS Communications. "It combines a supremely capable radio interface to the standards-based open SIP protocol that is rapidly becoming the acknowledged pathway to the convergence of voice, data, and video."

http://www.jps.com

Verizon Business Unveils Advanced IP-Based Services

Verizon (quote - news - alert) Business introduced two new Internet protocol-based capabilities for its Contact Center Services and VoIP portfolio to help businesses enhance customer-service operations and leverage the benefits of VoIP.

The new offerings are: IP Tollfree Service; IP IVR, an interactive voice-response system for contact center services; and new IP Trunking options, all featuring interoperability with Avaya enterprise communications software.

"We continue to advance and extend our VoIP offerings to meet customer needs where it matters most - at the heart of their business operations," said Tom Roche, vice president for network voice and data services for Verizon Business. "IP Tollfree and IP IVR will help redefine how companies implement contact centers in the future. As businesses increasingly make the move to IP, Verizon Business continues to outpace the industry, delivering one of the most complete suites of VoIP services available today."

Verizon IP Tollfree routes incoming toll-free calls over IP to enable greater efficiency and support multiple-contact media, such as phone calls, e-mail, or IM from around the globe. The service enables contact center agents to transfer calls using capabilities inherent to SIP, which enables real-time communication on the Internet. Since IP Tollfree is a network-based service, companies benefit from lower total cost of ownership because they do not have to own and operate costly gateway equipment.

Verizon IP IVR provides call processing in a pure IP environment over a carrier-grade, global network infrastructure, enabling customers to benefit from network efficiencies such as voice compression and dynamic bandwidth allocation. The service offers administrators an extensive selection of call-routing and processing features and terminates incoming calls to both TDM and IP endpoints, allowing customers to adopt IP technology at their own pace.

Verizon Business has designed and certified IP Tollfree and IP IVR to be compatible with Avaya Communication Manager with SIP enablement services (SES) 3.1 software and other leading SIP-enabled endpoints.

"Our research shows that there is a huge pent up demand for SIP-based call delivery, and Verizon Business is meeting that demand," said Robin Goad, lead analyst, contact centers, Datamonitor. "Businesses can improve customer service by working with suppliers that offer robust but simple-to-use-service offerings and whose portfolios reflect a clear IP strategy."

http://www.verizonbusiness.com

BandTel Partners with Quintum

By Cindy Waxer

BandTel (news - alert) and Quintum Technologies (news - alert) have formed a relationship that will allow customers to acquire a Tenor VoIP switch or gateway with the BandTel Services. BandTel is a provider of next-generation VoIP termination to the PSTN for high-volume telecom users such as call centers, enterprise users, teleconferencing companies, and IVR users. The company's VoIP network solutions are now fully interoperable with Tenor VoIP MultiPath switches and gateways.

Enterprise users will be able to access the combination of BandTel's VoIP network to deliver call center and enterprise users a cost-effective, fault-tolerant voice architecture and Quintum's Tenor VoIP MultiPath solutions.

According to Chris Dunk, president and CEO of BandTel, "Quintum's Tenor VoIP MultiPath solutions are the perfect complement to BandTel's network because both integrate so easily into an existing network, even if legacy devices are present and both guarantee dependable switching and routing. In addition, by mutually supporting the SIP standard, we are working together to advance the future of VoIP technology."

Because Quintum's Tenor switches support industry standards like SIP, they are wholly interoperable with BandTel's SIP Softswitch technology. Customers deploying Quintum"s MultiPath VoIP systems with BandTel's VoIP services will have the unique ability to connect call systems via a trunk-to-trunk transfer and cost-effectively terminate calls to and from any location in the world. This allows users to instantly unite their global businesses with multiple users in various locations.

In addition to Quintum, BandTel has partnered with several other names as well, including Sphere, Digium, AudioCodes, and VegaStream.

http://www.bandtel.com
http://www.quintum.com

Sipera IPCS 310 Supports More VoIP Environments

By Johanne Torres

VoIP (define - news - alert) security technology provider Sipera Systems (news - alert) of Richardson, Texas, announced expansion of its IP communications security solutions to more environments. The company's IPCS 310 will now offer protection of VoIP and IM to more environments, allowing enterprises, especially in the financial services and healthcare industries, to secure legacy systems and additional IP communications applications against attacks, misuse, and service abuse.

The IPCS 310 system monitors VoIP traffic, detects anomalies in call traffic patterns, and identifies threats to protect end-user devices and network infrastructures for enterprises with up to 1000 users.

The new upgrades features include multiple protocol support to protect VoIP assets in mixed protocol environments; border control to enable the deployment of SIP trunks and voice extranets, as well as extend VoIP infrastructure to mobile users; expanded IM protection for SPIM and IM spoofing to protect internal and external IM communications; IM logging functionality; and additional media protection to prevent threats that may be launched through the media channel after a VoIP connection has been established.

"Giving customers choices and flexibility is particularly important as new technology is integrated into existing enterprises," said Sipera's president and CEO, Seshu Madhavapeddy. "For enterprises of all sizes and configurations, comprehensive network-level and end-user security is a prerequisite to deploying today's growing suite of collaborative VoIP, IM, and other IP communications applications."

Madhavapeddy added: "With these new features, the Sipera IPCS 310 allows enterprises to efficiently take advantage of today's real-time, IP communication applications without compromising the network's security requirements."

http://www.sipera.com

CommuniGate Systems Rolls Out Upgrades for VoIP and SIP

By Laura Stotler

CommuniGate Systems (news - alert) has announced its "Trade In and Trade Up" program to enable carriers to upgrade to new value-added services including VoIP, SIP/XMPP secure IM and hosted PBX. The program will enable carriers to trade in legacy e-mail software, saving on support and maintenance services. They may then "trade up" to replace antiquated messaging systems and offer a full suite of IP communications solutions.

CommuniGate's initiative will enables service providers to smoothly migrate their communications services to an IMS-ready platform. It is targeted toward providers that have customers unable to access SIP-based IP communications networks because their email solutions use legacy technology.

The providers will be offered a full CommuniGate Pro license, replacing their existing ongoing support costs. A simple migration will then enable providers to offer a number of enhanced IP communications services. Carriers may choose which features of the CommuniGate Pro suite they wish to deploy. The modules offered include SIP Proxy, Session Border Controller (SBC), IP PBX, clustered voice through a SIP Farm, audio conferencing, XMPP support and voice mail.

"The feedback we are receiving is tremendous as the world is now adopting SIP based Communications. Mobility and productivity will rise and traditional closed and location locked access will die," said Jon Doyle, vice president of business development for CommuniGate Systems.

Doyle continued: "We will witness a fundamental change in the communications landscape over the next five years just as we saw in the early 90s with email becoming the communication standard medium for business," added Doyle. "Holding users to a location with a phone number, or charging them for roaming to other locations will soon be replaced with the mobility and portability of VoIP - where one address finds a person anywhere, anytime."

http://www.communigate.com

Pretty Good Security for SIP Communications

By Erik Linask

BorderWare Technologies (news - alert) and Pretty Good Privacy (PGP) founder Phil Zimmermann announced that BorderWare is poised to become the first commercial licensee of Zfone, a secure VoIP media encryption software solution also created by Zimmermann. By integrating integrates Zfone with BorderWare's SIPassure VoIP Security Gateway, this agreement brings a new level of security and ease of use to VoIP systems.

According to BorderWare, SIPassure is the industry's first VoIP security gateway that takes VoIP application security to a new level by combining the best features of an enterprise firewall, application layer gateway, and a Session Border Controller.

SIPassure is designed to secure all SIP-based applications, including VoIP services, video conferencing, and other messaging applications. SIPassure ensures that organizations using SIP-based communications remain harbored from impairment and service disruption from internal and external attacks, interference spam, and other threats.

The integration of Zfone media encryption with SIPassure ensures that BorderWare's customers have the security to ward off Spam threats, DoS attacks, eavesdropping, spying, and wiretapping,--all without limiting the performance and convenience customers have come to expect from their e-mail.

http://www.borderware.com

Patton's SmartNode VoIP CPE Certified by Thomson

By Erik Linask

Patton-Inalp Networks (news - alert) has announced that its SmartNode brand of VoIP routers has been certified by Thomson for interoperability with Thomson's Cirpack brand softswitches. The certification assures carrier customers of a smooth migration as they move from multiple legacy infrastructures to fully converged IP-based voice and data services.

Patton has actively sought softswitch interoperability for both its SmartNode and SmartLink families of VoIP and triple play platforms via its own Interoperability Program, which acknowledges third-party softswitches that passed tests proving they work with Patton's equipment - primarily in the area of live carrier deployments.

For use with Thompson's equipment, SmartNode gateways connect business and residential equipments to the next-gen VoIP features and services enabled by Cirpack (news - alert) using SIP. As such, SmartNode products are designed with the future of the communications industry in mind, capable of supporting new advanced products as they emerge. In addition to being SIP-compliant, SmartNode supports H.323, T.38 fax relay, fax bypass, modem bypass, Voice over VPN, AES/DES-IPSec voice encryption, and DownStream QoS.

"Patton brings a lot to the table," said Fabien Maisl, Marketing Director of Thomson's Network Intelligence Solutions Cirpack. "Their extensive portfolio is a great fit with our worldwide customer base. Carriers always prefer a single-source supplier, and SmartNode pretty well covers it. We are very pleased to welcome Patton as a certified technology partner.

http://www.patton.com

Interactive Intelligence Releases Upgraded Unified Communications Software

Interactive Intelligence, (news - alert) a global developer of business communications software, has released an upgraded version of its unified communications software, Communite.

Communite is a voice mail replacement system offering standards-based unified messaging, interactive voice response, and real-time communications services for all types of organizations, including large, distributed enterprises and higher education institutions.

The latest Communite release, version 2.4, was designed to further increase organizational productivity while minimizing infrastructure requirements with enhanced functionality for speech-enabled mobile applications, simplified creation of customized voice mail menus, flexible personalization options for presence management, and an alternate message store for unified messaging applications.

"These latest enhancements to Communite make it an even more compelling unified communications system for organizations with mobile employees, those looking to replace aging voice mail systems, and companies that must comply with increasingly stringent archive messaging requirements," said Yankee Group senior analyst, Ken Landoline. "With IP telephony driving faster adoption of unified communications solutions, Communite's built-in SIP support gives it an added competitive advantage."

New in Communite 2.4 is an enhanced Interaction Mobile Office(TM) application, which builds on its existing speech-enabled auto-attendant by adding a speech-enabled menu for message retrieval, status changes, and company directory access. A new telephone user interface enables organizations to use XML for easy emulation of existing voice mail menus, thus reducing user training requirements when replacing voice mail systems.

Other enhanced features such as new customizable status codes that can be applied to different messaging rules - for instance, "if out of office, forward calls to mobile phone" - give organizations unsurpassed flexibility.

http://www.inin.com

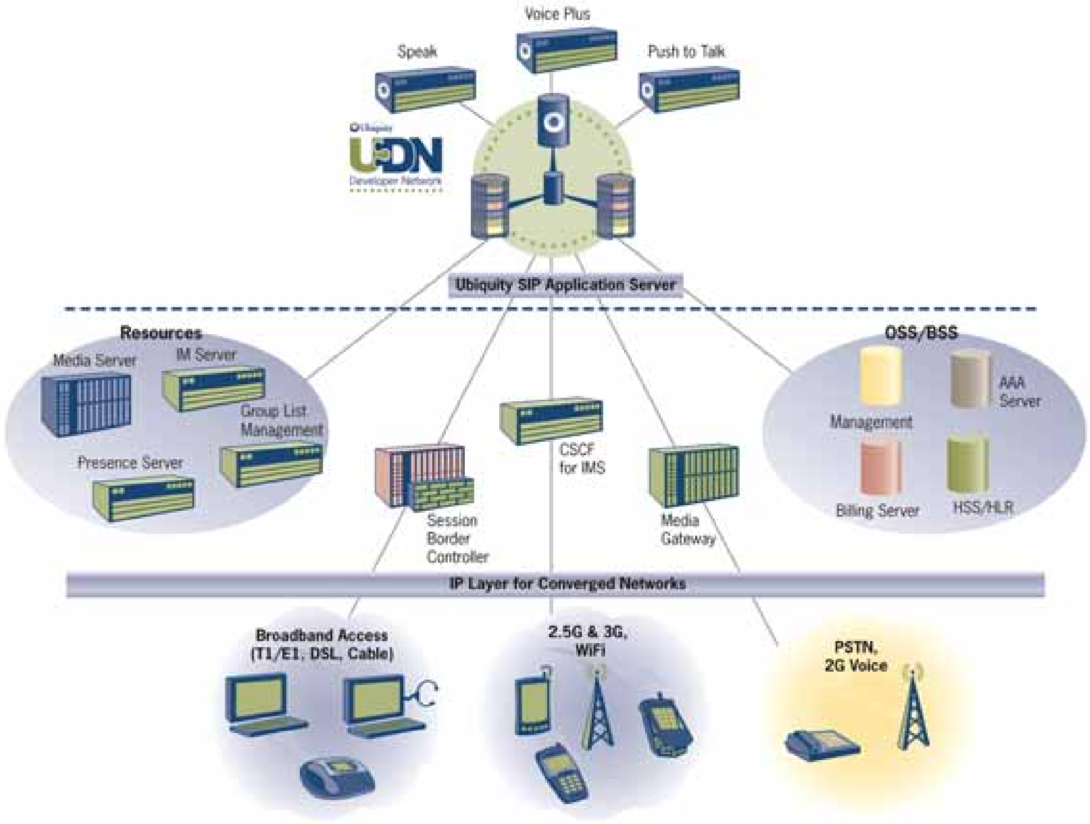
Huawei Deploys Ubiquity Software at Telefonica for IP Conferencing

Ubiquity Software (news - alert) has announced a joint win with Huawei Technologies, whereby Ubiquity and Huawei have deployed Ubiquity's IP conferencing solution at Telefónica Argentina, the first service provider to launch IP conferencing services using Speak from the Huawei/ Ubiquity relationship.

Ubiquity's Speak IP conferencing solution is a scalable, carrier-class, IP conferencing application that enables Conferencing Service Providers to offer hosted audio and Web conferencing services. Speak leverages SIP-based VoIP and Ubiquity's carrier-grade SIP Application Server (SIP A/S) architecture. This easy-to-use, browser-based solution offers a complete conferencing application feature set as well as a web portal for scheduling, initiating, managing and terminating
multi-party conferences.

"With years of experience deploying next-generation network solutions in over 20 countries, Huawei understands carriers' needs, and is experiencing a strong demand for SIP-based applications," said Mr. James Yuan, Vice President, Strategy & Marketing for Application and Software Product Line for Huawei Technologies. "Ubiquity's Speak IP conferencing solution allows us to market a world-class solution to address our customers' needs."

http://www.ubiquity.com
http://www.huawei.com



ON THE EDGE

Deep Inside an Emerging Telco

Building the Next National SMB Communications Network with SIP

by
Erik Lagerway

Question: **What does it take to build the nation's first real SMB telco?**

Answer: **Many people, a lot of money, and a cattle prod.**

There are so many ways to skin the cat these days and everyone seems to have their own opinion. When Shift Networks ultimately decided to move away from proprietary technology and make the move to open standards, they relied upon many for guidance and vision. I would hope that my contribution in this regard has had and continues to have a positive impact.

In the early days of planning it was an uphill battle, there were many questions and not a great deal of time to make the right choice. In order to build a network that would truly scale and be commercially reliable and viable is not something you just throw your hat at. There were many unanswered questions, like; "How will we interconnect with our carriers? Well, with SIP of course! "What kind of impact will SIP Peering have on client provisioning and account management? The quick answer: a positive impact! In the end, I feel we did make the right decisions and we are now on final approach as our new network nears customer trial readiness.

Something that is very important to the SMB users is resiliency and reliability. We want to know that our phones will always be on. Just like any other utility. When we walk into our offices and hit the light switch, the power should come on. This new network has some great geographical redundancy built into the core and at the premise there will be IP link failover. I am not sure that I have seen anything like it.

We want to know that our phones will always be on.

The edge of the network demands intelligence. Choosing the right partner for handsets, softphones, remote call control clients, and SDKs is very important. Thankfully I have some experience in this regard. Our SIP handsets are feature-rich and should not fall short of any SMB user's expectations. Softphones are in the mix and third-party call control applications that allow you to control your handset from anywhere you can get Internet access will also be supported by the network. E-mail integration, Click-to-Dial contacts, Presence, and Instant Messaging are now all on the map, thanks to open standards.

We have the endpoints, now how do they get the info they need to function in the network. Automatic provisioning is the key. Many VoIP (define - news - alert) providers underestimate the time it takes to provision an IP handset. Let's say it takes just a couple of minutes to provision a handset, multiply that times the number of users and we are now looking at a huge issue! Combine that with inventory stocking and the whole thing starts looking quite ominous. Our goal here is to provide a no-touch provisioning system so that the phones are drop-shipped directly to client sites. The client plugs the phone into the local network and voila! Everything just works. As we get closer to our production network roll, we should see this new provisioning methodology really take form.

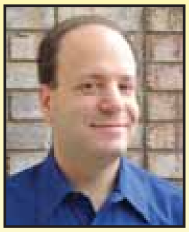
What about sales and marketing? How will we bring these new services to market? Well, thankfully we have a very competent group in charge of those efforts. As long as I meet my objectives and we deliver on our network milestones we can expect a smooth ride during our trials and production network rollout. From what I have seen of the marketing plan, the message is fairly straightforward and I think all those SMBs out there will certainly get it.

I also think the ILECs and their ilk have a wake-up call coming. The alarm clock will be playing The Greg Kihn Band's *Breakup Song*, and yes, that volume knob — it goes to eleven!

Erik Lagerway is chief technology officer at Shift Networks. (news - alert) For more information, please visit http://www.shiftnetworks.com.

SPEAKING SIP

Vishing and SPIT



by
JD Rosenberg

Technical folks seem to get a lot of satisfaction from coming up with cute names for complicated things. Computer security experts, in particular, have dreamed up a whole host of nice-sounding names for all the nasty threats that show up on the Internet. First, there was the term virus, which is actually an acronym (vital information resources under siege). Then, we had malware, spyware, worms, Trojan horses, and adware. With new threats came new terms, and the industry started to hear about spam, botnets, spamdexing, link bombing, phishing, and pharming.

Recently, phishing attacks have been launched using VoIP instead of e-mail — and voice phishing, or vishing, was born. The idea is simple. The attacker uses a VoIP version of a war dialer (a piece of software that dials a large number of phone numbers in sequence, like the one used in the film "WarGames") in an attempt to connect to a person. If someone answers, a recorded greeting says that a credit card needs to be validated by the bank. The person is asked to enter the credit card number and security code to validate the card. Since many people are used to phone-based mechanisms for validating credit cards, they enter the numbers and — voilà! — the attacker has harvested their credit card number.

Vishing is very similar to VoIP spam, sometimes called SPIT, for "spam over Internet telephony." So far we have seen very little of it. Is it something to be worried about?

Absolutely.

The economics of e-mail is one of the primary reasons for the widespread usage of spam. The cost of sending a piece of e-mail is so incredibly small that even if just a tiny fraction of recipients buy an advertised product, it's still worth doing. VoIP has the potential to drive the cost of voice calling down sufficiently low that the economics start to look like they do for e-mail. In addition, it is much easier to build applications like war dialers for VoIP than it is for the regular telephone network. No hardware is required, just a PC with IP access. This makes it a viable service for botnets to provide, which can further reduce the cost of making a VoIP spam call.

The damage that VoIP spam can do is far worse than e-mail. Imagine a world where your phone rings every two minutes, and the caller is a recording offering you some kind of unwanted product. Getting an e-mail every two minutes is bad, but a phone call is far, far more intrusive and time-consuming to deal with.

Unfortunately, it's not as easy to stop VoIP spam as it is e-mail spam. The vast majority of anti-spam technology is done by content analysis. The tools examine the message itself, looking for certain keywords in order to classify the mail as spam or not. Unfortunately, with VoIP, the content is very difficult to analyze automatically (it's streaming voice), and the content arrives only after the call has been answered. Content analysis isn't the only technique used to screen e-mail for spam, though. There are black and white lists, reputation systems, and payments at risk, among other measures. These techniques are as equally applicable to VoIP as they are to e-mail.

White lists are particularly interesting. The basic premise is that you maintain a list of senders (or callers in the case of VoIP) that you trust. Calls from those people get through immediately. You can manually add people to your white list (a buddy list is a great source!), or they can be added automatically through some other technique that tries to validate the sender. White lists, unfortunately, rely heavily on a way to securely determine the sender of a message. With e-mail, it is extremely easy to forge the identity of the sender. For white lists to work with VoIP, systems must be put into place that allow for highly reliable caller identities to be carried between providers. The IETF has completed specification of a mechanism that carries out strong cryptographic identity verification. This mechanism, sometimes called SIP Identity, will be crucial for the success of VoIP anti-spam techniques. The IETF is also producing an informational document that summarizes the problem of VoIP spam and the solution space.

So, next time your phone rings — be wary. It might be an attempt to sell you medication for erectile disfunction through VoIP. Shall we call this — Via-shing?

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PRESENCE ENABLED

Context is King

By Joe Hildebrand

“Content is King” declared Bill Gates in 1996. “Content is where I expect much of the real money will be made on the Internet, just as it was in broadcasting [where] the long-term winners were those who used the medium to deliver information and entertainment,” he continued.

Ten years later, an information explosion has turned that declarative on its ear. Content is sexy and certainly continues to be the public persona of most media. However, in the digital realm, Context is what now matters… just consider the threat that Google poses to Microsoft: Is that about Content with a capital C or about helping people find the right content within a specific Context? Today, nearly all information is accessible as a series of ones and zeros, and in less time than it will take you to read this article, you can publish directly to the entire world for zero incremental cost and vice-versa. We have shrunk the Earth and stuffed it with enough content that were it not for the balancing force of context it might collapse on itself like a neutron star.

As Gates predicted a decade ago, the democratization of content has created a wealth of opportunities for individuals and organizations of every size producing everything from the thoroughly useful to the thoroughly tasteless and everything in between. This is a double-edged sword, on one side offering valuable and readily accessible content where it wasn’t available before, on the other creating a glut of content, much of which competes constantly for our attention, with the high-value stuff often given the same weight and priority as the unwelcome distractions.

The democratization of content has created a wealth of opportunities.

A customer at a large brokerage firm frames the issue of context around the concept of information half-life, which is to say that the value of information delivered to decision makers at the point of a transaction degrades measurably with every fraction of a second. In this firm, information comes fast and furiously from hundreds of source points. To be useful, traders need to be presented only with the information that matters to them and only at the point it is most actionable. Within this firm’s data feeds, content comprises only the ‘what’ of information, whereas context offers the more dynamic variables of ‘who, when, where, how, and why’ that information is consumed.

Within many information-intensive real-time environments, “presence,” which begins with the availability of people, devices, or applications, is being extended with secure mechanisms for subscribing to the changes in geographic location, expertise, capability, and anything else that can affect a system or individual’s ability and or willingness to consume and react to the data. Within the mechanisms used for publishing and subscribing to real-time changes in the extended presence of these endpoints lies the key to enabling contextually aware data networks that deliver exactly the right information to exactly the right person, place, or thing, at exactly the moment the data is most valuable and or consumable.

Identity-Based Routing

For example, the Capital Wireless Information Network, CapWIN (http://www.capwin.org), is an information clearinghouse coordinating the emergency response efforts of over 40 federal, state, and local jurisdictions in the Washington, D.C. area. At its generic core, CapWIN is simply a router for streaming XML that turns disjointed raw data into contextual information, pulling together individuals into ad hoc groups based on their capacity and expertise in responding to immediate crises. For CapWIN, identity-based routing is of paramount importance as it represents the ability of applications, services, and contextually aware data to follow people — appropriately — as they shift between locations and roles, capabilities, and availability, moods, and immediate interests. For instance, were a chemical spill to occur within a CapWIN jurisdiction, an ad hoc response team comprised of the most local first responders and chemical experts from anywhere within the CapWIN domain is created. This group convenes in a secure chat room that is aware of its own context and able to alert and invite the right participants — based on their many presence variables — instantaneously. Using identity-based routing, experts and emergency crews are located wherever they are, regardless of the device or network they are using. Additionally, because their

identity and role can be factored into the CapWIN system in real time, incident rooms automatically assign the right command and control structure, i.e., determine who is in charge and responsible for specific decisions within this dynamically created group.

In telecommunications, we are finding that as service providers, device makers, application developers, corporate information managers, and advertisers seek to generate and route content to people that makes sense based on their current time and physical and mental space, it is becoming critical that systems and services employ advanced presence architectures that adopt the concept of identity-based routing. When the point becomes to better ensure that the interruptions that matter (phone calls, IMs, SMS, etc.) are filtered from the same types of interruptions that don't, it is extremely relevant that content accounts for much more than simply who I am and even what I want. Content must have context, presence, and understand identity-based routing in order to dynamically factor not just my name and delivery address, but also my role within an organization and as a private citizen, my inherent capabilities and interests within my many roles, and my given availability to consume specific content based on these variables. When all of this is weighed we can arrive at the relative importance of a given communications in the context of what, where, and when I am.

While reaching this end goal is no simple task, it is getting easier and the reference examples exist. As we have seen in some very high stakes environments, the technology and logic are already deployed and high value contextual information is available to decision makers in these organizations. Moreover, it is becoming apparent — again looking at the position Google now holds — that the companies that can layer context onto the information we need and want will in the end exert exceptional influence over the user experience and likely the creators and even distributors of the actual content we consume. All hail King Context!

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SPECIAL FOCUS

SBCs — Now and Forever: A Rebuttal

In a July 2006 *SIP Magazine* article about session border controllers (SBCs), the author suggested that the end of SBCs is near. I respectfully dispute this notion because SBCs solve real problems and are essential to the delivery of secure, high-quality services. Session border control is not a temporary function or network element as these problems are not going away. SBCs serve to overcome numerous challenges: securing the network, extending service reach, ensuring service quality, protecting profitability, and complying with regulations.

Furthermore, the SBC is transparent to the end-user — they don't know about it, nor should they. However, the capabilities that SBCs bring to our network allow us to deliver high quality and reliable services, qualities that we can use to differentiate our services and incorporate in our service level agreements (SLAs). These capabilities translate into a quality user experience and a happy customer!

NuVox is a rapidly growing, facilities-based integrated communications provider delivering a full array of communications services in 16 states across the Midwest and Southeast. NuVox employs SBCs in strategic locations throughout our network for three main purposes today:

Securing our network

Security is a challenge that will never go away and session border controllers are optimized to solve that problem. With dynamic and static access control lists and signaling rate limiting, we reduce the threat of external denial of service (DoS) and other malicious attacks. It all starts with our SBC being hardened against these attacks, including dedicated, high-performance hardware. The threats stop at the SBC without impacting its performance or the performance of legitimate existing or new calls. This, in turn, protects our important infrastructure—the gear that our customers rely on for service and we rely on for revenue—from being affected by hackers and attackers. Additionally, we can fully hide NuVox's core infrastructure topology from customers and peering partners as our SBC is a back-to-back user agent (B2BUA) and acts as a double NAT for layer 3 and layer 5 addresses. Our softswitches and application servers are hidden and not addressable, making the important revenue producing equipment more difficult to attack.

Security is a challenge that will never go away and session border controllers are optimized to solve that problem.

Ensuring service quality and availability

With SBCs at the access border, we can ensure that the NuVox VoIP network is up and available to service our customers, thus meeting our SLAs relative to availability and call quality. There are a host of SBC features that deliver on service assurance. Call admission control plays a role here by throttling the allowable number of sessions to any given device in the core of the network. That ensures that existing calls are not impacted by unusual busy periods and events. We can continue to serve customers while giving a busy signal to new attempts until the call patterns return to a serviceable level. Also, to ensure proper QoS levels are provided for all calls, the SBCs properly mark incoming packets for specific treatment by the upstream routers and switches as well as report on the actual call quality characteristics, which helps with reporting and problem resolution.

Extending service reach with hosted NAT traversal

NATs and firewalls at the customer premises break the hosted VoIP services model as they are security devices and, behaving correctly, they don't allow inbound and seemingly unwanted communication requests to enter the company network. SBCs deliver a solution for traversing that security boundary that is the easiest to implement and least disruptive to the customer's network and

security model. It requires no provisioning of subscribers on our end and no change to the equipment at the customer premises. Additionally, the IP address of the SBC in our network can be used as a trusted source for security policies.

While there are many initiatives that hope to solve the NAT/firewall traversal problem, SBCs solve that problem today. Waiting for the NAT traversal standards to finalize — and products to appear — would be tantamount to not offering services for the past several years and still waiting an indeterminate time for them. The standards, STUN and TURN, are just another approach to hosted NAT traversal — not necessarily better. SBCs will adopt the standards-based NAT traversal techniques when they are finalized — meaning the SBC will be a STUN/TURN server — allowing the service provider to choose the NAT traversal method they prefer. Since I want to offer services today, I'm not waiting, and I'll take my SBC to solve today's real challenges for reaching my customers.

In the future, we plan to use our SBCs for additional applications, including:

- **CALEA support** for our VoIP traffic, allowing us to offer our services that will be required to comply with lawful intercept regulations. With the upcoming May 2007 deadline for compliance, we already have the equipment in place for intercepting call content and call data.

- **Media transcoding**, where necessary, to increase the number of networks we can connect to, as well as efficiently engineer our network for bandwidth usage. Transcoding would also allow us to interconnect with other carriers that use other codecs in their network without requiring change at the hand-off point. Transcoding at the IP border in the SBC is much more effective than using our media gateways. SBCs are optimized for IP-IP borders as media gateways add additional latency as there are two transcoding steps in the process: IP to TDM and TDM back to IP.

- **SIP to H.323 Interworking** in order to easily connect to legacy VoIP networks and the majority of IP PBXs for trunking services. While SIP is by far the dominant protocol for new deployments and new services, there are numerous networks that still use H.323 and that will not change in the near future. Furthermore, our SBCs help resolve the inconsistencies between various vendors' implementation of SIP. The SBCs deployed with this interworking function at the access border means that NuVox does not need to make any changes to neither our internal VoIP core nor the networks of our customers. Signaling protocol interworking is another example of extending service reach.

There are no meaningful alternatives that can deliver all this functionality — security, call admission control, interworking, transcoding, QoS, lawful intercept — in one manageable, highly available, and scalable platform. Of course, not all SBCs are created equal. The rash of recent failures and cheap exits has more to do with products that did not meet today's challenges than it did with SBC functions not being necessary today or in the future.

In that earlier article about SBCs, the author cited a problem of registration floods after a network failure due to the fact that SBCs cache endpoint registrations. This issue can easily be overcome by using an SBC that can gracefully protect itself from these floods and by configuring the admission control policies on the SBC so that the network resources are not overloaded.

While SBCs are not free, we view SBCs as an investment in our customer base. SBCs deliver necessary functionality and solve fundamental problems in order deliver services that generate revenues. These IP communication services are the future of any service provider. Without SBCs, these services would not be possible. Or rather, we could offer services, but not truly deliver service differentiators like quality, security, and availability.

The bottom line is that SBCs solve many real problems associated with VoIP service offerings. The rich functionality of SBCs — they are used for much more than hosted NAT traversal — delivers the ability to competitively differentiate around a secure, high-quality, user experience. If you're a service provider looking to make money on VoIP and other session-oriented IP communication services, then a SBC is a wise investment.

Art Nichols is the VoIP Architect of NuVox Communications. (news - alert) For more information visit the company online at http://www.nuvox.com.

How to Develop and Implement an Effective IP Strategy

By Gustavo Garcia

Over the next two years, it's expected that 82 percent of contact centers will be running on IP telephony infrastructures. If you're in the process of implementing, considering implementation, or want to make sure you're getting the most out of your existing investment, how do you go about it?

In this article we will discuss the key features of IP technology, how it should be approached by businesses of all sizes, and the vast range of measurable benefits IP can generate in the contact center.

The advent of IP technology and IP telephony — or Voice over IP (VoIP) — allows contact centers and businesses around the world to make global communications instantaneous, simple and cost effective. To implement IP you need to know exactly what it is, what it can do for your business and how to make it work for you. Knowing how standards such as Session Initiation Protocol (SIP) can integrate into your business to address specific needs and requirements, and how your new IP infrastructure can be fired into action can ensure improved productivity and flexibility.

IP technology has reached a stage of maturity. It is being rolled out to increasing numbers of businesses, with many more beginning, or at least considering, deployment of it. A recent survey showed that 43 percent of businesses are planning, deploying, or have deployed IP telephony in the contact center. Within two years, 82 percent of contact centers expect to be running IP telephony infrastructures.

Despite this surge of usage, there is still some uncertainty surrounding IP technology — how it works, its full range of functionality and the benefits that it can bring to your business. Because businesses today are now talking in terms of "how?" and not "if,"' it is important to have a clear view on these issues.

How IP Can Help

IP implementations enable businesses, particularly contact centers, to enhance the way they operate telephony environments. Adding voice interactions to the data network enables voice to be managed in the same way as Web or e-mail channels. This means that priorities, business rules, and automated resolution can all be more easily integrated into the voice-based area of the business.

A telephony environment based on the increasingly significant SIP architecture can enable contact centers to separate voice application software from the underlying hardware infrastructure, which brings greater flexibility to the management of the business, because there are few limitations to further application deployment. SIP connects all IP endpoints and will enable applications to continue to revolutionize the way we communicate. Research shows SIP becoming more popular, with 30 percent of businesses deploying an IP infrastructure, and 38 percent of those investigating it, choosing SIP-based environments — compared to 14 percent who have already implemented.

Infrastructure is very important to the success of an IP deployment. Proprietary solutions have been popular in the past, but there is a growing movement toward Open IP because it offers greater flexibility. At a time when technology developments are ongoing, implementing a fully future-proof solution has to be a key concern.

What Can IP Telephony Do For You?

In today's mature IP environment, there are many categories and definitions of platforms available for the implementation of IP telephony. An open-standards IP platform enables companies to integrate leading edge applications, legacy applications and hardware into the new infrastructure, improving its functionality while limiting the level of capital needed. This helps retain important data with little risk of loss, and caps new infrastructure costs through PBX side savings. Because of the open nature of the platform, additional applications can be integrated into the infrastructure as requirements grow and evolve.

SIP is standards-based and media independent, and comes with a generic interface, making all applications easily interoperable. By bypassing unnecessary middleware, SIP creates a comprehensive yet transparent interface for all the applications in the enterprise. This helps agents be more effective, improves the service for callers and can also help reduce cost by up to 25 percent.

On the business side, a SIP-based solution can deliver considerable PBX savings. Because it integrates all applications and communication channels into the data network, there is no need for computer telephony integration (CTI). Thus, legacy PBXs are no longer essential. This power of open standards supports non-proprietary hardware and software for queuing, routing, and managing customer interactions. It enables these changes to be made while routing strategies, supervisory roles and key metrics will all remain essentially the same. This means that both the IT and the business benefits of the implementation can be considered.

The biggest benefits of an IP deployment tend to be longer term — the ability to mix time-division multiplexing (TDM) and IP telephony infrastructure components, as well as the flexibility to buy from multiple vendors to develop an infrastructure (pre-empting future expenditure). Gaining centralized, consolidated operations management improves business operations, while virtualization provides a single point of enterprise-wide control and routing to improve customer service. Overall, the extent to which a company is able to embrace IP technology will dictate the long-term benefits received.

How To Start An IP Implementation

An IP implementation should be designed so that it suits the business requirements. IP platforms can either be pure or hybrid, and can be added to both proprietary contact center environments as well as those with open architectures. The size and scale of the deployment should be customized to match the size of the business. Keep the limitations in mind — for example, a proprietary environment will require less up-front cost, but limits future options for growth and development.

Develop a migration strategy that is in line with your business goals — both present and future — and your infrastructure needs. By doing this, and taking a phased approach, your business will not compromise existing investments, and the implementation will not disrupt existing processes.

What Does My Business Need?

Matching your IP deployment to your business goals is critical. Large businesses should consider whether or not to consolidate multiple sites into a single, virtual contact center, as IP can reduce the volume of the contact center infrastructure by restructuring it into a single site. This reduces the maintenance and upkeep because the infrastructure is all in one place and it enhances the manageability of the enterprise. A SIP-based environment ensures there are no disparate systems, so all of the information needed is assimilated and readily accessible, and any action that needs to be taken can be affected with one change, rather than via a coordinated effort to bring every site up to speed.

- *Are you a small business owner looking to increase the scope of your services by bringing in remote and home-based workers?* For a small central site, employees in other parts of the country that have critical business value can be easily integrated into operations at a fraction of the cost of TDM solutions. Remote agents simply need an Internet connection and an IP headset. Because the entire infrastructure is hosted in a central site and applications can be transparently linked through SIP, one only needs to download the application to a desktop.

- *Are you trying to incorporate local branch operations into your service?* There has been a long-standing conflict between the low cost of a contact center and the customer-friendly local branch — but an IP platform enables you to compromise. Local branches can be integrated into the contact center network easily and cheaply. This way, customers get through to the same people every time — building a valuable relationship — and the investment needed is much more cost-effective than running a branch office itself.

- *Do you need to outsource some of your contact center operations, but are reluctant to relinquish complete control of the day-to-day running of your operation?* Through an IP platform, you can see the same management information as the supervisors who are on location at the contact center — and at the same time. This means that changes to staffing levels can be done instantly, and you are constantly up-to-date with the performance of the agents.

- *Do you want to make your operation more resistant to disaster?* Open standards-based IP platforms future-proof your investment by allowing for multi-vendor application integration as the business advances. Any new technology advances that are made can be added without compromising existing solutions. In addition, the ability to move calls around more easily means that any local problems with electricity or media inputs can be dealt with more easily by re-directing traffic to other areas of the business. You can also integrate your back-office into your data network, ensuring that the whole process is in sync and manageable from a single point. The reduction in infrastructure required on the PBX will also translate into savings in terms of both expenditure and indirectly through reduced maintenance, as the SIP server will enable a single access point to monitor all the elements of the infrastructure.

Making It Work

The balance to strike is a complex one among flexibility, reliability, functionality, and cost. Different architectures will create a slightly different balance of these factors.

For a low-cost IP deployment that doesn't need extensive functionality, a hybrid environment based on a proprietary system is the best option. It does not require the same level of financial or technical investment as an open platform and can provide immediate IP functionality and benefits.

For a solution to solve today's business issues and prepare the company for tomorrow's as well, an open standards-based platform is the best option. Open IP will adapt to new challenges more easily as it allows integration off-the-shelf applications from multiple vendors whenever necessary.

The simple way to make IP deployment work for your business involves thorough planning, rather than the deployment itself. Which elements of the enterprise need to be integrated into the platform — the remote and home-based workers, off-site experts and the back-office? These may be vital parts of the business, but ensure that they become more integrated as a result of the IP deployment, otherwise the enterprise will remain disjointed and the investment is fully maximized. It's possible to create a SIP-based environment where all applications — contact center, enterprise and back-office — are integrated into the IP network, with every employee connected and the whole enterprise operating from the same central network.

MANAGEMENT TOOLS

Once deployment is completed, it is important to have an ongoing management structure. An IP platform offers a great deal more management information on the entire enterprise and increases the level of control that you can exert. Using this in an efficient way to manage the unified operation will further enhance the effectiveness of the deployment.

Workforce management solutions help coordinate agent resources across multiple shifts, departments, and sites more effectively. Reactive intra-day scheduling reflects changes in any part of the business, which helps enhance agent productivity across the enterprise.

Skills-based routing can be integrated into workforce management as can other value-add solutions (such as skills management software) to increase productivity and service levels. Routing calls to agents with the right skill sets improves service to the customer, reduces the pressure on the network from internal transfers, and improves resolution rates for agents. An IP platform lets you capitalize on remote experts integrated into the network. These solutions enhance the portfolio of management tools available, and enable you to coordinate the wider environment created by IP telephony.

Measuring The Benefits

An IP deployment will provide initial cost savings. Gradually, network and toll charges will visibly decrease, and the effectiveness of customer service will experience a positive spike.

IP migration is a strategic move, rather than just a cost-cutting exercise, so the real benefits arrive in due time. A business choosing to integrate video capabilities in three years time, can do so in the same way as voice or Web channels with an Open IP infrastructure. The Open IP platform enables video to be merged into the data network so that it becomes simply another aspect of the infrastructure. Without the option to do this, the choice is limited to "rip and replace" or "miss out" somewhere down the line.

An Open IP platform running from a SIP server also helps integrate new business applications and features to support new initiatives. If you're restricted by your infrastructure whenever you need to launch a new campaign, your investment is failing you. Make your infrastructure investment work for you.

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SIP & ENUM — Foundations of Peering

By Eli Katz

VoIP peering is becoming an increasingly important part of VoIP providers' strategies and operations, both from an engineering and a marketing perspective. Peering enables service providers to reduce off-net PSTN termination costs, expand network coverage for their services, and, more than ever, empower customers to experience higher quality broadband codecs and video on off-net calls.

As broadband and the IP communications applications running on it increase in popularity, the Session Initiation Protocol (SIP) and Electronic Number Mapping (ENUM) standards are providing the foundations for the inevitable industry shift to multi-lateral VoIP peering.

What is VoIP Peering?

Peering is a term that was established decades ago in the IP bandwidth space. ISPs either "peer" with each other, (i.e., freely exchange traffic), or buy transit from each other, where there is an imbalance of data flowing one way or another. For example if 100,000 users of a consumer broadband service are downloading MPEG video, large files and other content from a 10,000 Web sites around the world, then the consumer broadband service will have to buy transit IP bandwidth to accommodate.

In telecom it is minutes rather than data that flow between service providers. Classic telecom interconnects, whether achieved via VoIP or TDM (define - news - alert) technologies, are similar to IP transit, in that the source of the calls pay per minute to deliver traffic to the destination. Peering in the telecom environment, a free exchange of minutes, will only work if call minutes flowing in either direction is broadly balanced — and this is something that rarely happens in a traditional telecom environment due to the marketplace's reliance upon transit carriers (especially for overseas calls).

Surprisingly, minutes between residential users on different networks are likely to be balanced, even when the size of the networks differs significantly. Imagine one VoIP service provider with one million consumers and another with only ten thousand consumers; as often as one of the million wants to call one of the ten thousand, one of the ten thousand will want to call one of the million. This natural tendency towards balanced traffic exists only when calls flow directly between the originating and terminating service providers, without the involvement of a transit carrier. This end-to-end delivery is referred to as peering and is complicated by one significant issue; you can only deliver a call directly to the terminating service provider if you know that a number belongs to them. Thus peering must be enabled by a "location" or "discovery" function that tells an originating operator whether a dialed number belongs to another VoIP service provider.

Modeling peering relationships on traditional telecoms interconnects implies that each bilateral peering relationship would require:

- Sharing of subscriber phone numbers.
- Independent commercial contract.
- Interconnection and interoperability testing process.
- Ongoing maintenance as new services and equipment are brought on line.

It is easy to see how scaling and security issues abound when undertaking multiple interconnects using this approach.

A more rational and scalable approach is multilateral peering. Each service provider has a relationship with only one entity, the central peering facility, which delivers the following improvements:

- Privacy, by acting as custodian of the phone numbers.
- Easier Interoperability, by providing signaling interoperability and insulating providers from technical changes implemented by others.
- Security, by defining standards and policing behavior.
- Simplicity, by defining a set of commercial and operational standards to which all members must adhere.

Counter-intuitively, signaling interoperability is often the most complex of these challenges from a technical perspective, even when everyone is using SIP.

SIP

Created from the ground up to be user friendly and easily extensible, SIP is becoming established as the protocol of choice as VoIP technologies are deployed. Large incumbents (such as British Telecom in the UK) and 3GPP IP Multimedia Subsystem (IMS) have selected variants of SIP as their primary signaling protocol. Already the most popular VoIP signaling protocol available, SIP will probably be at the core of any peering relationship in the future — so why is interoperability a challenge?

SIP is becoming established as the protocol of choice.

While basic at its core, SIP has grown tremendously since its inception in the late 90's. Contrary to its stated purpose of utmost simplicity, SIP today comprises myriad RFCs and drafts, and as such, interoperability between different flavors and implementations cannot be taken for granted. Each individual VoIP service provider may have a different SIP implementation, driven by different equipment vendors with their own preferences.

For example, in SIP there are several different fields in which to pass on Caller ID information. With each VoIP service provider using a different method, a multilateral peering service must identify and normalize these methods to ensure feature transparency, in this case by delivering the Caller ID to the terminating operator.

In addition to being used by endpoints to initiate communications, SIP is also used by network elements used to relay these requests. The network elements are tasked with relaying the signaling as well as fulfilling the "location" function by "finding" the remote party and relaying this information to the caller. In SIP this is accomplished by using a method called SIP redirect.

There are however, other protocols that can be used for this "location" function — ENUM being the primary one.

ENUM

ENUM, (news - alert) with its roots in DNS, enables originating networks to query a registry in order to discover routing instructions for e.164 numbers that are associated with IP devices on other networks.

As most VoIP networks are primarily connected to the PSTN for service, each end point is identified by a standard e.164 number. The ENUM registry allows the service provider to find the IP address corresponding to these E.164 number masks and route the call without utilizing PSTN termination.

Among the advantages over SIP in supporting the "location" function, is that ENUM is specialized for traditional phone numbers (whereas SIP can just as easily support e-mail addresses) and also provides support for augmenting routing information such as number porting and other similar functions.

Conclusion

ENUM and SIP provide key features and functions that facilitate peering, but also leave many options and decisions for each network architect. Due to the rapid growth of the market participants and the many different technical implementations, a neutral multi-lateral peering service is the logical path for the future.

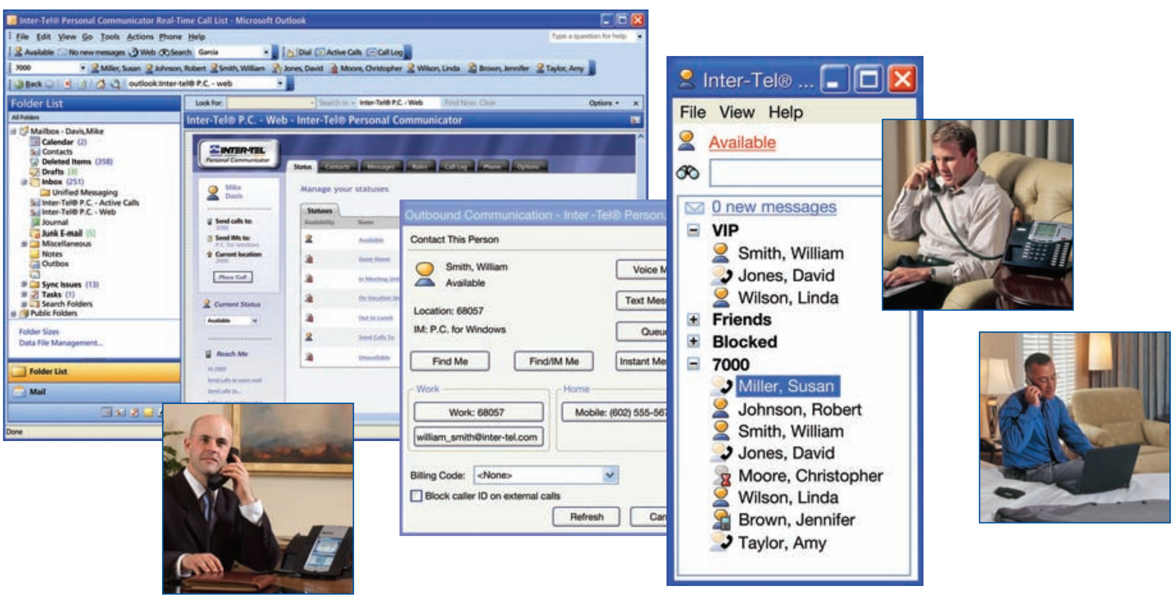
Eli Katz is chief executive officer of XConnect. (news - alert) For more information, please visit the company online at http://www.xconnect.com.

ADVERTISING INDEX

Optimize Business

Yours, *Not Theirs*

The Inter-Tel® 7000 Redefines Business Communications.



Replacing old technology with new solutions allows you to operate more efficiently and drive revenue growth. But will your investment continue to deliver, outpacing the threat of obsolescence? A communications solution from Inter-Tel will. We offer the next-generation communications technology today that others can only promise for tomorrow.

The Inter-Tel 7000 multimedia communications system embraces the future of business communications. Built on open-ended, SIP-standards-based architecture, it allows you to leverage the business tools that make the most sense for your business, when you need them and as they become available. It's freedom of choice — both today and tomorrow — from the proprietary technology other vendors would prefer you remain locked into.

Why compromise? The Inter-Tel 7000 delivers the solutions you require ***today***, the options you expect for ***tomorrow*** and the investment protection you insist upon ***always***. The choice is clear — with Inter-Tel your business is ***Always ON***.

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Always **ON**