TMC Labs Review: pbxnsip

www.sipmag.com Volume 2 • No.3 May 2007

TMC SIP MAGAZINE™

The Authority On Session Initiation Protocol

SIP & Open Source

QoS in SIP-based Networks

SIP & ENUM

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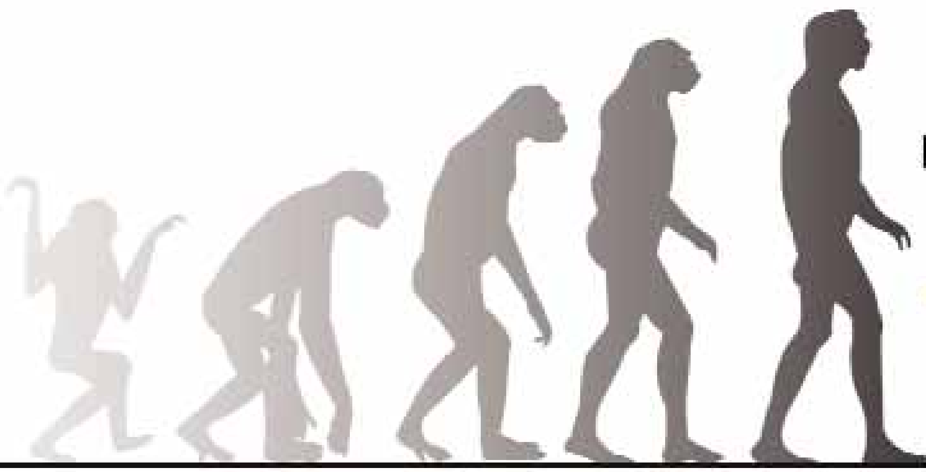

In July SIP MAGAZINE becomes Unified COMMUNICATIONS


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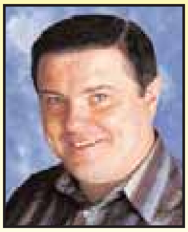
EDITOR'S NOTE

Goodbye and Hello



by
Richard "Zippy"
Grigonis

SIP magazine is about to undergo a major transformation.

As Rich Tehrani and TMC announced last month, _SIP_ is becoming _Unified Communications_ magazine. The premiere issue will appear in July. The new magazine will contain a section exclusively dedicated to SIP, and the same SIP columnists will be found there.

Why the change? Now that the "gee-whiz" aspects of the SIP protocol (and of IP for that matter) are slowly disappearing as it becomes a mainstream technology incorporated into many different IP Communications devices, it's time to focus on what we can actually *do* with the plethora of new and exciting IP and SIP-based products and services now appearing at a quickening pace. This is not surprising if one examines the way a new technology is developed and undergoes successfully adoption by vendors and customers.

For example, in 1878 the public would have eagerly read a magazine entitled *Light Bulbs* ("Every shape, size and color bulb is reviewed by our capable staff… Gas lighting, arc lights and incandescent bulbs compared… Tesla's new fluorescent bulb… Plus 10 tips for selecting the right kind of light bulb for your home!") In 1992, a magazine on the new and astounding technology of CD-ROMs would have been a hit — I believe there were even some expos, conferences and seminars held at the time. In the late 1990s I was even asked by an investment banker and some friends of mine to be part of a project to publish and oversized, glossy magazine called… *IP!* It would have been quite an ego boost, but the magazine would have lasted only until the telecom bubble popped a few years later. (As it happens, the project went nowhere.)

All of this demonstrates that, while individual, new technologies are the subject of intense scrutiny, these things-in-themselves (to borrow a term from Immanuel Kant) become pedestrian after a while, because they simply serve as the building blocks for products and applications that actually do something. By the end of the 1990s, people were not interested in the embedded operating system used by voice modems — they wanted to know what they could do with them to save money and be more productive. (As it turned, out, of course, voice modems themselves would be eclipsed by newer technology and an entirely new cycle began.)

The mythic unified messaging technology of the 1990s was the progenitor of modern Unified Communications (UC). UC is much more than simply listing voicemails, faxes and emails on a single interface, as was the case with its predecessor. Today it has more to do with presence, FMC (Fixed-Mobile Communications) and IMS (IP Multimedia Subsystem).

All of this just goes to show how TMC not only continues to stay on top of the world's rapidly growing and evolving technologies, but it also follows them to maturation and fruition in the everyday, real world.

We hope you'll enjoy our new magazine and website!

Richard Grigonis is Executive Editor of TMC's IP Communications Group.

SIP MAGAZINE™

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SIP MAGAZINE™

Volume 2/Number 3 May 2007

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feature articles

Employee healthcare costs out of control?

- **Group Telecom Equipment Purchasing Plans**
- **Employee Healthcare Insurance Plans**
- **Solution Reseller Owner Training**

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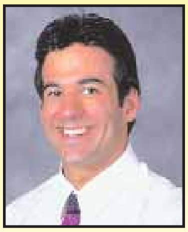
PUBLISHER'S OUTLOOK

IMS Collaboration: Tekelec/HP



by
Rich Tehrani

SIP, of course, is a major protocol that makes possible IMS (IP Multimedia Subsystem). IMS is something many service providers need but the complexity of implementing IP multimedia subsystem solutions has certainly been a cause for operators to deploy such solutions more slowly than some would have hoped. Beyond complexity, another factor which needs to be overcome is interoperability between disparate vendor's systems.

For this reason a number of organizations such as the IMS Forum have been focusing on plugfests to ensure IMS vendor A can interoperate seamlessly with vendor B and C.

The need for interoperability is not lost of companies like Tekelec and HP who began collaborating last year and more recently made an announcement that their respective platforms will work together.

Specifically the companies have aligned the roadmaps of the Tekelec TekCore Session Manager, a CSCF (Call State Control Function) platform and HP OpenCall home subscriber server or HSS.

The solution includes IMS core network infrastructure, service enablers, operational and business support system linkages and application service offerings that enable the delivery of subscriber-centric services across wireless, wireline and broadband networks.

TekCore provides CSCF capabilities to enable the control of next-generation multimedia traffic. TekCore also provides SIP signaling router functionality, allowing operators to inexpensively upgrade their next-generation networks (NGNs) to IMS — generating new revenue opportunities.

The IMS core network elements in the solution include the following:

- The CSCF and HSS mentioned previously.
- Service enablers such as the media resource function (MRF), presence server, electronic numbering (ENUM), group list management and voice call continuity (VCC) platforms
- Multimedia applications such as enhanced voice services, instant messaging (IM) and multimedia content sharing (e.g., "see what I see")
- Integration with back-office and legacy systems.

The companies explain that convergence cannot happen in a vacuum but instead needs a convergence framework which requires a good deal of planning. In addition they feel such a framework needs to address the following:

1) A unified signaling and control infrastructure that unites signaling and control procedures across multiple network types. Furthermore, this is a prerequisite to enable uniform access to applications and services across heterogeneous networks.

2) Providing a real-time unified view of the customer. This view may include the users communication's preferences, information regarding whether the subscriber is currently on a call, and a list of people engaged in a conference call with the subscriber. This real-time view of the customer is essential to enable subscribers to seamlessly access services and applications as they move across networks and device types. A consolidated awareness of the subscriber by the network is also necessary to enable the network to know how to deliver calls and further determine the willingness of the subscriber to accept the call.

3) The network needs to be able to deliver the necessary media or content to the subscriber across a variety of different access technologies and be capable of adapting the media to the access network and device.

The companies point out an instant cut over to an IMS-based network is often impractical and instead service providers need to consider migrating to IMS while still engaging their NGN and TDM networks.

Of course leveraging disparate networks is analogous to creating monstrous problems one may liken to Frankenstein.

For as a subscriber moves from network to network they likely experience different control procedures and applications. From a carrier perspective, similar, but different applications may need to be deployed in each environment creating complexity from a management and operations perspective.

Moreover each network type has its own control layer, its own applications, and its own subscriber data. To enable convergence this must change. However, in enabling convergence we must also realize that a greenfield approach, that moves us completely to IMS is often not a possibility given the investment that has been made in existing services, and constrained capital budgets.

According to the companies, the first step in moving towards convergence is to implement a unified control layer. Why? A unified control layer can facilitate access to applications and services regardless of the network type. The second step is to provide a unified view of subscriber data. These two steps will enable the development of converged applications and services.

The joint IMS solution is based on ATCA-based blade servers and service providers are trialing this solution at the moment. In addition, the companies have a multiplayer game demo running in the lab.

My take on this news is that it is good to see two major players in the service provider space coming together to make it easier for service providers to build next-generation networks and applications. One benefit of this collaboration is faster time-to-market for service providers worldwide.

A wise approach here is the realization that service providers are not going to have greenfield opportunities and as such a hybrid solution makes sense at the moment. Tekelec says they will be working with HP more closely in the future and furthermore they will work with other vendors more closely as well. This news is good for the market and likely signals a trend toward more vendor cooperation which is in the best interest of everyone involved and will help further the IMS market.

Be sure to attend the Communications Developer Conference (http://www.commdeveloper.com) next month in Santa Clara, CA where a number of service provider topics will be focused on such as IMS development and an Open Source Solutions Workshop for service providers sponsored by Pactolus (news - alert) (http://www.pactolus.com).

INDUSTRY NEWS

http://www.tmcnet.com/633.1

Sphere, Quintum Ally on Certified Remote Office Solution

Sphere Communications (news - alert) and Quintum Technologies (news - alert) announced the companies have certified the interoperability of Sphere's Sphericall IP PBX application with Quintum Technologies' VoIP gateways. Both companies believe the combination provides a powerful set of options for enterprises to ensure reliable communications across all office locations.

http://www.spherecom.com
http://www.quintum.com

http://www.tmcnet.com/634.1

Touchstone Technologies Announces Ultralite Series

Touchstone Technologies (news - alert) has introduced a new Ultralite Series of its WinSIP, Win323, and WinEyeQ call generation and monitoring and analysis products. This new series services an emerging need in the marketplace for high quality, portable, and cost-effective tools that can generate, monitor, and analyze true voice and video over IP traffic.

http://www.touchstone-inc.com

http://www.tmcnet.com/656.1

Aculab Launches ApplianX Product Line

Aculab (news - alert) announced a new line of appliances, different from any products previously supplied by the company. The new ApplianX (news - alert) range has been designed to enable solution providers to benefit from the latest technologies without having to incur development costs and integration time. With this in mind, ApplianX offers deployment-ready products based on Aculab's Prosody X IP media processing cards.

http://www.aculab.com
http://www.applianx.com

http://www.tmcnet.com/635.1

Verso to Acquire VoIP Gateway Supplier sentitO Networks

Verso Technologies (news - alert) has agreed to acquire privately held sentitO Networks, (news - alert) provider and distributor of VoIP gateway solutions for telecommunications service providers, in a stock transaction. Founded in 2000, sentitO provides the sentitO Open Network Xchange architecture, which enables innovative voice services by providing media and signaling conversion utilizing SIP technology.

http://www.verso.com
http://www.sentito.com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IBM Business Partner

http://www.tmcnet.com/636.1

Damaka Peer-to-Peer SIP Software Platform Launches DialIn and DialWorld Features

damaka, (news - alert) a fast-growing Peer-to-Peer SIP software company, launches DialIn and DialWorld services to complement its communication and collaboration product suite. With this launch, damaka offers a complete voice solution — PC to PC, PC to Phone (DialOut), Phone to PC (DialIn), and Phone to Phone (DialWorld) using its revolutionary Personal Softswitch.

http://www.damaka.com

http://www.tmcnet.com/637.1

Polycom Expands VoIP Desktop Line

Polycom (news - alert) now has two entry-level phones available, that are cost effective as well as feature rich — the SoundPoint IP 320 and 330. Both are built with full duplex speakerphones using the same Acoustic Clarity Technology that have put Polycom in the conference rooms at countless businesses. Both also sport an easy-to-read graphical LCD and integrated PoE support and typical enterprise class features.

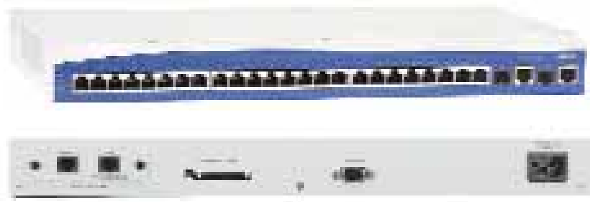
http://www.polycom.com

http://www.tmcnet.com/638.1

Amity Systems Intros Multimedia Conferencing and Broadcast Communication Platform

Amity Systems, (news - alert) a provider of video collaboration and broadcast solutions has introduced its latest platform to bring enterprise networks and service provider networks instantaneous collaboration via video, voice, and content collaboration, regardless of the number of users or location of users. The solution boasts quick and easy point-and-click set up from a buddy list, as well as presence information and a broadcast feature.

http://www.amity-systems.com



http://www.tmcnet.com/639.1

Adtran Simplifies Hosting of VoIP Apps for SMBs

Adtran (news - alert) has released an all-in-one platform to simplify hosted VoIP, Internet access, and business connectivity services for small and medium businesses and distributed enterprises. Converged IP access is delivered by the NetVanta 1355 by combining SIP gateway functionality with advanced routing, PoE switching, security, Layer 3 QoS, and survivability features in a single platform.

http://www.adtran.com

http://www.tmcnet.com/640.1

Veterinary Supply Company Using BorderWare's SIPassure for Secure Remote Connectivity

TW Medical Veterinary Supply is using BorderWare's (news - alert) SIPassure security gateway, which will allow remote users to connect seamlessly to the company network. It will also enable TW Medical to expand its operations beyond its central office, and will make all the pieces of the company's phone system compatible — including its firewall, its VoIP-enabled PBX and its IP phones. SIPassure also provides the Session Border Control functions needed to allow all the components to interconnect seamlessly and securely.

http://www.borderware.com

http://www.tmcnet.com/642.1

ESI Announces Communications Server Family

Estech Systems Incorporated (ESI), (news - alert) a manufacturer of digital and IP-based telephone systems, announced important changes to its product line, including a new family of communications server products designed to offer customers an expanded choice of system platforms. ESI also announced an array of product enhancements designed to provide additional flexibility to its Visually Integrated Phone (VIP) suite.

http://www.esi-estech.com

http://www.tmcnet.com/641.1

'Fringsters' Can Now Use Any SIP Provider to Make Mobile VoIP Calls

Fring, (news - alert) which makes a software application enabling 3G and dual mode WiFi phones to make free, peer-to-peer VoIP calls over mobile networks, announced that its software now enables fring users (known as "fringsters") to choose any SIP provider to make free mobile VoIP calls to regular phones as well. That means in addition to being able to make free mobile calls between "fringsters" and "Skypers," as well as over IM, Google Talk and Windows Live Messenger (MSN), callers can use any SIP-based provider.

http://www.fring.com

http://www.tmcnet.com/644.1

Sipera Viper Lab Identifies SIP Vulnerabilities That Threaten VoIP

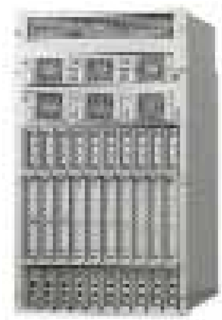
After two years in stealth mode, reviewing, cataloging and analyzing VoIP and SIP vulnerabilities, Sipera (news - alert) VIPER Lab released at CTIA several threat advisories for WiFi/dual mode telephones from vendors including RIM, HTC, Samsung, Dell and D-Link. Sipera VIPER Lab also released information about a number of SIP vulnerabilities. These vulnerabilities can disable phones calling features, disconnect calls, and freeze phones, causing significant enterprise communications disruptions.

http://www.sipera.com

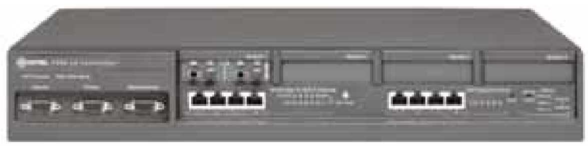
http://www.tmcnet.com/643.1

SPEC to Develop Standard Methods of Comparing SIP Server Performance

The non-profit Standard Performance Evaluation Corp. (SPEC) has formed a new subcommittee to develop standard methods of comparing performance for servers using the Session Initiation Protocol (SIP). Current SPEC member companies committed to developing a new SIP benchmarking standard include CommuniGate Systems, HP, Intel, IBM, and Sun Microsystems.



http://www.spec.org



http://www.tmcnet.com/645.1

Mitel Delivers Direct SIP Connection to Microsoft Exchange Server 2007 Unified Messaging

Mitel (news - alert) announced it has further advanced the completeness and ease of integration between its flagship Mitel 3300 IP Communications Platform (ICP) and Microsoft Exchange Server 2007 Unified Messaging with direct SIP connection capabilities. Mitel's embedded SIP integration eliminates the need for a separate SIP gateway as a go-between from a 3300 ICP SIP connection to an Exchange Server 2007, resulting in support for multiple forms of Unified Communications including voice, email, and fax.

http://www.mitel.com

http://www.tmcnet.com/646.1

Acme Packet Certified with Veraz to Deploy VoIP, IMS

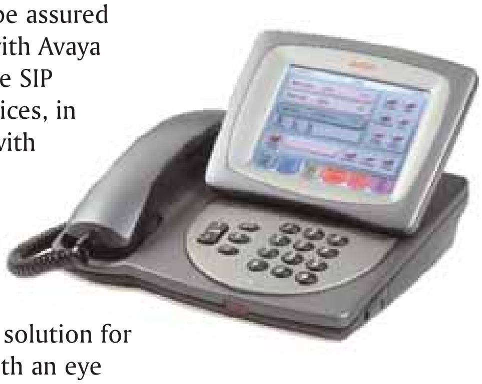
The latest vendor to become part of Veraz Networks' (news - alert) Veraz Open Solutions Alliance (VOSA) is Acme Packet, which has been certified as a "development ready" member with its Net-Net line of SBCs. Acme Packet's (news - alert) products have successfully undergone interoperability testing with Veraz' ControlSwitch softswitch and the I-Gate 4000 media gateway family, and have already been successfully deployed in a number of live networks.

http://www.acmepacket.com
http://www.veraznetworks.com

http://www.tmcnet.com/649.1

BandTel Brings SIP Trunking to Avaya DevConnect Program

Avaya (quote - news - alert) has named SIP Trunking provider BandTel (news - alert) into its Avaya DeveloperConnection (DevConnect) program as a Gold member. As part of the DevConnect program, BandTel's services will be assured compliance with Avaya solutions. The SIP Trunking services, in conjunction with Avaya's VoIP products, will provide a flexible, reliable, and cost effective solution for companies with an eye toward a converged voice and data infrastructure.



http://www.bandtel.com
http://www.avaya.com

http://www.tmcnet.com/647.1

OKI Develops CenterStage NX3200, the Industry's First Session Border Controller for NGN

Oki Electric Industry announced it has developed a session border controller for next-generation networks (NGN), its CenterStage NX3200. "CenterStage NX3200 enables carriers to interconnect their networks, which hadn't been done in the past because it was difficult to process large capacity traffic in high speed with different IPs, such as IPv4 to IPv6," said Kichiro Akino, president of Network Systems Company at Oki Electric Industry.

http://www.oki.com

http://www.tmcnet.com/648.1

Solegy's Managed PBX Solution Delivers Business VoIP Without the Hassle

Hosted VoIP provider Solegy announced the launch of a managed PBX solution that will enable service providers to offer business customers a single-sourced solution for all of their voice needs. Solegy's Managed PBX — part of its ServicePDQ software platform — delivers all the features and functionality of a traditional PBX while adding a range of new IP-based services, enhanced flexibility and expanded user control.

http://www.solegy.com

http://www.tmcnet.com/650.1

InGate SIP Solutions Part of Level 3 VoIP Technology Alliance

InGate (news - alert) has been invited to become part of Level 3 Communications' (quote - news - alert) TAP (Technology Alliance Program), which provides an environment where various industry participants can come together to provide complete, tested, solutions that run on the Level 3 network. Ingate's firewall technology enables SIP-based live communication while maintaining control and security at the network edge, and its SIParator devices connect to existing firewalls to enable SIP communications.

http://www.ingate.com
http://www.level3.com

http://www.tmcnet.com/651.1

4PSA VoipNow 1.4 Simplifies Hosted PBX Business

Rack-Soft announced the most comprehensive update of 4PSA (news - alert) VoipNow released so far. With more than 30 new features and functions, 4PSA VoipNow 1.4 is the new version of the popular hosted PBX platform for ISPs and hosting providers.

http://www.4psa.com

http://www.tmcnet.com/652.1

Arcosoft Intros VoIP Call Status Display Software

More small and medium-sized businesses are using VoIP phones, but lack the tools to effectively use them. Arcosoft (news - alert) has come forward to fill up the void with the introduction of VONaLink TeamOnCall.

http://www.arcosoft.com

http://www.tmcnet.com/653.1

Aspect Enhances Contact Center Experience with SIP Trunking from BandTel

Aspect Software, (news - alert) which focuses solely on the contact center space, has adopted SIP technology, and has successfully completed interoperability testing of its contact center products with BandTel's SIP Trunking solution. In fact, Aspect has been incorporating SIP connectivity into its solution set for some time, and has now taken the next step to providing end-to-end SIP connectivity.

http://www.aspect.com

http://www.tmcnet.com/654.1

CounterPath adds PGP's Zfone Media Encryption to its VoIP Softphone Solutions

VoIP softphone developer CounterPath Solutions (news - alert) announced it has signed a deal with Pretty Good Privacy to integrate Pretty Good's Zfone, a security, privacy and compliance solution, into its Eyebeam 1.5 and X-Lite 3.0 products. In so doing, CounterPath will add a new level of security to its VoIP softphone solutions.

http://www.counterpath.com

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- ☑ Increase revenue by reducing time-to-market for new products
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SPEAKING SIP

The Many Faces of Ice

by
JD Rosenberg

Without a doubt, one of the most challenging issues that VoIP system designers and network operators face is firewall and Network Address Translator (NAT) traversal. These days, almost every home with broadband Internet access has a NAT device — after all, NAT is the primary function of the broadband home router, the little magic that allows you to connect multiple computers to a single Internet connection. Most enterprises have one or more firewalls, and many smaller ones run NAT as well. Even some service providers use NAT; it is not uncommon for a cellular phone to have a private IP address. While NAT and firewalls are not a problem for traditional client-server protocols like those used for the web and e-mail, they are a huge problem for VoIP.

The industry has responded to this problem with many different solutions. These include Application Layer Gateways (ALGs), which add SIP awareness to NAT and firewalls, Simple Traversal of UDP Through NAT (STUN), which uses a "ping server" of sorts to allow low-cost traversal in consumer applications, and Session Border Controllers (SBCs), a close cousin to ALGs. SBCs have won the largest part of the market share of NAT and firewall traversal solutions. All of these techniques have their problems, and so the IETF worked steadily on producing a one-size-fits-all solution. That solution is called ICE: Interactive Connectivity Establishment.

ICE is a peer-to-peer cooperative NAT traversal solution, in which endpoints work with each other to discover paths through the network via a series of connectivity checks. This discovery is done in concert with network servers that help provide relaying and address translation functions. Work on ICE began in early 2003, and finally, a long four years later, it is now complete. ICE is extremely effective. It is robust, finding media paths even in the most complex network topologies. ICE makes sure that the called phone won't ring unless a bidirectional media path is up and running. No more ghost rings and one-way audio that are common problems in VoIP. ICE is efficient, using relays and suboptimal paths only when absolutely necessary. It works across a broad range of environments without changes in configuration. It also provides lots of hooks for policy and allows for an evolutionary path from existing SBCs to ICE-based SBCs.

However, an interesting thing has begun to happen. ICE is also solving problems having little or nothing to do with firewall and NAT traversal. These include security, IPv6 transition, and dual-homing.

What does ICE have to do with security? Many VoIP systems today allow a malicious client to use the VoIP network to launch a denial-of-service (DoS) attack against a desired target. This attack, called the voice hammer, allows a single call-setup message to direct an 80 Kbps stream of packets (and possibly higher bandwidths) at a target device. This attack is easy to launch: An attacker sends a SIP INVITE message but lies about its media address, pointing to the target of the attack instead. Once the call is established, the called party will begin sending media toward the target. ICE prevents this attack. The called party won't send any media at all until the ICE connectivity checks have taken place. Those checks happen along the media path, and in this case, they will fail since the target of the attack won't respond to the checks. Consequently, no media is ever sent and the attack is prevented.

What does ICE have to do with IPv6 transition? One of the primary transition techniques is to use a dual-stack client, one that has both an IPv4 address and an IPv6 address. This introduces an interesting problem: When the dual-stack client makes a call, which address does it include in its INVITE as the target for media, IPv4 or IPv6? At the time it makes the call, it doesn't know the capabilities of the called party, which could be IPv4 only, IPv6 only or dual stack. ICE has emerged as the solution to this problem. The caller includes both addresses, uses ICE's connectivity checks to figure out which pairs work, and then uses them.

More generally, ICE helps dual-homed endpoints — those with more than one IP address. They are more common than you might think. My laptop has three IP addresses — one on the Wi-Fi network, one on the wired Ethernet, and one on my VPN. When I make a call from my softphone, which one should my laptop use? With ICE, my softphone would include all three, and then ICE would be used to dynamically figure out which one works. In fact, ICE can help me pick the one with the lowest latency, in order to optimize my experience in the call. ICE can also have configured policies to ensure only a specific address (such as my VPN), gets used.

These three applications are just the beginning. ICE can address other problems because it adds an important piece of functionality to SIP — exchange of messaging that follows the media path prior to call establishment and prior to the transmission of media. This small but important change will, I predict, make ICE a protocol for all seasons, not just the winter of NAT and firewall traversal. ICE is already considered one of the core SIP specifications by the IETF, and I anticipate we'll see more and more reasons for this over time.

Jonathan Rosenberg is the co-author of SIP and SIMPLE. He is currently a Cisco Fellow and architect for the IP Communications Business Unit in the Voice Technology Group at Cisco (quote - news - alert) (http://www.cisco.com).

PRESENCE ENABLED

Think (Really, Really, REALLY) Big

by
Joe Hildebrand

Back before we were storing our photos, music, movies, and other rich media on our computers, it was hard to conceive of the need for a 100GB hard drive on a personal computer — much less the possibility that such a hard drive would be insufficient for one person's storage needs. But now we know better. I know I'm constantly scrounging for space on my laptop.

Yet sometimes when we talk about **presence** and the imperative to make it massively scalable, we hear two objections:

1. Even the largest enterprises don't need to scale beyond a few hundred thousand concurrent users.

2. Even the largest carriers don't need to scale beyond a few million concurrent users.

Those objections have a surface plausibility, until you realize that many people already have multiple points of presence: mobile phone, PDA, laptop running several presence-aware applications, not to mention services such as a weblog or photo site. Add to that the fact that any addressable application, device, sensor, or service is a potential point of presence, and you begin calculating the need for scale exponentially.

I might have 5, 10, 50, 100 or perhaps many more devices and services associated with my aggregate presence, and you might have just as many devices and services that want to know about my presence. And that's just presence about people. But standalone devices and applications have presence too. Think presence-enabled vending machines, roads, garage doors, appliances, pets, workflows, inventories, programmable logic controllers in factories, utility grids, news stories, discussion forums, and so on.

It remains to be seen how some of these nodes will create value by generating or consuming presence. But we have seen the number of applications, mashups, services, devices, and sensors proliferate. There is no reason to think that the pace of creating such nodes will do anything but accelerate.

We know that many of these nodes will be presence-enabled because basic and extended availability information gives the consumer awareness of transient state changes that are significant for the purpose of real-time communication and collaboration. Presence is a kind of glue because it unifies communication across nodes, and it is a kind of catalyst because it drives further communication between nodes.

This is true in a wide range of real-time applications, whether based on SIP, XMPP, or some other technology. In today's real-time world, you don't necessarily know that you want to initiate a communication session (VoIP, instant messaging, whiteboarding, etc.) until you have presence about other people or applications on the network.

Indeed, presence is already proving useful for delivering on the promise of autonomic computing (fast reactions to changes in computing system state) and for making other processes autonomic (think of the world's best traffic light timing, real-time supply chains, and the like).

Given all these factors, the conclusion that we will need massive scalability in our presence systems is close to obvious.

Once you think beyond instant messaging as the only presence-enabled application, you start to see presence as a kind of communications middleware. The catch is that if you try to attack the problem with traditional polling or transactional processing middleware, your deployment is going to die a painful, senseless, and untimely death.

Painful because you are very quickly going to feel the need or desire to connect an ever-increasing number of presence points — and a polling-based architecture will collapse on the weight of presence pings alone.

Senseless because full transactional processing is simply overkill for the types of messages these services need to send and receive.

Untimely because your presence architecture will publish stale information, not real-time presence. That someone or something was available is of little value compared to knowing that they *are* available right now, in real time.

We noted that presence provides transient notification of significant state changes. From an architectural perspective, the key is that presence is transient and timely, so high scale and low latency are both critical.

To achieve scale, presence middleware needs to be event-driven: messages are exchanged only when something changes. To achieve low latency, it needs to dispense with the notion of guaranteed delivery: 99.9% message reliability is plenty good enough when publishing an "on the phone" presence change. And to deliver significant information, it needs to know what it doesn't know: i.e., it needs to be open and flexible enough to allow for a nearly infinite number of extensions to meet future requirements.

The power of presence for real-time communication and collaboration has only just begun to make itself felt. Innovative enterprises and service providers need to start thinking big if they want to take full advantage of the presence revolution.

Joe Hildebrand is CTO of Jabber, Inc. (news - alert) For more information, please visit the company online at http://www.jabber.com.


CAI7000 PBX
Circa 1997

TMC Labs Review

Powerful VoIP phone systems can be had at a much lower cost than five years ago. That is partially due to inexpensive open source solutions, such as Asterisk, which runs on Linux. Although Asterisk is a very popular IP PBX solution, because it runs on the Linux operating system, it may intimidate smaller businesses with limited or no Linux expertise. While there are ways of getting Asterisk to run on Windows, it doesn't have professional support. So, what options are available to a small business to have an inexpensive VoIP phone system that runs on Microsoft Windows? Well, one interesting option comes from pbxnsip, whose core roots came out of snom, manufacturers of inexpensive SIP phones. Indeed, snom is well known for its SIP expertise and superb SIP interoperability, so it was no surprise to TMC Labs that pbxnsip is a 100% SIP-based IP PBX.

pbxnsip
1600 Osgood St
Bldg 20 Suite 223
North Andover, MA 01845
Website: http://www.pbxnsip.com

Pricing:

Various versions are available, including, but not limited to:

General Enterprise:
- Office 10 - $690
- Office 25 - $1,490.00
- Office 200 - $8,950.00

Call center functionality:
- Center 25 - $2990.00
- Center 50 - 4,950.00

RATINGS (0–5)

Installation: 5
Documentation: 4
Features: 4.25
GUI: 4
Overall: A-

pbxnsip (news - alert) offers a virtually plug-and-play multi-tenant IP PBX, which can be installed in just minutes using your typical Windows installer. TMC Labs installed pbxnsip in a Windows 2003 Server very quickly with no installation issues. All that had to be done after installing was launch a web browser and go to the local logon URL, http://localhost. The pbxnsip program has its own built in web server to serve the pages, so make sure IIS is not running on the machine to avoid port conflicts on port 80 and 443. The system offers three levels of login, so you can login as the system administrator, a domain administrator, or a user. After creating the accounts, SIP phones retrieve their configuration information directly from the PBX once you bind the MAC address to the extension via the GUI. Unlike competing solutions, you don't have to set up a separate tftp server and create the configuration files. pbxnsip comes with a built-in tftp server that generates the necessary configuration files on the fly for popular SIP phones, including Polycom, Cisco, and snom. The pbxnsip system can also configure itself by using a configuration wizard. The installer goes to a web screen, answers some questions, and then copies the URL over and inputs it in the configuration screen. Then it will download the xml created from the wizard and automatically configure itself. http://www.pbxnsip.com/configurator/wizard is an example of what can be done.

Figure 1: Web-based admin screen

Once logged on, it was easy to quickly setup extensions, hunt groups, auto attendants, and TDM trunks, as well as SIP trunks. The interface (See Figure 1) was pretty easy to navigate, though, in some ways, the GUI was too basic and we did find some things a bit confusing. There was no "tool tips" or help options next to field names to describe the fields and give sample values. Since the target audience for this product is the small to mid-size business (SMB), a little more work on the administrator GUI to make things easier would be a nice touch.

pbxnsip comes with your basic telephony features — parking, transfer, conferencing, hunt groups, voicemail, auto attendant — but it also has some pretty nifty advanced features, like call forwarding, hot-desking, and even the ability to email the current call record (CallerID) to yourself. For example, suppose you are on the phone with a client and need to call them back. If you configured your extension with your email address, instead of asking the person on the phone to give you their phone number and write it down you can direct the PBX to send you an email with the call details by simply pressing *63 after the call. The system will send you an email with the Caller ID, the duration of the call, and the time of the call. Another advanced feature useful in the call center space is the ability to barge in to an agent's call, as well as teach, or simply listen.

The pbxnsip PBX uses https, SIPS, SRTP, and SDES to make the communication to your PBX secure. Simply by using SDES-capable devices, your voice calls will stay as secure as your https traffic. For the exceedingly security-conscious (e.g., government offices), the professional version allows you to enforce end-to-end security. The professional version will only allow calls only if the other side of the call also supports secure calls. If that is not the case, it will tear down the call. For trunks, the administrator can specify which trunks are treated as secure and which trunks really need to communicate via secure protocols. Even the management interface can be secured since the system supports https.

On the trunk side, since pbxnsip is 100% SIP-based, it easily communicates with any SIP-based ITSP. It can also communicate with the PSTN network using any standard PSTN gateway, such as AudiCodes, Mediatrix, Vegastream, Quintum, and so on. It is also worth mentioning that pbxnsip comes with built-in mini-session border controller (SBC) that allows your remote users to register, even if they are behind NAT. On a related note, pbxnsip supports ENUM, which will allow for direct SIP-to-SIP URI dialing.

One productivity enhancer is the auto attendant's "anonymous screening" feature, which can intercept anonymous CallerID calls and ask them for their name before putting the call through to an extension. Then you have the option of accepting or rejecting the call. Similarly, the PBX users can tell the auto attendant that they do not want to be disturbed or that incoming calls should be redirected to internal or external numbers. Users can register several phones with the same number (home and office) and the auto attendant will ring them simultaneously (find me/follow me). Hunt groups are supported and can call extensions by parallel or sequential forking. For each stage, you may specify which extension should be called and how long the PBX should stay on the stage. If all stages should fail, you can send the call to an automatic extension, such as an auto attendant or a park orbit, or to an external number.

pbxnsip fully supports message waiting indicators (MWI). In addition, if configured, you can have the PBX email you when a voicemail has arrived (unified messaging). The mailbox supports the usual functions — like urgent message marking, moving messages to other extensions, personal greetings, and PIN code access — from outside. The system can also call you your cell phone if someone leaves you a voicemail.

pbxnsip supports intercom paging — what it calls "Push2Talk" — which enables you to call a Push2Talk extension number, and all associated extensions will receive a one-way audio call and play back your announcement. For instance "Sales personnel please comes to my office" or "Chinese food is here." Also, SIP phones that support the IETF draft dialog state will be able to show you which lines are ringing and in use. You will be able to pick up calls in your hunt group and from your colleagues before your phone starts to ring by just pushing the key next to the LED.

pbxnsip does support call recording, simply by pushing a record button on your phone. However this feature currently works only with snom phones since it has a special record key that sends any INFO with record on/off when pressed. In the 1.5 release, the * and # keys could trigger recording on non-snom phones, but that would sometimes wreak havoc when checking remote voicemails, so pbxnsip enhanced the feature in 2.0. Now, users can have all the calls recorded coming in from a particular agent group or hunt group, or all simply all outgoing calls. The feature is now more granular and the .wav files are in the extension directory to be played back later. This is an optional feature in 2.0 and is part of the call center edition. Finally, the PBX includes support for multiple conference bridges that can be pre-configured. Participants simply call into the conference room, or they can be blind transferred into the conference number, which can be password protected with a PIN.

Also new in the 2.0 release is the user portal, allowing users to log in to the system and listen to their voicemails, view their call detail records, and set up the forwarding and cell phone numbers. The cell phone feature is particularly nice, since now, when someone calls your extension, it will automatically ring your cell phone at the same time or with a configurable delay. This is nice if you are on the road or even in the office away from your desk. Another new feature in 2.0 is the integration with Exchange Server 2007, which provides an integrated email/voicemail box. In this mode, all the voicemail prompts and dialogue are served and stored from the Exchange 2007 Server.

It is also important, especially in the SMB market, to be able to simply plug in the SIP phones and automatically get the next available extension without having to jump through hoops. pbxnsip supports this ability to auto-assign extensions (See Figure 2). Administrators can define that extension numbers are assigned on a first come/first served basis, that extensions are assigned only if no one else is registered, or explicitly specify the MAC address to a specific extension.

Other features:

- Queues with on-hold music (support for CD/MP3 players)
- IVR support with database lookup using SOAP XML (professional version only)
- View active calls, call history, and generate CDRs

Trunks:

- Calls placed outside of the PBX are handled by trunks. Calls to trunks are handled by the B2BUA of the PBX, so that advanced features like transfer are not visible on the trunk side. This is an important feature for many ITSP providers that are not able to provide you this feature. You can connect trunks to a local or remote PSTN gateway.
- Dial Plans with ability to use regular expressions for powerful digit matching
- SNMP allows measurement of busy hour call attempts and busy hour calls, how many registrations the PBX keeps, and how many internal calls the PBX has open.

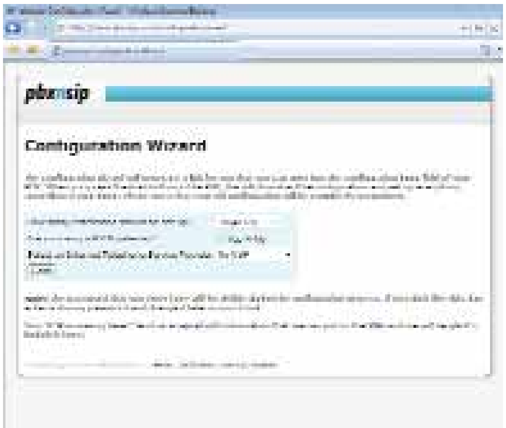

Figure 2: Automatic extension configuration

CONCLUSION

Miercom, a partner lab for TMC's publications, tested pbxnsip and ran overnight benchmark/performance tests using a 3.4 GHz Pentium D machine running Windows 2003 server. They were able to blast 60 simultaneous calls using WinSIP and encountered no errors. Considering the average 4:1 (busy to idle) call ratio, that would equate to a 250 seat IP PBX system, which is pretty impressive. We should mention that, while pbxnsip runs on Windows, since it is written in C++, it is also available on Linux and NetBSD. Overall, we liked the easy to use web-based administration tool, which made configuring and maintaining pbxnsip a snap. The fact that it easily installs on a Microsoft Windows 2003 Server will make Microsoft shops happy. It also runs on XP PRO and even Vista. As previously mentioned, it could use some more context-sensitive help, but overall, the admin GUI was pretty good. Most impressive of all is the advanced features you get for a rock-bottom price. The SMB market looking for a feature-rich IP PBX that won't break the bank would do well to consider pbxnsip. Demo licenses can be attained from pbxnsip's website at http://www.pbxnsip.com/TMC

SIP and Open Source

By Richard "Zippy" Grigonis

In recent years, open source telephony software has made as great an impact on the telecom scene as has Internet telephony itself. SMBs (Small and Medium-sized Businesses) are tantalized by the prospect of setting up a $900 to $2,500 IP PBX. Established PBX makers, however, are horrified that open source software has "moved the decimal point" over two places, and that an ever-expanding group of nameless, faceless people scattered around the globe are working diligently — and at little or no salary — to usurp their long-time dominance of the industry.

The earliest and premier open source PBX software package was Asterisk, which runs under another famous open source software project, Linux. Mark Spencer, the president of Digium (which makes I/O and other hardware that's compatible with Asterisk) got the Asterisk ball rolling about seven years ago. Asterisk now offers just about anything a business would want in terms of a PBX, such as voicemail, IVR, auto-attendant, overhead paging, call parking, VoIP, PRI compatibility with many central office switches and codecs.

Ironically, Asterisk's native protocol is not SIP, but IAX (Inter Asterisk Exchange) — IAX2, to be exact, a protocol based on UDP (User Datagram Protocol) protocol. That's not a problem, since Asterisk can work with SIP too. Asterisk can act as a SIP client, server and media gateway. In Asterisk, every call is placed or received on a separate logical "channel" that connects the Asterisk server and some other VoIP server, such as one in a company's branch office. The two primary Asterisk VoIP channels are for IAX and SIP. Ironically, IAX is better at penetrating any given company's NAT (Network Address Translation) firewall barrier than SIP (or Ye Olde H.323 protocol for that matter), since it needs only a single port, UDP 4569.

SIPfoundry (http://www.sipfoundry.org), an international open-source community dedicated to speeding the adoption of SIP applications as well as the underlying technology, has focused more on SIP. To be specific, an open source, native SIP and Linux-based PBX called sipX. SIPfoundry (news - alert) calls it an ECS (Enterprise Communications Server). The system is based on SIP URI addresses that the company over time believes will replace standard PSTN phone numbers. sipX has a modular architecture and supports the exchange of just about any kind of real-time information — voice, video, IM, collaboration, etc. The modular sipX system runs on standard Intel servers and allows 12 different server processes to coexist on one server, or they can be distributed to different hardware systems. It also offers call traffic load balancing and high availability redundancy between call control components.

Instead of being based on a B2BUA (Back-To-Back User Agent, which is a SIP logical entity that can receive and process INVITE messages as a SIP User Agent Server), sipX implements a true SIP proxy architecture. This enables one to revel in some of the more interesting aspects of proxy servers, such as forking (A forking proxy can forward a SIP request to multiple SIP addresses and return the responses to the sender.)

The sipX package has the “look and feel: of an IT application, and it supports “plug and play” IP phone management. It also supports Pingtel’s ACD Call Center server.

Speaking of Pingtel (http://www.pingtel.com), they’ve taken SIPfoundry’s raw open source code and have fashioned it into enterprise-level PBX and SIP router solutions. Pingtel (news - alert) retains the SIPfoundry term ECS or Enterprise Communications Server, but Pingtel’s SIPxchange ECS is much more. Pingtel has added enterprise-grade quality, reliability, support, documentation to the original SIPfoundry system, in much the same way that Red Hat has “spruced up” the open source Linux code produced by the Fedora open source community. (Indeed, SIPfoundry was founded by Pingtel, the ReSIProcate community and some members of the Vovida community.)

Despite its many improvements, the SIPxchange ECS is still capable of providing low-cost IP PBX, key system, branch office, and call center solutions, any of which are capable of integrating with legacy TDM and IP networks.

It looks like the Pingtel’s business model of improving upon and packaging the open source telephony architecture — adding bells, whistles, documentation, support and professionalism — could the be the future method of choice for monetizing open source telephony.

Richard Grigonis is Executive Editor of TMC’s IP Communications Group.

FEATURE ARTICLES

Exploring QoS in SIP-Based Networks

By Richard "Zippy" Grigonis

Since the dawn of VoIP, the chief concern of users has been Quality of Service (QoS), or the quality of a voice or video transmission. But what is this "quality"?

From the perspective of a typical user, a phone call (or video conference) either sounds good or it doesn't.

Network engineers, however, have a different definition of QoS. To them, technically, QoS is an analysis of delays in IP packet arrivals (anything above 150 milliseconds is considered unacceptable), packet loss (you could consider these packets as having an infinite delay in arrival), and jitter (variations in packet delay).

Still another view of QoS comes from IT department managers who hold a more business-like perspective: QoS tends to be linked more with the telecom provider's SLA (service level agreement) and so the network must meet a certain level or standard of availability, such as no more than an hour or two of downtime per year, that users and applications will be allocated the proper bandwidth, and that the network's performance will meet some specific requirements favored by the business.

While many technically-minded people think of QoS as dealing with impediments in the flow of packets through a network, QoS in fact encompasses the workings of all seven layers of the OSI (Open Systems Interconnection) Model. Operating systems (most operating systems don't handle real-time processes very well), competing data streams, communications protocols, scheduling and traffic management issues, all come into play. Indeed, QoS can involve any process or network elements stretching from one endpoint to the other, since the quality of a conversation is only as good as the weakest link in the transmission "chain" between endpoints. Everything from a burned-out router to a buggy softphone application running on a laptop with a defective network interface card can affect QoS, regardless of the pristine nature of the rest of the network. Thus, the ultimate judgment of QoS is the subjective rating of the actual user of IP Communications.

The most popular way to measure telephony voice quality is the MOS or Mean Opinion Score, a number between 1 and 5 used in an attempt to quantitatively express the subjective quality of speech in communications systems, particularly digital networks carrying VoIP traffic. Anything above a 4.0 is considered toll grade.

It's quite easy to achieve a MOS of 4.4 on a LAN, even using big (1,500-byte) clunky Ethernet packets, since the bandwidth is high and the enterprise owns the whole LAN, and can guarantee the quality of network equipment and cabling end-to-end. Once voice or video packets are sent across the public WAN, however, the MOS can easily drop to 3.0, since no individual corporation owns network paths from end to end.

Bandwidth vs. Prioritization

The two principal competing methodologies for keeping QoS at acceptable levels has been bandwidth overprovisioning *versus* traffic engineering or prioritization.

Provided that all of your network elements are in tip-top shape, "limitless" (i.e. optical) bandwidth yields superb QoS for all traffic flows. These utopian conditions are almost always impossible to achieve, which means that network operators must resort to the second option, traffic engineering.

Traffic engineering itself falls into two categories:

Integrated Services (resource reservation), where network resources are apportioned according to an application's QoS request, and subject to bandwidth management policy. The ReSerVation Protocol (RSVP) provides the signaling to enable network resource reservation.

Differentiated Services or "DiffServ" (prioritization) where network traffic is classified and network resources are apportioned network resources according to bandwidth management policy criteria. The delivery of packets is "prioritized" in the network, depending on the application that originated the packet, thus yielding the concept of Class of Service (CoS). As is the case with many systems (Cisco, for example), applications are associated with three grades of service. Voice, video and other real-time applications get the highest prioritization and thus the most preferential treatment by the network elements. This is the most popular way of maintaining high QoS, if only because some 800-pound gorillas (Cisco, Juniper and Avici) favor it.

The principal technique that tends to be used in a differential services environment is MPLS (Multi Protocol Labeling Switching) which takes aggregates of packet streams and determines the priority of packet delivery based on labels inside (encapsulated by) the packet headers. MPLS does its job inside of routers and can handle multiple, non-IP protocols, so it works with ATM, IPX, PPP or Frame-Relay. It can even run directly over the OSI data-link layer.

Note that a kind of resource reservation occurs with Diffserv and MPLS, but its "guarantee" of quality is more statistical in nature. Instead of monitoring the call status and updating the reservation of resources (as in the case of RSVP), in DiffServ the call status is monitored and the necessary bandwidth is calculated. RSVP comes closer to continually achieving toll-grade quality than DiffServ techniques, but the cost of resource reservation with RSVP, which involves many communications with UDP messages to and from routers, is too high.

Similar QoS challenges occur in wireless networks, perhaps more so, since even 3G wireless connections are of lower bandwidth and are prone to interference but nevertheless must support multimedia applications with pretty high QoS requirements. The IMS (IP Multimedia Subsystem) framework specifies that end-to-end QoS support requires signaling, traffic regulation and resource allocation capabilities. QoS signaling can provision and enforce QoS parameters between endpoints and operates in the OSI application layer, network layer and link layer. Session-specific QoS parameters can be exchanged via SDP or SIP header fields.

Quiz That QoS!

Since the whole future of IP Communications hinges on voice and video quality, a vast sub-industry of companies that deal with testing and monitoring networks for QoS has arisen. Psytechnics (http://www.psytechnics.com) for example, became a major force in this business after it was spun-off from British Telecom with backing from 3i.

Psytechnics' (news - alert) products furnish performance information regarding the design, optimization, and monitoring of PSTN, mobile and VoIP telecommunications networks. The company prefers to use the term QoE, or "Quality of Experience" to describe their bailiwick, rather than QoS. They feel that QoS is merely a network-centric, technical analysis of bits and bytes on a per application or link basis. The user experience is really of primary concern, so they've replaced QoS with a more user-centric paradigm, QoE, that scrutinizes individual user experiences, with in-depth application intelligence that captures experience information on a per-user and per-session basis, said to be invisible to more rudimentary
QoS tools.

Recently, Psytechnics performed a voice quality performance evaluation study of a pre-release (beta) version of Microsoft Office Communications Server 2007 and the Microsoft Office Communicator 2007 desktop VoIP solution. The study also included a comparison of the Microsoft solution to some prototype USB handsets as well as Cisco's CallManager 5.0 and 7961 IP phones. Both the PC VoIP solution and the IP phone were evaluated by both real end-users (the subjective tests) and by Psytechnics' QoE software (the objective tests). Psytechnics' evaluation reveals that "Overall, the one-way listening speech quality provided by the combination of Microsoft's client and a USB handset was consistently better than that provided by Cisco's IP phones and CallManager, whether using G.711 or G.729." Psytechnics has demonstrated that the quality of Microsoft's offering is high enough so that companies should feel free to integrate voice communications with PCs, which in turn suggests that they could eliminate the purchase of expensive IP phones almost entirely if they so desire.

As Mike Hollier, CTO at Psytechnics, says, "This evaluation emphasizes the positive transformation that software-based VoIP solutions will have on unified communications and telephony in the future. The familiar PC can now outperform the IP phone."

Ironically, for its new Response Point small business solution, Microsoft has begun partnering with handset vendors to make compatible endpoints; e.g., the D-Link DVX-2000, Quanta Syspine and Uniden Evolo.

Another company in this space, Telchemy (news - alert) (http://www.telchemy.com) is known for its VQmon and SQmon families of service quality monitoring and analysis software products, technology that enables both service providers and enterprises to monitor and manage the performance of VoIP, IPTV, IP videoconferencing, high definition Telepresence, 3G / 4G mobile and other converged real-time services. Telchemy's products provide real-time visibility of service quality, accurate estimates of user experience, QoS, IPTV QoE, VoIP QoE (MOS scores and R factors), and detailed analysis of the root cause of quality degradation.

VQmon integrates into both network infrastructure and test equipment. Once there it can provide perceptual quality scores for every call, reporting metrics using RTCP XR (RFC3611), SIP RTCP Summary Reports and other key protocols. Telchemy's also relies on OEM probes and distributed active monitoring applications.

Telchemy's latest technical achievement is DVQattest/EN, an active test tool for VoIP, IP videoconferencing and high-definition telepresence service assurance. Available as software for license to OEMs and network equipment vendors, DVQattest/EN uses VQmon technology to provide network assessment, pre-deployment testing, SLA monitoring and advanced network troubleshooting for enterprise networks. DVQattest/EN sessions can generate 200 concurrent VoIP streams, 20 concurrent high definition 1080p simulated IP video streams and a range of network diagnostic tests, with an interactive application for configuration and reporting. Despite its high level of sophistication, the DVQattest agent is small enough for direct integration into network equipment.

The Quality of Packets is Not Strained

All in all, quality of service is really an umbrella term for a collection of defined user experiences and technologies which allow network-aware applications to request and receive predictable service levels in terms of data throughput capacity (bandwidth), latency variations (jitter) and propagation latency from QoS-enabled IP networks which can respond to requests from critical applications for either resource allocations or differentiated levels of service among shared resources.

Ultimately, however, the only thing that matters is whether or not you like the way a voice call sounds or a video call looks.

Richard Grigonis is Executive Editor of TMC's IP Communications Group.

VoIP Quality of Service Myths

The folks at Psytechnics (http://www.psytechnics.com), the masters of IP Communications testing, are fond of pointing out and clarifying some of the cherished myths about IP-based voice quality still believed by may IT departments.

Myth: IP QoS will solve all of my VoIP issues.

Reality: Increasingly the challenges in major deployments are not at the network level in the IP infrastructure, but are at the application (voice level), in interconnecting with the PSTN or legacy voice networks, which can introduce noise and distortion or intermittent faults at gateways.

Myth: A good IP SLA tool will measure QoS for voice.

Reality: IP SLA tools measure network level QoS performance very effectively, but they do not tell you anything about what an individual session (a call) on the network experiences. Additionally, a user's experience of the call is affected by much more than the IP performance. This is why Quality of Experience tools are increasingly being used in VoIP roll outs. Especially tools that work across both the traditional voice world and the VoIP world.

Myth: VoIP quality is lower that the PSTN or traditional voice networks

Reality: New forms of encoding voice (CODECS) are starting to be used which actually produce voice calls that give a higher perceived quality than traditional phone calls. This can be verified with perceptual measurement tools, such as the ITU approved PESQ, used by the telephone companies to measure their voice quality.

Myth: This CODEC is better than that CODEC.

Reality: Different CODECs (compression/decompression methods) do result in different quality of voice, but also different IP handsets and gateways respond in very different ways to packet loss and jitter in the network. This is why handsets must be chosen carefully, and any IP measurement tools must take account of the brand of handset or they will be inaccurate.

Myth: Voice quality in an IP network is based on Jitter and latency in the network.

Reality: Real network deployments show that there is often little correlation between the MOS (Mean Opinion Score) that IP systems and IP management tools report and the actual user experience. This is why standards-based quality assessment tools are useful when a deployment is underway. You can achieve consistent and accurate measures that will help you pin-point faults, rather than over-optimizing the network or wasting money on excess bandwidth.

ENUM and a Shared Registry Infrastructure: Now Comes the Hard Part

By Steve Granek

ENUM was developed as a set of IETF specifications to enable the Internet community to translate telephone numbers (PSTN addresses) into Internet addresses. When you hear people talking about "public ENUM," this is what they mean. It was hoped that ENUM would be a catalyst for allowing IP services addressed by telephone numbers — such as VoIP, MMS and video conferencing — to be delivered between users globally, just as the PSTN can deliver circuit-switched phone calls anywhere in the world.

Further, ENUM was seen as necessary to ensure that end users using PSTN and VoIP phones could talk to one another, thus maintaining a global voice community during a period of convergence. Based on what Public ENUM was conceived to do, it's very possible that it may remain a useful model for an open, egalitarian and user-centric communications system in which "endpoints" rule.

However, end users generally don't give a hoot about ENUM, VoIP or anything else that we "tech types" in the industry spend inordinate amounts of time thinking about and planning for. The consumer only cares about two things: whether they can reach anyone (and, increasingly, any service too) anywhere at any time via phone, and whether it's going to cost them a reasonable amount to do so. (We've seen evidence of this in countries that have rolled out public ENUM to lukewarm receptions. Clearly, the old maxim "build it and they will come" doesn't apply to public ENUM, at least not at present.)

In parallel, carriers, service providers, content providers and emerging new business entities face major connectivity challenges as they try to roll out new services. As a result, many industry insiders have analyzed the possibility of using what some term "private ENUM" or "carrier/operator ENUM" technology to assist with interconnectivity. If the problem was actually the same as for end users, this use of the term ENUM would be benign, but in fact it leads to lots of confusion, as we shall see. *So, to be clear, this article is NOT about "public ENUM."*

It's easy to see why some players — especially emerging VoIP-based service providers — are so interested in a "private" version of ENUM: They simply want to avoid PSTN charges. Whenever they can complete a voice call without traversing the PSTN, they shield their business models from the economics of the PSTN environment for call termination. Whereas a single service provider in a single region can realistically accomplish this without ENUM technology or any of its constituent parts, things can become a bit more unwieldy as soon as scale becomes an issue — and this is where ENUM technology can be of best use. ENUM technology can facilitate the linking of disparate network "islands," and is imperative when customers are seeking to interconnect using phone numbers.

Interestingly, and perhaps surprisingly, the market in which ENUM technology has been most commonly deployed is to support mobile picture phone services (MMS) between mobile operators. This is because within the United States, MMS was the first carrier-launched native IP multimedia service for which phone numbers were used as addresses — and local number portability made it difficult to properly identify the address of a cooperating carrier's multimedia service center, or MMSC. (Those unfamiliar with wireless can think of an MMSC as a "softswitch" for picture phone services.) ENUM technology was particularly useful in this instance, because it provided a pragmatic way to reuse existing telephone number data. NeuStar has been operating such a service for Tier 1 mobile operators since late 2003, and there are other similar services in the market as well. As a testament to its efficiency, it's been adopted for many SMS services as well.

Addressing v. Routing: Similar, But Not The Same

Addressing and routing are often lumped together in discussions about ENUM. Addressing is the primary focus of ENUM technology and is arguably the foundation for network and service interoperability, but defining its role in IP is a bit more challenging.

With the advent of a multi-service IP environment, telephone numbers have gone from static port addresses on a switch to versatile "subscriber tokens" for a variety of services including MMS, SMS, Instant Messaging, Push-to-Talk and presence. In this scenario, ENUM technology can make the difference in identifying the service for which the number is being used. ENUM reliably answers the question "what network element address must I use to communicate with the subscriber associated with this number?"

It's tempting, yet a trap, to assume that as long as you know the "voice carrier of record" for a particular number, you will be able to interface with their border element. That's because service delivery for subscribers' IP services are likely to be shared more and more. Going forward, it is reasonable to expect a shared service environment as the norm, and as such, the infrastructure and governance structures the industry creates must be designed to support such an environment. In case anyone thinks this is a pipe dream, we already see Mobile Virtual Network Operators (MVNOs) choosing to operate some services themselves, but relying on their network partner for voice services.

Routing, on the other hand, is the application of policy by a sending network, and it all starts with the ending address of a particular service. In this way, the sender is empowered to apply the business policy of its choosing. Here's a very rough analogy: I'm a traveler, and I want to fly from New York to Paris. I initiate the transaction by purchasing a ticket from the airline that can fly me to France on the terms most acceptable to me. At this point, I turn over the control of routing policy to the airline, but I have total control to start, because I know the destination. I am not *forced* to route through that airline; I may choose another, or perhaps I'll decide to take another mode of transport.

All of this may sound completely obvious, but it is often overlooked when considering the many variables surrounding "private" ENUM. With IP services, the operator of a particular service needs to advertise the *address of ingress* for that service to its trading partners. And this may be the heart of the matter — provisioning at scale in a multiparty dynamic environment. For the rest of this article, let's analyze what's needed from a conceptual standpoint.

The Need For A Shared Registry Infrastructure

Earlier, I described an environment in which many network operators cooperate (and often compete) to fulfill the service expectations of their subscribers. In the case of communication, this expectation is near universal global reach. What is most needed now is a way for every operator to declare to all other operators:

- The service the operator provides
- The telephone number associated with the service
- The address of the network element with which to handshake (e.g., a URI)

Service operators need an environment where, by policy, they can expose only the information they want — and then, only to parties to whom they want to expose it. They argue that unlike public ENUM, this should be a *private* environment.

As we have seen, a single phone number may have multiple service operators associated with it (e.g., MMS), and the existing regulatory environment ensures that there should be *one* authoritative answer. This may not be the case with emerging services, so one of the things needed most is a set of agreed-upon rules so that the service declarations that operators make are understood, trustworthy and serve to empower routing policy — and are NOT used to abuse the cooperative nature of the system.

Put another way, we need a *multi-service shared registry infrastructure* where this information can be cross provisioned — a kind of authoritative "Yellow Pages," if you will. The Number Portability Database and its surrounding business processes in North America (such as the Number Portability Administration Center, or NPAC) are an early, "pre-IP services" attempt to achieve this kind of dynamic system. As it currently stands, the existing voice addressing infrastructure i s simply inadequate. It may be *adaptable* — in fact, some in the U.S. have suggested adapting the NPAC to fill the role — but the industry, collectively, will have to decide the right course forward.

From here, the rest of the ecosystem falls into place. In fact, there's been much work done on this using ENUM technology (and even SIP) as the query/response protocol. There are a number of high-quality ENUM caching servers and services designed to answer queries — a rough analog of SS7 SCPs. Some network elements do not support ENUM yet and must use SIP for now, though ENUM is generally thought to be the more efficient protocol for this.

Separation of Roles

What else can we learn from the NPAC? It seems to work best when the operator of the registry/addressing infrastructure cannot exploit its position, but rather operates in a way to maximize choice and competition in the rest of the ecosystem. With the NPAC, there are authorized users who get neutral and equal access to data. The registry operator does not have any say as to the contents of the database; it simply synchronizes the declared service addresses industry-wide. It cannot "game" the addressing system for its own benefit, or to the benefit of any specific "partner." Further, it has no special privileges regarding use of the data; in fact, it does not own the data.

Empowered with information, service operators are free to apply their own policy to routing. Should they choose to outsource a route, that partner can use the same destination information to make similar decisions. Notice that in all cases, the router has access to the destination address information, and thus can freely apply policy. The end service operator is the provisioner (what I called the "declarer") of the service ingress address. This is essential in maintaining the distinction between the addressing and routing roles — and, ultimately, the integrity of the overall system.

Business: Does ENUM mean "free?"

This discussion would be incomplete without mentioning business models. Many argue that the whole point of ENUM is "free termination" or "bill and keep" business models. They may be right — but they may not. Regardless, that is a different part of the interoperability problem and tangential to solving the problem of IP service addressing. It is not the role of an addressing infrastructure to impose inter-party business rules — not now in the PSTN, and not in the future with rich IP services. That's up to trading partners.

Conclusion

While some think there ought to be a single central database, others contend that there should be a system in which one master database is logically divided into a series of authoritative master databases. Both fall into the category of "registry infrastructure." Without such a registry infrastructure, we will have many network elements that are ENUM-ready and capable of answering queries — but we will have no assurances that the answers are right or trustworthy. The registry master(s) must be single authoritative databases; some have called this concept "the golden copy." In other words, with no authoritative registry infrastructure to maintain the integrity of a global addressing scheme, interoperability will be hopelessly inefficient at best, and more likely simply not work very well. I reiterate: *Addressing is the foundation for all interoperability and routing.*

Steve Granek is Vice President, IP Services, NeuStar, Inc. (news - alert) For more information, visit the company online at http://www.neustar.com.

Do SIP Services depend on the emergence of a global ENUM system?

Answer: SIP and ENUM are often lumped together, but SIP does not technically "need" ENUM. At its most basic, ENUM simply translates phone numbers into IP service addresses; services that are not addressed with phone numbers do not need ENUM. However, it is clear that phone numbers are not going anywhere anytime soon. Even when people choose a name to dial on our mobile phones, the underlying infrastructure uses a phone number — and for someone addressing a service to the IP world from the PSTN world, a telephone number is the only choice.

In addition, services that are NOT SIP-based can also benefit from ENUM; we've cited the MMS example as one that already exists. In summary, ENUM and SIP are not co-dependent. (Having said that, SIP services will be rolled out in a world where services will interoperate between the PSTN and the IP world, and ENUM will certainly ease adoption of SIP services as a result.) IMS standards — all of them — assume the need for an inter-party addressing environment based on ENUM technology.

ADVERTISING INDEX

Q & A

60 Seconds with Pedro Quintas, Founder & CEO of COLLAB

by
Richard "Zippy" Grigonis

Pedro Quintas founded the highly innovative 3G audio/video call center company COLLAB (http://www.collab.pt) in 2003. Previously, he was an integral member of the team that developed and launched Altitude Software (then called Easyphone), an award-winning company working in the international call center marketplace. After holding various positions, he was appointed Altitude's Chief Technology Officer in 1998. Prior to joining Altitude Software, Quintas was project leader for the team that designed and developed one of the key solutions at SSF, Portugal's leading provider of business apps to the finance and credit industry. He holds a computer sciences degree from the University of Portugal, where he also earned a grant for research into artificial intelligence and also completed the International Executive Program at INSEAD.

Richard "Zippy" Grigonis recently spoke with Pedro Quintas about SIP and the 3G audio/video call center business.

RG: Does SIP function well in a mobile video/call center environment? Did you need to do any "special" modifications or extensions to SIP?

PQ: The option of going with SIP proved to be the right one, especially in a multimedia environment. Mobile video calls are just one example of the many rich call scenarios that are already appearing and will continue to evolve with the widespread adoption of IMS (IP Multimedia Subsystem) architectures and Instant Messaging. Another example already being launched in some countries is video sharing developed on top of an IMS architecture. In our implementations to date, we have been able to stay within the SIP standard, with the only special modification being the video fast update for improved video call picture quality.

RG: Is SIP really easier to work with than competing protocols such as H.323?

PQ: SIP is easier to work with, since it was inspired by web technologies, so it is easier to grab than H.323 by the developers. The other big advantage is interoperability, which with SIP becomes a reality, whilst with H.323 was a big problem. Our solution could have not reached the development stage it has today, in terms of fault-tolerance, multimedia features and interoperability, if it had been developed using H.323.

RG: As I understand it, COLLAB's call center system can be used by enterprises or it can be scaled up to be provided as a service by a carrier or service provider.

PQ: Yes. It is also quite flexible. Many of our customers use it as an audio call center, but it can be upgraded to a video call center quite easily. In fact, our system supports multiple channels of communication between a customer and contact center agent. And although it's a pre-IMS system, it provides many of the same features and is compatible with IMS.

The way people communicate is changing. Videocalls, emails, SMS, MMS, file sharing, web collaboration and presence information have become commonplace. Companies are switching to VoIP and telco operators continue to evolve to IMS architectures, enabling rich-call scenarios and presence awareness. The way companies communicate with their customers also continues to change and contact center architectures must evolve to meet these challenges. The industries most likely to spearhead the revolution are telcos, banking, healthcare, travel and entertainment. Medical screening, technical help-desk and product promotion will be among the successful applications. Current architectures which use CTI [Computer Telephony Integration] to integrate with proprietary voice systems cannot cope with such changes. Standards, such as SIP, have already emerged and continue to shake up the established industry landscape and value chain.

RG: So SIP helps with providing this flexibility or versatility?

PQ: Oh yes. SIP is the cornerstone standard of future IP Communications and helps make possible our [COLLAB's] One Contact. Basically, we offer an IP-based 3G call center software-only solution that offers full multimedia contact management. It seamlessly integrates video calls, enables multi-location contact centres in a distributed model and reduces deployment costs. We target 3G mobile operators willing to expand their 3G offer in Europe and Asia; NSPs [Network Service Providers] who may provide their corporate customers with the contact center infrastructure as a service in a hosted model, and contact center outsourcers who are doubtless eager to reap the benefits of costs reduction and flexibility in capacity and geography. Finally, enterprise contact centers can benefit from COLLAB too. We address the marketplace through regional and global distributors such as HP, and with our own salesforce.

It's quite an exciting business.