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TMC

INTERNET TELEPHONY®

The IP Communications Authority Since 1998™

Volume 10/Number 5 May 2007

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

By Richard "Zippy" Grigonis

Giving Birth —
and Other Strenuous Activities

Some impending births have got our attention these days.

First, that venerable maker of wireless and wireline hardware and software "building blocks," Aculab (news - alert) (http://www.aculab.com), has announced a bold new product line for the converged networks of both enterprises and service providers. Called ApplianX, it includes a range of voice and video IP gateways, SIP trunking gateways, media servers and related equipment. ApplianX (news - alert) is based upon Aculab's own innovative Prosody X technology, which itself was an editorial topic of mine.

The ApplianX hardware's form factor is the familiar 19-inch rackmount, which nestles quite comfortably in racks brimming with existing infrastructure equipment. Up to 4 E1 or T1 trunks can be housed in 1U "pizza box" sized-server and 80 trunks in a cPCI, carrier class chassis.

The first two ApplianX products to roll off the production line are the ApplianX IP Gateway and ApplianX VoiceXML Media Server. The ApplianX IP Gateway comes with 1, 2 or 4 E1 or T1 trunks that support various E1 and T1 protocols, plus the PBX integration protocols DPNSS and Q.SIG. The ApplianX VoiceXML Media Server is a standards-based media service platform for service center and enterprises that can be connected to either TDM or IP networks. The media server can execute applications written to VXML 2.0 and CCXML 1.0.

The ApplianX product line is notable in its use of scalable, high availability configurations, which makes it suitable for even the largest carrier class service providers and enterprises. Configuration and administration is done with an integrated HTML web server for remote configuration and monitoring, and there's support for the usual suspects of SNMP, RADIUS, CDRs [Call Detail Records] and event logging.

As for the second birth, we at TMC must confess that we're the mother, father and midwife! Last month, Rich Tehrani announced in these pages that TMC would be publishing a bold new magazine, *Unified Communications*. For those of us who remember the "killer app" hoopla of "unified messaging" back in the 1990s, it's gratifying to see that the technology has been quietly evolving (thanks in part to IP, more stable operating systems and cheap broadband connections) and has progressed to the point where the field deserves a magazine of its own.

But this will be a magazine unlike any other. Whereas the form factor of the ApplianX is the tried-and-true 19-inch rackmount, the "form factor" of *Unified Communications* will be, in the word's of the Monty Python troupe, "something completely different". That "something" takes into account the primacy of the digital and web incarnations of today's magazines. In fact, it's a "something" that may end up influencing the whole magazine industry.

I'm sure you're itching to learn more - and as we get closer to our July launch date, Yours Truly will reveal more details about TMC's latest and most exciting creation. IT

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

INTERNET TELEPHONY
The IP Communications Authority Since 1998™
Volume 10/ Number 5 May 2007

Cover Story

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INTERNET TELEPHONY® encourages readers to contact us with their questions, comments, and suggestions. Send e-mail (addresses above), or send ordinary mail. We reserve the right to edit letters for clarity and brevity. All submissions will be considered eligible for publication unless otherwise specified by the author.

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What's On TMCNET.com Right Now

To stay current and to keep up-to-date with all that's happening in the fast-paced world of IP telephony, just point your browser to http://www.tmcnet.com for all the latest news and analysis. With more than 16 million page views per month, translating into more than 1,000,000 visitors, TMCnet.com is where you need to be if you want to know what's happening in the world of VoIP.

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

The Benefits and Uses of Hosted Contact Center Solutions for Municipal Government
Municipal governments play an important role in the fulfillment and upkeep of their town's and citizens' needs. To maintain and ensure successful governing, it is essential that all interactions be as effective as possible.
http://www.tmcnet.com/657.1

Cable Operators: Do You Know Who Your Business-Serviceable Prospects Are?
How can cable operators more effectively sell their residential services if they knew - at the instant a customer call came in - whether that customer was serviceable. That way, customer service agents wouldn't spend valuable time pitching products their company couldn't deliver and the customer wouldn't walk away frustrated.
http://www.tmcnet.com/658.1

Security Practices for Enterprise Internet Telephony
Most enterprise VoIP traffic is transported over private IP networks. Many companies are excited by the prospect of carrying voice over the public Internet as well. Internet telephony promises numerous functional, global reach and economical benefits - but Internet traffic is also subject to a wide array of threats.
http://www.tmcnet.com/659.1

Asterisk on Apple TV Tutorial
The Asterisk 1.4.2 on AppleTV "soup to nuts" tutorial is courtesy of Steven Sokol from Sokol & Associates, Inc., who was the initiator behind this project when he proposed a "bounty" on loading Asterisk on Apple TV and which was successfully won by Jeff Gambera (aka l0rdr0ck).
http://www.tmcnet.com/660.1

Is John Chambers the Jack Welch of Tech?
Cisco's incredible strength comes from its ability to leverage its strong brand and enourmous salesforce to sell virtually every sort of product in telecom and datacom. This report makes it seem like the outlook for the competition is bleak.
http://www.tmcnet.com/661.1

TMC's Whitepapers of the Month
Visit TMCnet's Whitepaper Library (http://www.tmcnet.com/tmc/whitepapers), which provides a selection of in-depth information on relevant topics affecting the IP Communications industry. The library offers white papers, case studies, and other documents that are free to registered users.

Securely Enabling VoIP Remote Users The Need For a Comprehensive VoIP Security Solution
Enterprises of all sizes are deploying VoIP as a simple, cost-effective means to implement voice services across their organizations. However, the Internet is an insecure means of transmitting data because there are opportunities for modification and eavesdropping.
http://www.tmcnet.com/662.1

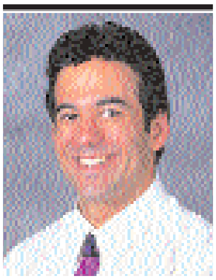
Update: Latest Lessons Learned from WiMAX Trials
WiMAX is short for Worldwide Interoperability for Microwave Access. It is a standards-based technique for delivering high-bandwidth connectivity over extended distances. Due to the rising interest in WiMAX as a networking technology, AT&T has conducted field trials, and gained valuable knowledge and insights that can help with network design, planning, and installation.
http://www.tmcnet.com/663.1

TMCnet's Channels and Global Online Communities provide the latest, most comprehensive news, analysis, and case studies for all your IP Communications needs.

TMCnet's Call Recording Channel
Call Recording is needed today more than ever due to increased legal requirements, security concerns, and training needs. SMBs and large enterprises both find themselves with call recording needs, and several companies are now focusing more attention on their needs - especially in the SMB space, which is receiving increased attention across the board. Visit the Call Recording Channel for the latest news, features, and case studies. Sponsored by Telrex.
http://www.tmcnet.com/channels/call-recording

TMCnet's Internet Fax Channel
Businesses are under ever increasing pressure to reduce costs, and one potential for doing just that lies in better control over their telecom expenses, which is why Telecom Expense Management (TEM) is gaining increased exposure. Visit the TEM channel for the latest stories, news, and interviews, explaining how your business can benefit from such a solution. Sponsored by Rivermine.
http://www.tmcnet.com/channels/telecom-expense-management/

By Rich Tehrani



Explosive Small Business Communications Growth

Software defined radio (SDR) systems are somewhat of a holy grail in technology as they use software to shift frequencies and modulation schemes while utilizing multipurpose underlying hardware. In a perfect world you could use software defined radio technology to receive cellular calls, WiFi, bluetooth, GPS, AM/FM, Sirius, XM, HDTV, etc.

Imagine if your smartphone used all the communications modes above with need for just a single processor - wouldn't that just be amazing? Of course this will likely be possible some day but for now there are hardware components which handle each of the above functions. In many cases multiple functions can be combined on a single integrated processor module or chipset.

If SDR was ubiquitous today we wouldn't worry much about the various new flavors of 802.11 such as A, G, N, etc… Why? Because our devices would all be software-upgradeable, allowing us to take advantage of the latest wireless standard without the need to forklift our existing access points and devices.

For wireless service providers the problem is even bigger as base stations are very expensive and forklifting a base station is a daunting prospect. But with newer and more efficient wireless technologies continuing to be developed how do you combat the ever-growing cost of throwing out the old ands supporting the new?

One way is to consider SDR for you base station via the technology being developed by Vanu Inc. (news - alert) The company was founded in 1998 by Dr. Vanu Bose the son of Dr. Amar Bose who made a name for himself in high-end audiophile products ranging from headphones to home theatre and commercial systems.

Vanu develops software-defined radio solutions for wireless service providers and similar to the world of HMP (Host Media Processing) you use the CPU as your DSP. For those of you who are familiar with the DSP resource board market you know companies like Dialogic and Aculab make HMP solutions today that once required proprietary boards.

But leaving HMP and heading back to the world of SDR, Vanu's Anywave Radio Access Network software runs on Linux boxes with Intel processors.

One wonders why there just aren't more companies in the SDR space. I would expect about 20 players to be pumping out products by now.

Vanu's company uses a single underlying software architecture to generate a variety of wireless communications modes.

One major benefit to a software approach based on off-the-shelf hardware is the dramatic decrease in cost achieved when compared to fixed function, proprietary systems. As a software house, Vanu can instantaneously take advantage of Moore's law without having to do additional development work.

This cost savings is passed along to the carrier and in addition the provider now knows they are able to upgrade the base station when needed to support yet another standard. So far Vanu has shown they can operate a combined GSM, CDMA and iDEN base station through SDR technology.

The benefits of working with Vanu Inc. seem to be endless from lower power consumption to lower cost to future-proofing your investment. Still a major service provider may not feel like they should bet the farm on a small company as they need to make sure their suppliers are around for the long haul.

But then again when you consider Vanu is a software company you begin to realize the company has much less at risk compared to typical hardware manufacturers.

Where SDRs are especially attractive is at a company like Sprint where they need multi-mode radio base stations. In addition the femtocell market could be another place where SDR makes a great deal of sense allowing an enterprise or home to support a number of different wireless standards.

The rural telco market is a further area where SDR can help providers deploy low-cost wireless networks. To that end, the Massachusetts-based SDR company announced recently it will partner with Globecomm Systems (news - alert) to provide turn key-based base station solutions while allowing the latter to focus on the hosted switching service.

One wonders why there just aren't more companies in the SDR space. I would expect about 20 players to be pumping out products by now. Is the technology too new for others to take the plunge? The concept has been around for fifteen years but perhaps CPUs have just recently become powerful enough to do a good job.

Could this be a technology that languished for a few years like VoIP (define - news - alert) and one vendor is needed to shake up the whole market. I don't hear too much about SDR from the major telecom equipment providers so I wonder if there is a downside I haven't considered.

Vanu seems to be acquiring customers in niche markets and one would imagine they are talks with the major players. I believe a single tier one service provider announcement is all that is needed to really make SDR a more popular term like FMC or IMS. Until then, we will have to focus on Vanu and the handful of other players in the space and wait for one of them to shake things up.

Software Defined Radio should certainly shake up the telecom landscape and make things interesting by, well, redefining the rules.

Redefining Dialogic

Speaking of redefinitions, the "new" Dialogic has taken on a more comprehensive technological persona.

The communications development world used to revolve around Dialogic. Ten years ago you virtually were compelled to buy a DSP resource board from this company or one of its smaller competitors if you wanted to develop an application such as unified messaging, voicemail, IVR, speech, recognition, ACD or just anything else.

In fact, for a number of years VoIP gateways were sold almost exclusively based on DSP resource boards. Larger telecom equipment providers would repackage systems with these boards through an integrator and claim the resulting gateways as their own.

I have taken many plant tours at industrial computer companies where I had to promise not to disclose the logos I saw throughout the plant. These were typically gateways under construction with boards from NMS or Dialogic.

In the late 1990s, while Dialogic was enjoying a nice time atop the enabling technology perch, the company received an offer from Intel it just couldn't refuse. The processor king had the hope of integrating Dialogic's core technology with Intel processors allowing HMP (Host Media Processing) solutions which were optimized for Intel's chips. In addition, Dialogic played a nice role as part of a growing communications division within Intel.

Last summer Dialogic (news - alert) was spun back out of Intel and was purchased by Eicon Networks who subsequently changed the name of the combined company to Dialogic.

The announcement was a source of major buzz at TMC's VoIP Developer Conference - now called Communications Developer (http://www.tmcnet.com/voipdeveloper). I was in a standing room only session as the news was presented to the anxious developers.

A great deal has changed over the years and perhaps most importantly you can now develop applications without the need for DSP resource boards.

VoIP has changed the way the communications development market works and now you can build voice applications without DSP resource boards or even HMP-based solutions.

But while voice development has become somewhat easier and less expensive to accomplish, video is still a different world and video developers still have to grapple with processor intensive applications which benefit from fixed function hardware and DSP resource boards.

This may explain Dialogic's CEO Nick Jensen's passion about video. In my frequent conversations with Nick, he exudes excitement as he talks about the opportunity in video. It isn't TV over the Internet that excites Nick, but bidirectional video streaming.

Nick tells me that 2G phones were used to send videos and pictures but 3G will take advantage of live video for gaming, video gaming and ringtones. He goes on to say that Singapore and Japan are way ahead in these areas and Europe is catching up. The U.S. is behind this curve still but will catch up eventually.

The way Nick sees it, video ringback tones will be hosted by a company for a fee and the videos will play based on the Caller ID of a caller. He sees the teen and consumer markets as the drivers for this sort of service.

From there his vision is that video greetings will become popular and a person - let's say a high-value customer - will be greeted with a customized voice or video greeting when they call. This is a way to make customers feel more welcome and indeed this is similar in concept to having a gracious host or hostess seat you at a restaurant.

He also sees video playing an important role in the future of dating and social networking sites. Moreover in the video space he sees the need for certain games such as poker, bridge and chess to have video support as people playing these games want to see each other.

He also sees a potential for video rooms on auction sites where you can see other bidders. In online virtual auction houses you will be able to simulate a live auction experience. Obviously there was some discussion here about the Skype acquisition by eBay.

Nick also believes the enterprise space will see video adoption and voice and speech applications will be upgraded to support video. Nick thinks people will want to work for companies who use video in this fashion and throw out the notion of synchronizing meeting minutes with one another. So he even sees video as a way to attract and retain talent!

When you are mobile you have things worth showing. A foreign airport, a theme park, a skyline, sunset, sunrise, animals, everything and anything. In fact I recently wrote a bit that touched on video touring as another area of potential growth. IT

Nick thinks that in about six months, Dialogic will be even further along in video - he even used the term "major player" in providing video building blocks to companies who in turn will be building tomorrow's leading edge video applications. It just so happens Nick will be keynoting Internet Telephony Conference & Expo in Los Angeles (http://www.tmcnet.com/voip/conference) in about five months. I invite you to register now for this event which takes place Sept 10-12 and make sure you are there to listen to Nick's evolving vision of the video opportunity in communications.

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ENTERPRISE NEWS

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http://www.tmcnet.com/605.1

Taridium (news- alert) Delivers Visual Mobile Voice Mail to IP PBX Customers

Taridium has added visual voice mail to its mobile interface, which allows users to access their office IP PBX from any number of devices - mobile device, home SIP phone, desktop IP phone, or even a TDM phone - and includes its unified voice mail solution.

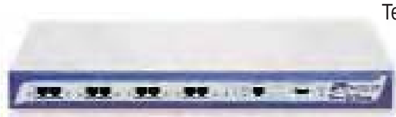
The visual voice mail component enables accessing voice mail through the web browser any mobile phone, including checking missed calls.

http://www.taridium.com

http://www.tmcnet.com/606.1

Quintum (news- alert) Addressses VoIP Continuity in PBX Branches

Quintum Technologies has announced that the survivability function, first introduced in the



Tenor AF and AX lines, is now available on the digital Tenor DX and BX VoIP MultiPath Switches. Quintum addressed the need to assure business continuity in branch office locations in the event that connectivity to the main IP PBX or IP Centrex is lost.

http://www.quintum.com

http://www.tmcnet.com/607.1

Small Businesses: An On-premises VoIP Solution for You

IPcelerate, (news- alert) provider of advanced VoIP products and technologies, has announced a new application suite designed specifically for the small business segment, called IPsmartSuite, which will address the particular needs of varying types of small businesses, at a price point that it affordable and with a solution that is manageable for companies between 5 and 50 employees.

http://www.ipcelerate.com

http://www.tmcnet.com/608.1

Streamline (news- alert) Hosts VoIP Service to Boston's SMBs Streamline Networks is offering a hosted VoIP communications solution to the SMB market throughout Massachusetts by using Natural Convergence's (news- alert) silhouette product. The company is also educating the SMB market on the simplicity of a hosted VoIP solution.

http://www.streamlinenet.com
http://www.naturalconvergence.com

http://www.tmcnet.com/609.1

Avistar (news- alert) Intros New Desktop Videoconferencing Software

Video collaboration provider, Avistar Communications announced the release of a new software solution, AvistarVOS v9.1 to provide users with increased deployment flexibility and simplicity. The new solution is aimed at making it easier to provide mobile workforces with the ability to utilize enhanced visual communications regardless of the endpoints in use.

http://www.avistar.com

http://www.tmcnet.com/610.1

RTX (news- alert) Adds Dual Line Capability and PBX-like Features to DUALphone 3088 for Skype RTX has made radical improvements to its DUALphone 3088 for Skype, adding support for three additional handsets as well as PBX-like functionality. In fact, the company claims the DUALphone 3088 is ideal for home-based businesses looking for an inexpensive phone system which enables both VoIP and traditional PSTN calling.

http://www.rtx.dk

http://www.tmcnet.com/611.1

Grandstream (news- alert) Unveils 6-line SIP Phone

Grandstream Networks has unveiled the GXP2020 enterprise SIP phone. The GXP2020 provides high voice clarity, a comprehensive set of advanced call features, multi-language support, security protection, automated provisioning and broad compatibility with leading SIP platforms.

http://www.grandstream.com

http://www.tmcnet.com/612.1

AirWalk (news- alert) Unveils CDMA Femtocell for Homes, Small Enterprises

3G IP radio access network company AirWalk Communications has unveiled a femtocell for CDMA voice and

data coverage in homes and small office buildings. The femtocell can be directly integrated into the operator's existing RAN via a broadband connection, thus delivering enhanced voice and data service without any hassle.

http://www.airwalkcom.com

http://www.tmcnet.com/613.1

Reignmaker (news- alert) Increases Hosted VoIP Presence

Atlanta-based Reignmaker Communications is one of the many hosted VoIP providers that has been making significant inroads since it made the commitment to enter the market back in 2003. Now, Reignmaker has acquired CyberSouth Networks and Swfttel Corp.

http://www.reignmaker.net

http://www.tmcnet.com/614.1

VoIP Phone Maker Snom's (news- alert) Products Now Interoperate with Patton's (news- alert) SIPxNano VoIP phone maker snom announced its products are now interoperable with Patton Electronics Company's SIPxNano - a tiny IP PBX designed for small businesses with under 30 extensions.

http://www.snom.com
http://www.patton.com

http://www.tmcnet.com/615.1

AuPix Releases New IP-based Desktop Videophone Platform

AuPix, (news- alert) which makes high-quality business videophones, has released a new videophone platform which is said to deliver better frame rates and improved video quality. The platform, based on Texas Instruments' DaVinci technology, represents a blending of DSP performance and SoC integration to deliver high-quality video to the desktop at affordable prices.

http://www.aupix.com
http://www.ti.com

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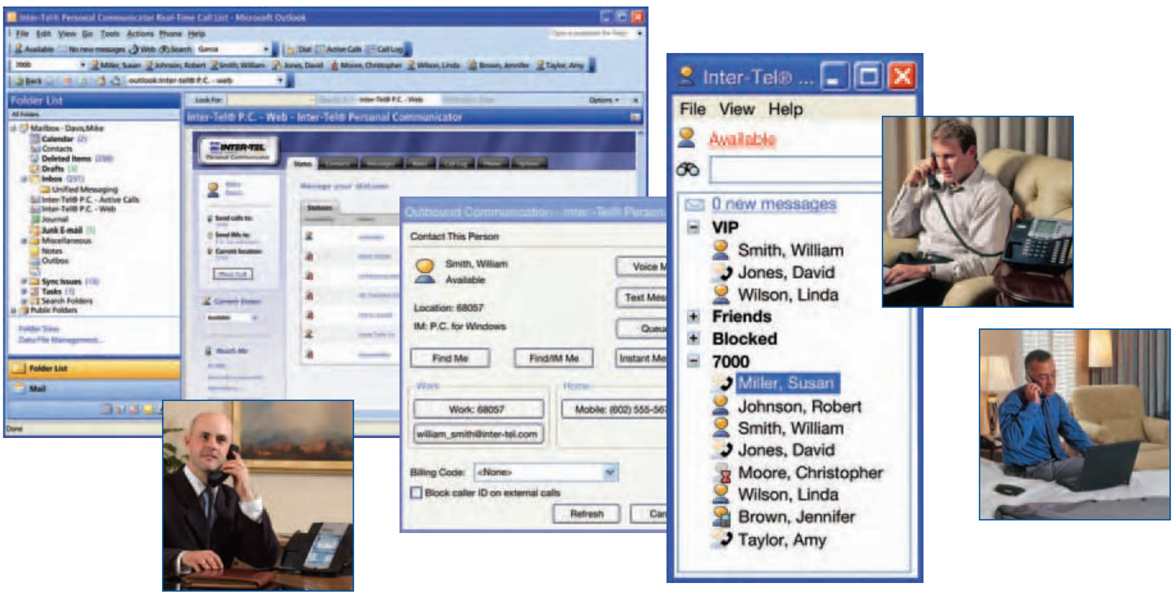

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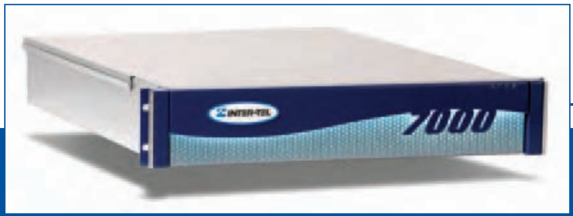

Always **ON**

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http://www.tmcnet.com/618.16

VoIP Peering Provider VoEX Builds Three New Interconnection Hubs

VoIP peering company VoEX (news - alert) continues to build out its global managed network. The company announced it has extended the network into three of the world's largest interconnection hubs: 60 Hudson in New York, Equinix in Chicago, and Telehouse in London, using switching solutions from Sonus Networks.
http://www.voex.com

http://www.tmcnet.com/619.1

Juniper Networks Enables Mobile Operators to Expand Service Offerings

A number of new features and enhancements have been introduced by Juniper Networks (quote - news - alert) for its Steel-Belted Radius SIM/AKA Server that enables mobile operators to reduce operational complexity and increase potential revenue opportunities. The new version is a key part of Juniper's Session and Resource Control portfolio and delivers SIGTRAN support.
http://www.juniper.net

http://www.tmcnet.com/620.1

Tatara and ip.access Partner on Femtocell-Based FMC Solution

Tatara Systems (news -alert) has been working on several new femtocell partnerships, among them ip.access, which develops femtocells (as well as picocells), and the newly agreed partnership between the two companies will integrate ip.access' (news -alert) Oyster 3G femtocell with Tatara's Mobile Services Convergence portfolio, offering mobile providers access to SIP-based IP networks to extend their coverage.

http://www.tatarasystems.com
http://www.ipaccess.com

http://www.tmcnet.com/621.1

M5 Takes Unified Messaging to the Next Level

Hosted VoIP provider M5 Networks (news - alert) has partnered with SimulScribe (news - alert) to offer M5Scribe, which offers customers the option of receiving transcribed voice mails via email, with an attached audio file, should they need to listen to the message at a later time. This option will allow executives, mobile employees, any users, in fact, additional options for checking voicemails, as the messages can be delivered to any email account.

http://www.m5net.com
http://www.simulscribe.com

http://www.tmcnet.com/622.1

WOW! Offers TV Caller ID: Converged Services Move Mainstream

In a bid to differentiate itself in the market, WOW! Internet, (news -alert) Cable and Phone has announced the deployment of Integra 5's (news -alert) new TV Caller ID application. With the new TV Caller ID application, the name and number of incoming calls are displayed right on a users TV screen before the phone even rings.

http://www.wowway.com
http://www.integra5.com

http://www.tmcnet.com/623.1

Personalize With Video Ringback Tones

With all the attention paid to video applications and with video capture enabled on many mobile devices, video mail was also only matter of time. Telenity, (news -alert) however, takes it all to a new level, combining the ringback tones and video with its video ringback tone service for 3G networks. Users can now define music videos, news clips, or personal video messages as their ringback tones.
http://www.telenity.com

http://www.tmcnet.com/624.1

Esnatech and Iwatsu Team Up on Unified Communications Solution for SMBs

SMBs using communications solutions from Esna Technologies (news - alert) and Iwatsu Voice Networks (news - alert) have gained a new level of integration. The two companies announced the merging of their Telephony Office-Linx (Esnatech) and Enterprise Communications Server (ICS, from Iwatsu) solutions into a single platform. The new Enterprise Communications Suite ties together telephony, mobility, messaging and enterprise-class presence in a system that is affordable for SMBs.
http://www.esnatech.com
http://www.iwatsu.com

http://www.tmcnet.com/625.1

DragonWave's Inexpensive Horizon Compact Wireless Carrier Ethernet System

(news - alert) Carriers wanting to deploy WiMAX 4G personal broadband solutions, or some other wireless next-gen service, but who are faced with a backhaul bandwidth problem, can now find relief in the form of a small (9-inch-square) indoor/outdoor wireless Carrier Ethernet backhaul platform called the Horizon Compact, from DragonWave.
http://www.dragonwaveinc.com

http://www.tmcnet.com/626.1

Cisco's Enhanced XR-12000 Router and Multi-Service Blade Brings Virtualization to Service Providers

Cisco Systems (quote - news -alert) has made some enhancements to its XR-12000 platform to enable more efficient delivery of resource-hungry services such as TelePresence. Cisco's virtualization and hardware upgrades will bring greater flexibility, quality and high availability to the world of both service providers and the customers they serve.
http://www.cisco.com

http://www.tmcnet.com/627.1

Sipera Enhances VoIP Security for IMS

Sipera Systems' (news -alert) IPCS 520, designed to protect both subscribers and the service provider's infrastructure from application-level attacks, now incorporates features to ensure secure deployment of dual mode phones and the associated infrastructures. Sipera combines various real-time security features, like anomaly detection, filtering, behavior learning, and verification, into one device.
http://www.sipera.com

http://www.tmcnet.com/628.1

UTStarcom Introduces GSM Capability to Its All-IP Wireless Network Platform

UTStarcom (news - alert) has introduced GSM capability on its MovingMedia 2000 end-to-end all-IP wireless network platform. MovingMedia 2000 enables mobile operators to offer CDMA-based voice and data services over IP to end users.

http://www.utstar.com

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Each NEWS item can be found in its entirety on TMCnet.
Point your browser to the URL above the story you wish to read.

http://www.tmcnet.com/592.1

Starent, Airspan Deploy Mobile WiMAX

Starent Networks (news - alert) and Airspan Networks (news - alert) will jointly deliver an end-to-end 16e Mobile WiMAX network solution. Starent's Access Service Network (ASN-GW) Gateway and Home Agent (HA) will be incorporated with Airspan's MIMO-based 802.16e-2005 HiperMAX and MicroMAX Base Stations for Mobile WiMAX network deployments.
http://www.starentnetworks.com
http://www.airspan.com

http://www.tmcnet.com/593.1

Microsoft Unveils Deepfish

Those faithful mobile users who dared to use the Internet on their mobile and agreed that they deserved a better treatment, now have their prayers answered. With Microsoft (quote - news -alert) plunging deep into the world of mobile Internet with a mobile web browser called Deepfish, surfing the Internet on your mobile may finally start being real fun.
http://www.microsoft.com

http://www.tmcnet.com/594.1

Google To Go? LG Electronics Pre-Installs Google Apps

Enhancing the fusion of mobile technology and Internet services, wireless handsets manufacturer LG Electronics (news - alert) and Internet giant Google (quote - news - alert) have inked a global partnership to pre-install all Google's services on one million LG mobile phones. Initial shipment to U.S., Europe, and Asia is expected by second quarter of this year.
http://www.lge.com
http://www.google.com

http://www.tmcnet.com/595.1

Websense Enhances Security for Wireless Internet Users

Websense Wireless, a division of Websense Inc., (news - alert) which provides Internet filtering and enterprise security software, has introduced the Websense

Wireless URL Categorization Engine. The new engine features a special ThreatSeeker technology that classifies URL's and filters them for blocking or allowing accesses.
http://www.websense.com

http://www.tmcnet.com/596.1

Radcom Unveils QTrace for 3G Mobile Net Operators

Network test and service monitoring solutions provider Radcom (news -alert) has introduced Qtrace to enhance its Omni-Q network and service quality management system. QTrace facilitates real-time troubleshooting and emergency call detection by enabling call searches across multiple sites.
http://www.radcom.com

http://www.tmcnet.com/597.1

Mobile VoIP Provider Truphone Letting US Customers Make Free Calls

(quote - news -alert) Mobile VoIP provider Truphone announced that U.S. customers with WiFi-ready Nokia smartphones will be able to make free international calls to landlines in 40 countries throughout April, May, and June. This is effectively an extension of the company's current price promotion (ending on March 31), during which customers have been able to make free mobile VoIP calls across the U.S. and Canada.
http://www.nokia.com

http://www.tmcnet.com/598.1

Intervoice (news - alert)Strengthens Unified Communications with Launch of Next-Generation Voice Portals

Imagine being able to use your web-enabled 3G mobile phone to browse and book an airline flight by viewing flight schedules and seating assignments on a touch screen and then clicking to choose a flight, speaking your seating selection and receiving a text message with the option to connect to a live agent to finalize your flight. This is what Intervoice is bringing forward in its latest step toward unified communications.
http://www.intervoice.com

http://www.tmcnet.com/599.1

BlueAnt Wireless Intros BlueAnt Supertooth Light

BlueAnt Wireless has introduced the BlueAnt (news - alert) Supertooth Light Bluetooth Speakerphone, the latest addition to its hands-free speakerphone technology offering stylish design, easy to use features and no installation requirements, including features like DSP for noise & echo cancellation, up to 15 hrs. talk time or 800 hrs. standby, full duplex and more.
http://www.blueantwireless.com

http://www.tmcnet.com/600.1

Jabra Unveils Convertible Bluetooth Stereo Headset

Verizon Wireless and Jabra (news - alert) have announced the Jabra BT8010 convertible Bluetooth stereo headset with a personalized phonebook and caller ID. The multi-functional Jabra BT8010 can be turned into a fully functional Bluetooth stereo headset for listening to music with the addition of a separately purchased stereo earpiece.
http://www.jabra.com

http://www.tmcnet.com/601.1

D-Link Boosts Performance for Wireless VoIP Users

D-Link (news - alert) has made available smaller form factor versions of its notebook ExpressCard adapters and desktop PCI Express adapters, both of which also incorporate the latest draft 802.11n technology. Both new products provide increased speeds to home or business wireless networks, while at the same time, extending the range of the solutions through the integration of wireless N standards.
http://www.dlink.com

http://www.tmcnet.com/602.1

AuthenTec Unveils Biometric Fingerprint Sensor

AuthenTec (news - alert) has launched the AES1710 biometric fingerprint sensor for the wireless market. According to the company, the fourth generation biometric fingerprint sensor meets critical performance standards necessary for widespread mobile commerce (M-commerce), security and convenience applications for full featured phones, and smart phones.
http://www.authentec.com

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DEVELOPER NEWS

http://www.tmcnet.com/578.1

Sangoma Offers Affordable T1/E1 Card

Sangoma Technologies (news - alert) announced the A101D single port E1/T1/PRI card with carrier-grade echo cancellation. Said David Mandelstam, CEO of Sangoma: “It is significant because, for the first time, medium size offices will be able to offer true carrier-grade voice quality. The smallest systems have had access to enhanced audio by using our A200D cards that provide hardware echo cancellation right down to two channels, while the larger systems needing two T1/E1 ports (46 to 60 voce channels) have had the same support using our A102D card.”

http://www.sangoma.com

http://www.tmcnet.com/579.1

Alcatel-Lucent Driving Fast Optical Nets

The recent OFC/NFOEC conference in Anaheim gave Alcatel-Lucent (news - alert) the stage to present several papers the telecom giant is conducting in an effort to develop future high-speed networks able to transmit voice, data, and video signals. The company envisions future optical networks transmitting higher and higher amounts of data - in many terabits per second - over optical fiber. This would make it possible, for instance, to transmit data from more than 600 DVDs per second.

http://www.alcatel-lucent.com

http://www.tmcnet.com/580.1

Dialogic PCIe Boards Certified by Intervoice

Recognizing the need to ensure its media boards are compatible with evolving technologies, Dialogic Corporation (news - alert) announced that its recently unveiled PCIe DMVB boards have now been certified for and are deployed in Intervoice’s Media Exchange Platform. The boards are now embedded in a key element of Intervoice’s (news - alert) marketing strategy.

http://www.dialogic.com
http://www.intervoice.com

http://www.tmcnet.com/581.1

Connect One Launches Secure Serial-to-Wi-Fi Module

Connect One (news - alert) has released Socket iWiFi, a secure serial-to-WiFi embedded device server that quickly and easily connects devices running machine-to-machine (M2M) applications to 802.11b/g wireless LANs and protects them from network attacks. The new device uses Connect One’s iChipSec CO711AG IP communication controller chip.

http://www.connectone.com

SIP NEWS

http://www.tmcnet.com/582.1

Micromethod Launches SIP Presence Server

Micromethod Technologies (news - alert) has launched the SIPoint server, comprising a SIP proxy, registrar, XCAP functions, and SIMPLE presence engine. By using SIP Instant Messaging and Presence Leverage Extension (SIMPLE) technology and providing a core set of routing, presence, and resource management functions, the SIPoint server can be applied to wide array of real-time VoIP communication applications and services.

http://www.micromethod.com

http://www.tmcnet.com/583.1

SIP-based IP PBX Demonstrates Compatibility with Microsoft’s UM Solution

Software-based IP PBX developer pbxnsip (news - alert) has announced its software has been proven interoperable with Microsoft Exchange 2007 Unified Messaging (UM) platform. Not only does this mean customers can connect to Microsoft’s platform, which dominates the business market - many PBX vendors require a gateway to connect with Exchange 2007 UM - but it also means that businesses can communicate using SIP connectivity, saving expensive network bandwidth.

http://www.pbxnsip.com

CHANNEL NEWS

http://www.tmcnet.com/584.1

AudioCodes Completes Purchase of CTI Squared

AudioCodes, (news - alert) a leading developer of Voice over Packet technologies and Voice Network products, has completed the acquisition of CTI Squared, (news - alert) a provider of enhanced data and voice messaging and communications platforms to service providers and enterprises.

http://www.audiocodes.com
http://www.cti2.com

http://www.tmcnet.com/585.1

Intelliverse Allies with WebEx, Jabra, and Creative on SMB Communications

As a part of its Talking Planet Business and callEverywhere hosted VoIP platforms Intelliverse (news - alert) will offer the WebEx (news - alert) WebOffice solution, Creative (news - alert) Webcams, and Jabra (news - alert) headsets. The company points out those SMBs rely on their communications networks to help them reach out and collaborate with remote locations to do business.

Leveraging the new partnerships, Intelliverse will enable SMBs to choose the components that work for their needs and keep them easily connected to their multiple offices, customers and partners—while giving them the capabilities and features of one large office, the compaies said.

http://www.intelliverse.com
http://www.webex.com
http://www.jabra.com
http://www.creative.com

http://www.tmcnet.com/586.1

Sonus Networks Acquires Zynetix to Expand Wireless Portfolio

Sonus Networks (news - alert) has announced a definitive agreement to acquire privately held Zynetix Ltd., designers of GSM infrastructure solutions. According to Sonus, integrating Zynetix technology with its IMS portfolio is a critical step in developing Sonus’ wireless portfolio of 3G and GSM next generation solutions addressing the demand for carrier-class, end-to-end IP based networks. Sonus is seeking to accelerate the wireless IP-Voice market as part of its long-term growth strategy.

http://www.sonusnet.com
http://www.zynetix.com

http://www.tmcnet.com/587.1

Cisco Acquires WebEx to Extend Reach in SMB Market

Cisco Systems (quote - news - alert) has acquired WebEx, (news - alert) a supplier of on-demand collaboration applications, as the network company seeks to extend its vision for Unified Communications, particularly within the SMB market.

http://www.cisco.com
http://www.webex.com

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http://www.tmcnet.com/588.1

Five9 Intros On-Demand Call Center Packages for Small Businesses

Five9, (news - alert) a provider of on-demand call center solutions, has introduced two call center packages designed for small businesses. The Five9 Call Center Suite Small Business Edition, which includes the Small Business Edition and Five9 Inbound Call Center are aimed at providing small businesses with an affordable way to significantly boost both sales and customer service levels.
http://www.five9.com

IP CONTACT CENTER NEWS

http://www.tmcnet.com/589.1

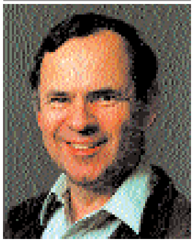
Altitude Software to Unveil New Contact Center Solution A new contact center solution aimed to boost a center's ability to embrace industry-wide change is being announced by Altitude Software, (news - alert) a provider of independent contact center solutions. The Altitude uCI 7.5 software suite includes Altitude IP Contact Center functionality and streamlines IT investments and optimizes human resources to provide businesses with a SIP-based multimedia contact center solution.
http://www.altitude.com

http://www.tmcnet.com/590.1

Intervoice and RSA Improve Telephone Banking

In a bid to assist financial institutions' contact centers with the need for multifactor authentication for their telephone banking environments as well as provide enhanced functionalities like speech technologies, Intervoice and RSA, (news - alert) The Security Division of EMC, have announced an agreement to integrate their offerings.
http://www.rsa.com
http://www.intervoice.com

By Tony Rybczynski



P is for People in Telephony

IP Telephony is inevitable, but just because you can do something, should you?

Successful deployment of IP Telephony starts with defining your business goals, and establishing a cross-functional team which is aligned with these goals. If you think of IP Telephony as dial tone replacement or a voice packet as just another packet, you will likely be unsuccessful in both business and technology senses. If you think of IP Telephony as a real-time application with users who have very high expectations for performance, reliability and security, you are on the right path. Given that IP Telephony brings together end user service functionality and IP networking, organizational and skill set convergence is very important to successful deployment of IP Telephony.

The End User is King

Some enterprises jump into IP Telephony without addressing its realities. Learning on the job may move the yards, but could just as well result in cost overruns due to poor planning, failed implementation and service disruptions, and dissatisfied users. The old adage that "nobody even got fired for going with a leading vendor" doesn't apply to IP Telephony, particularly if the dissatisfied user is a business decision maker or CXO.

Getting everyone on the same page with an IP Telephony initiative starts with an understanding that a key metric is user satisfaction. The user in this case could be the end user him or herself or the business owner (e.g. a hotel manager), who wants calls to be handled in a manner that is consistent with business processes and operation.

A systematic approach is to address the needs for specialization in IP Telephony across various parts of the organization, recognizing that smaller enterprises may not have dedicated personnel to provide these functions.

Not surprisingly, it starts with the user perspective. The enterprise needs to establish a voice application specialist, accountable for ensuring that user requirements are met, training needs are fulfilled and the customer's voice is heard. This function would ensure that the required feature/functionality (including numbering plan, call coverage and attendant functionality) is delivered, not just when deployment is complete but also during the transition. It would also ensure that business needs are addressed, including regulatory and business continuity. Added responsibilities include setting up pilots, program tracking and communications with client groups, as well as customer satisfaction tracking, Wired and wireless telephones and mobile and desktop clients must be matched to user needs, including desktop convergence across telephony and PC environments. Consideration must also be given to ancillary devices such as voice recording, transcription devices, kiosks, facsimile, headsets, and alarm lines. Finally, the voice application specialist should work with users to determine which display-assisted applications (e.g. directories, conference call management and various forms of alerts) should be provided on IP phones.

The network engineering group expertise needs to be extended to cover IP Telephony Communications Servers, the voicemail or unified messaging servers, and contact center servers; as well as of gateways to the public network. WAN services including CO trunks (e.g. based on PRI) need to be engineered to ensure the user Quality of Experience is met, SLAs need to be managed across virtual private networks and billing accuracy needs to be verified. Other new responsibilities include VLAN administration, and ascertaining PoE and UPS requirements in wiring closets (for powering of IP sets even under failure conditions).

The operational group likewise is impacted by convergence. This starts with being a key stakeholder in the roll-out and planning of IP Telephony, providing a valuable perspective on testing, diagnostic and operational requirements. This not only includes the running of the converged infrastructure but also supporting help desk services for lines of business, end users of IP desktop and WLAN phones and of soft clients on PCs, laptops, tablets and various mobile platforms. Another important dimension is managing voice carrier solicitations, contracts, departmental charge back and toll fraud prevention.

A Workplace Vision Enabled Through a Technology Vision

Successful execution of an enterprise IP Telephony strategy depends on people, in two dimensions. The IT organization needs to take full advantage of networking skills in the traditional data group and end user understanding in the traditional telephony group. This is a significant cultural shift that drives the establishment of trust across two previously silo'ed groups, which had not worked together, while enabling new career development paths for employees. Furthermore, the converged IT organization needs to engage the client groups and lines of business that will be the direct beneficiaries of IP Telephony. After all, with IP Telephony and more generally Unified Communications, a technology vision needs a complementary workplace vision that embraces mobility and productivity. IT

Tony Rybczynski is Director of Strategic Enterprise Technologies at Nortel. (quote - news - alert) He has over 20 years experience in the application of packet network technology. For more information, please visit http://www.nortel.com.

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By Brough Turner

Why There's No Internet QoS and Likely Never Will Be

OK, I got your attention. Of course, service level agreements are widely available and that's one form of quality of service (QoS). But the popular meaning is certain packets get priority over other packets and there is no such prioritization on the Internet backbone and very little anywhere in the Internet.

QoS at the Core of the Internet

Prioritization only matters when links are saturated. Once you get beyond the access network, every link in the Internet - local, regional, national or international - is carrying multiplexed traffic from many, many users. Multiplexing many, bursty flows results in relatively predictable traffic. Traffic volumes vary by time of day, but links don't saturate, except as a result of poor engineering or other link failures. Either case generates a rapid response from any ISP that expects to remain in business.

In short, except at the edges; i.e., the access network, Internet links may be heavily loaded but are not saturated. Combined with relatively high capacity links, this means typical delay variations are fractions of a millisecond and packet loss is negligible; i.e., best effort is good enough even for low latency applications like voice telephony. Except during major failures, the effect of QoS in the Internet backbone is negligible.

Consumers won't pay a premium for performance improvements they can't see! They might be induced to pay for a brand (after all, people pay premium prices for branded water), but as yet no such brand has emerged. And if one does, consumers will be paying for that brand, not for QoS technology per se.

Broadband Access Links

The one place in the public Internet where limited, highly specific QoS measures make sense is at the consumer end of an asymmetric broadband access link. Typical residential DSL connections offer a few megabits per second (or less) to the home, but only a few hundred Kbps from the home to the Internet. Unlike links between core routers, traffic on such access links is very bursty and bursts can saturate the link. If there are no active peer-to-peer applications then, most of the time, little or no traffic flows on most residential broadband connections. Suddenly, someone sends an email with an attached file or photo which saturates the outgoing link for many seconds as several megabytes of data go out at perhaps 250 kilobits (~31 kilobytes) per second.

Individuals can benefit from simple priority queuing at their end of a broadband access link.

Typical residential access links are narrow pipes where the cost of purchasing more capacity is prohibitive, if it's possible at all. On the other hand, we can control the routing policy for what goes out on the link by deploying an appropriate residential router at our end of the broadband access link. As a first approximation, one would like to give priority to VoIP packets (and gamers may want to give priority to specific multi-player games). Simple priority is a good first step, but may not be enough on slow links.

Slow links have an added problem due to large packet serialization delay. VoIP packets are typically less that 150 bytes while a web page or email is typically delivered in ~1500 byte chunks. At 250 Kbps, a 1500 byte packet takes ~50 milliseconds to pass over the link. If a VoIP packet arrives just after a 1500 byte packet has started, it doesn't matter that the VoIP packet is priority to be sent next, it will have to wait for the current packet to complete. Intermittent 50 ms delays are handled by a jitter buffer at the other end of the VoIP connection, but only at the expense of an additional 50 ms of delay. If the uplink is slower than 250 Kbps, serialization delays are even longer.

Luckily this is one place where priorities work and can be imposed. Indeed most consumer VoIP devices incorporate simple priority and some include the ability to fragment large packets (so as to reduce serialization delays). And, because it's useful for both VoIP and gaming, this functionality is showing up in popular residential routers from Linksys, Netgear and the like.

Brands Can Command a Premium, But Internet QoS Never Will

Individuals can benefit from simple priority queuing at their end of a broadband access link. But they are not going to pay for benefits they can't see, so we're unlikely to see prioritization from ISPs. Operators interested in premium services should focus on branding and perhaps on facilitating simple priority queuing in the access network. IT

Brough Turner is Senior VP of Technology, CTO and Co-Founder of NMS Communications. (news - alert) For more information, please visit the company online at http://www.nmscommunications.com.

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is for **Technology**
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By Marc Robins

GotVoice Juices the Voice Mail Marketplace

Just when people started dissing voice mail, saying how they don't check their messages all that often and you should just email them (or SMS or IM them) if you want them to get back to you - and this from people in the "voice" business, including some editors-in-chief of voice-centric technology trade publications - along comes a company that helps to breathe new life into the space.

GotVoice (news - alert) (http://www.gotvoice.com) - a company launched in 2003 but just coming out the gate with its service (it needed several years of development to make it so seamlessly transparent and easy to use) - has come up with a new free voice messaging service that allows consumers to manage their mobile, home and work voice messages online, from a single, unified web interface.

Actually, such changes have been afoot in the enterprise voice messaging space for some time now, thanks to the slow but steady burn under unified messaging technology. But I can't tell you how many times in years past I've heard people say, "This is the Year of Unified Messaging!" - only to be quite underwhelmed by the offerings or the tepid adoption by the marketplace of even well-developed enterprise solutions. Problem was, businesses didn't want to pay what the vendors were asking, and the benefits were a bit too fuzzy for many.

GotVoice, however, has the potential to popularize so-called unified messaging to the masses - and truly invigorate the market for all types of UM solutions - including those of the Enterprise ilk. The developers hail from RealNetworks, Microsoft, Starwave, AT&T, and Corbis, and really seem to have come up with an easy, yet elegant mash up of the Web and voice messaging.

The company's "secret sauce" - a competitive advantage that will be hard for others to replicate in a short time - is that it spent four years learning how to seamlessly integrate with all major cell phone carriers, residential lines including those offered by cable and VoIP providers, and corporate voicemail systems - with no carrier or work-place involvement.

GotVoice combines the ease of email with voice messaging, and the company claims that it is the only messaging service that allows users to visually take control of their voicemail without requiring a new number, a forwarded call, or text messaging services.

In fact, the service doesn't require users to make any changes whatsoever to their normal phone use. To provide the highest level of flexibility, GotVoice leaves messages on the original voicemail system, so users can check their voicemail as they always have, if they choose. Whether users opt to receive voicemail via email or at their personal GotVoice Web page, GotVoice enables them to easily send, receive, create and store all of their voice messages-in the same way they use email. There is no software to install and users don't even need a phone to access messages.

GotVoice service highlights include:

Access Voicemail Anytime, Anywhere: GotVoice checks multiple voicemail boxes, such as home, mobile and work boxes, up to 24 times a day, then delivers messages to a user's email or makes them accessible at the user's personal GotVoice Web page. Users can also create and send messages using their PC or phone.

Send, Receive and Create Voice Messages without a Phone: GotVoice allows users to send, receive, and create voice messages without the need for a phone. Users can create messages with their voice or use a quick text-to-speech feature.

Organize, Save, Forward : GotVoice makes it easy to organize, save and forward important messages, just as with email. It's easy to sort through important messages and forward them to multiple recipients all at once, without additional call forwarding charges.

Silent Sending: GotVoice gives users the option of sending voice messages without causing the recipients phone to ring - great for late night messages, traveling in a different time zone, or moments when you just don't want to talk.

One Message, Many Recipients: Simply record a message using your PC or phone, and distribute it to up to 200 contacts.

Personalization: GotVoice makes it easy to create personalized greetings and outgoing messages. Add your favorite MP3 music to your voicemail greeting, choose a custom celebrity voice greeting or select background tracks from an audio library.

The Fun Factor: GotVoice reminds users that voicemail can also be fun. Create your own message by recording your voice over your own music. Users can also forward favorite or funny voicemails to friends, while preserving precious messages from loved ones. IT

Marc is Chief Evangelism Officer of RCG (Robins Consulting Group), a leading marketing, communications and business development consulting firm 100% dedicated to the IP Communications industry. For more information about RCG, visit http://www.robinsconsult.com, email marc@robinsconsult.com, or call 718-548-7245.

SMS May Be Your SOS

By Kelly Anderson

There is a silent enemy of personalized services, multiplexed service bundles, and the consolidation of multiple providers. This isn't the risks anyone speaks about on CNN and it is given little if no air time by Wall Street Analysts. If it is even mentioned by a company it is a footnote in a quarterly report. Unfortunately, with the little attention it receives, it will prove to be one of the most important and successful indicators of whether consumers can trust the company and will accept them as their ongoing service provider.

Subscriber Management (or SMS) is defined as a combination of hardware, software, and human interactions that help organize and operate a company's business - but it is far more than that. The SMS contains all customer relevant information and is responsible for keeping track of placed orders, credit limits, invoicing and payments, as well as reporting reports and statistics. This has been an increasingly difficult task for operators.

SMS has faced many new challenges in the new communications market. Engineering and process challenges have plagued the efficient management of subscriber information. This trend is caused by subscribers picking up more and varied services from one provider that are offered through varied networks, and also by the recent mergers. Large-scale migrations of customer data are major operational projects in the aftermath of the merger. Not only does the difficulty begin with merging together two formats from disparate systems and network control tools, but also the synchronization of different data elements and incompatible detail of data - causing more than a few headaches and long hours for IT and engineering departments through the merger process.

I happen to know of a true life consumer story of what a merger looks like when systems and networks are not in synch. My friend Brenda was a victim of the one of the major wireless mergers last year. She was perfectly happy with her service when the merger happened (actually it was about six months later when the name on the bill changed and they started answering the phone differently). Her service consisted of a four phone family plan that made her average bill about $180 a month. Her teenage daughter asked for one of the new hot pink phones for her birthday. The specific phone she wanted showed up on the website and Brenda went to order it. Much to her dismay, the phone was not compatible with her service (signaling issue), and, she couldn't even switch her plan over to that service type because she had to "live" out the terms of the contract (which was another year) with the current service. Not only was her data not in the database of the customer representative she called, but her contract was not even transferable. Needless to say, after the year left on her contract, she switched to another provider.

Creating a data plan and implementing that using a flexible and accommodating protocol is a winning strategy.

A large-scale study was released in 2004 using the American Customer Satisfaction Index, and stated that the 28 large-scale mergers that happened between 1997 and 2002 were unsatisfactory to 50 percent of their customers. Even after two years, customers that hadn't left were still significantly more dissatisfied than they were with the previous company.

I have a feeling that the wide-range marketing plans for mergers and new product timelines and not going to wait for IT tasks like the migration and monitoring of data, network devices, and systems. So is there an industry plan around the alleviation of issues surrounding the monitoring and service issues surrounding multi-faceted networks, large-scale mergers, and the mismatch of data new, disparate products?

There are several industry groups concerned with the topic of diverse networks, user experience, and customer impacts to mergers. This is where the solution to this industry-wide problem could be alleviated by using standards. One of the things IPDR does is to create consistency in getting data from the network in the required time, provide that data to whatever end system need for that data downstream. Data is such a crucial issue for today's operators that have complex IP-based services, so putting the joining two companies with different monitoring needs on top of that makes a lot of sense. The seamless (at least to the consumer) migration of networks needs to allow a consolidated view of data and consumer usage and behavior. Creating a data plan and implementing that using a flexible and accommodating protocol is a winning strategy. As president of IPDR.org, I invite providers who provide complex services to take a look at the IPDR Service Specification and protocol (http://www.ipdr.org). It may just make a future decision to merge a lifesaver for your customers. IT

Kelly Anderson is President and COO of IPDR.org, a collaborative industry consortium focused on developing and driving the adoption of next-gen service usage exchange standards worldwide.

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By Hunter Newby

Cutting Out the Middleman

Arbinet, "the leading provider of solutions to simplify the exchange of digital communications," has taken a step into the turbulent ocean of business-model disruption with its recent announcement of PeeringSolutions for the U.S. domestic market. Arbinet solutions "simplify the exchange of digital communications in a converging world". This does not specifically mention, or limit them only to voice, but their current primary focus is the exchange, transaction management and managed services which streamline performance and improve profitability of voice services for Members. Those members include, Telecom Argentina, Telefonica Argentina, Columbus Communications Jamaica, Telefonica Chile, Telefonica Peru, Telmex Chile, and Verizon Dominicana to name a few. The sea-change from arbitraging minutes to peering endpoints may be gradual, but the direction is polar.

As their press release explains, "PeeringSolutions allows U.S. Domestic carriers, including competitive local exchange carriers (CLECs), cable and mobile operators, to peer and exchange traffic with each other regardless of their network technology." This in and of itself is a great service because translations and trans-coding will haunt us all for as long as there are competing vendors and technologies. Beyond this initial level of mediation the service goes to the root of calls by performing number matching. As the release states, "Arbinet queries each inbound call attempt against a database supported by the SPIDER Registry and sends routable calls via its switches to terminating Members based on instructions from this database. Settlement for these calls is determined and agreed upon by the providers, with a variety of settlement options including traditional paid settlement and bill and keep agreements, allowing carriers to maintain full control of the economics of their business." This type of call rating is also known in the industry as "seller-sets" and basically uses ENUM in a Least Cost Routing (LCR) fashion.

There are some real benefits and interesting dimensions to this new offering that essentially get summed up in a quote from the original press release that was provided by a member of the executive team at PaeTec.

"Joe Ambersley, Executive Vice President, PAETEC Communications (news - alert) Vice Chair/Treasurer of COMPTEL commented, "PeeringSolutions gives a competitive carrier a simple way to establish and manage paid peering relationships with other carriers in the U.S. and internationally. The carrier maintains control of its payment terms for terminating minutes and benefits from the high call quality generated by direct routing to and from other paid peering members. In addition, the competitive carrier eliminates intermediary carrier termination fees, improving its cost structure. PeeringSolutions also eliminates expenses associated with carrier access billing (CABS) and provides interoperability between TDM and VoIP networks, delivering the benefits of VoIP peering today without requiring costly network upgrades."

The first thing that stands out is that this service offering is "paid-peering". This is interesting in terms of its efficiency and the economics of minutes. Since the Arbinet Members are largely in the business of generating revenue from minutes they continue to look for ways to protect that revenue and eliminate costs from the network to increase the overall margins. This makes a lot of logical sense.

Peering implies a free exchange of traffic. This assumption comes from the legacy definition of peering in the ISP sense that still exists today. The exchange of IP traffic over a common Ethernet switch fabric is multi-lateral, or open to all and settlement free. There is no cost from one network to the other for transit. Paid-Peering also exists in the ISP world. The technical exchange doesn't change, but the economics do. Paid-peering is less expensive than transit, but more than free.

There are multi-lateral voice peering services that exist today as well and they facilitate a completely free exchange of voice traffic between the endpoints of the network operator participants. This is essentially an extreme position to take from the perspective of some CLECs and other endpoint owners that still make money on minutes, and the Arbinet service helps them ease into the future by allowing them to set a rate to terminate a call to any one of their endpoints. One technical benefit of this functionality, as Joe Ambersley stated, is the high call quality from direct call routing. Here is where the business model disruption comes in and it gets really interesting.

If those service providers that possess endpoints all begin to contribute their numbers and query the Registry for outbound calls, at some point a majority of all calls between "minutes" carriers becomes direct and essentially on-net to each other. Clearly there are quality benefits to this, but what happens to the IXCs, the "long distance arbitragers" that Arbinet's business was founded on? They get cut out. It's just like how transit providers get cut out when ISPs and other IP networks peer directly. Whether it is paid-peering, or true multi-lateral, those that control the endpoints are the "landowners" essentially and the middlemen go away.

This technology and overall industry change comes as no surprise to Arbinet and as Cliff Radziewicz, Vice President Business Development for Arbinet stated, "We are proactively addressing this business model sea-change and therefore will be able to better control our destiny. Most service providers, regardless of type, will still need help with TDM/VoIP conversion, routing decision tools, interconnection infrastructure, and settlement during and after the transition to a peering world, and Arbinet will be ready to serve them." Where that leaves Arbinet with their existing customer base depends largely on the speed at which the new service develops. The faster "all" the numbers and associated positive hit rate increases in the registry the faster the middlemen IXC's decline. This is not to say of course that all endpoints will become on-net in the future, so there will always be the need for IXCs in some form, but it does prove that owning the number, or identifier, for the endpoint on a network is the value and power for operators going forward. IT

Hunter Newby is chief strategy officer for telx. (news - alert) For more information, please visit the company online at http://www.telx.com.

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COMMUNICATIONS

By Steven Johnson

SIP Trunking: Is It Right for Your Enterprise?

SIP trunks offer businesses a unique opportunity to rapidly reduce their communications expenses, while leveraging the benefits of IP communications. They create one converged network for both data and voice, increasing a business' flexibility and future-proofing against an ever-changing communications environment.

A SIP trunk is the use of SIP to set up communications over the Internet between a customer location and an Internet Telephony Service Provider (ITSP) which transfers the SIP calls to the PSTN. Unlike in traditional telephony, where bundles of physical wires were once delivered from the service provider to a business, a SIP trunk is a logical connection from one point to another over the public Internet.

The Benefits of SIP Trunking

SIP trunking eliminates the need to purchase BRIs (Basic Rate Interfaces), PRIs (Primary Rate Interfaces) and local PSTN gateways. This means that capacity can be increased without the expense of adding another full E1 or T1 and without upgrading the hardware to support the additional connections.

SIP trunking also reduces costs by eliminating the need for separate voice and data connections, and expands the potential for communications convergence using both voice and data together.

A further advantage is that by outsourcing PSTN connectivity calls may be terminated close to the called party, resulting in reduced long distance charges.

The Need for Standards

SIP is the standard of choice for IP communications today and has been adopted by the major PBX and phone vendors and service providers. Connecting to a service provider network is further improved by use of the SIPconnect(SM) standard. This set of best practices was developed by the SIP Forum to ensure a seamless connection between the enterprise and the service provider. SIPconnect meets a critical need by eliminating many of the unknowns and incompatibilities of interfacing two networks for delivering quality voice and other services.

SIP trunking... [eliminates] the need to purchase BRIs, PRIs or PSTN gateways.

Infrastructure

A relatively small investment is necessary to benefit from the use of SIP-based communications and to enjoy the cost savings that accrue from using SIP trunks. For the enterprise, converting to VoIP usually involves the purchase of an IP-PBX, IP phones or soft clients, and a SIP-aware firewall or edge device to maintain security while admitting VoIP traffic.

Delivery And Quality of Service

Although the IP pipeline can carry much more traffic than a traditional connection, it is important to employ proper quality of service (QoS). Voice and video are very susceptible to delay, which means that some QoS procedures should be in place to guarantee priority delivery of these packets vs. other information downloaded to the converged network.

Despite some perceptions to the contrary, the core network of the Internet is often not a bottleneck today. The last mile and the customer network can be. With the right QoS prioritization and admission control at the enterprise edge, this issue can be eliminated.

Security

Using a firewall that's specifically designed to handle SIP communications will provide the best defense against unwanted activity. Full SIP proxy technology allows for advanced filtering, verification and routing, as well as dynamic control of the opening and closing of media ports. Some products offer encryption of the signaling using Transport Layer Security (TLS) and of the media using Secure RTP (SRTP) or other algorithms.

Authentication with the service provider is also critical. While some IP-PBX equipment can support this natively, others cannot. A full SIP proxy firewall or other edge device may offer this capability as well.

Another security-related issue is redundancy. A fully SIP-capable firewall or customer premises device can provide a robust system for securing full VoIP redundancy, as traffic can be routed to a back-up carrier if the primary carrier is unavailable.

Summary

SIP trunking offers enterprises the benefits of converged communications and saves substantial expense by eliminating the need to purchase BRIs, PRIs or PSTN gateways. It also reduces expenses by terminating calls closer to the called party. A robust enterprise solution combined with a SIP trunk from an ITSP realizes the promise of safe global connectivity, over the Internet, so long envisioned by the voice over IP pioneers. IT

Steven Johnson is President of Ingate Systems. (news - alert)
For more information, visit the company online at
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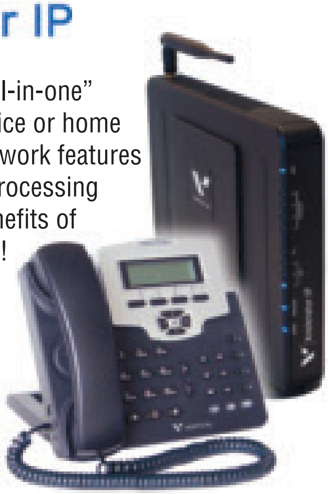

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Will Somebody Please Fix Intercarrier Compensation

By John Cimko

The FCC has been trying to do something about intercarrier compensation (IC) for at least six years. Progress has been slow. Almost everybody agrees that the system is broken. But nobody can agree on the best way to fix it.

What exactly is the intercarrier compensation problem? And why is the issue important to VoIP providers and their customers?

Intercarrier compensation is a system by which carriers pay each other for use of their respective networks and facilities to originate and terminate communications. IC princi-pally consists of access charges (payments from long distance carriers to local carriers) and reciprocal compensation (payments between local carriers). Before the AT&T dives-titure, most of these cash flows took place within the Bell System. Regulators and carriers fashioned rate structures that included long distance rates that subsidized local exchange service to subscribers in rural and other high-cost areas. After divestiture, the FCC adopted an access charge system that replaced some of the Bell System's internal cash flows, and that continued subsidies for local exchange services.

The problem with intercarrier compensation is that its subsidies and rate structures have not worked well in a competitive environment marked by the rapid growth of new tech-nologies. IC rates are uneconomic because they're generally based on embedded costs, instead of forward looking costs that would better promote efficient provision of service. Worse, the rates are set at different levels, depending on the type of traffic being carried, the type of carrier handling the traffic, and the geographical jurisdictions where that traf-fic originates and terminates. These variations in rates for essentially the same service have distorted the market, hampered competition, and slowed the adoption and deploy-ment of new technologies.

Almost all stakeholders agree that the current IC system cannot be sustained much longer. But most of these stakeholders are fighting over the best way to fix the problem. A key element of the dispute involves whether the current system, which is grounded in the circuit-based public switched telecommunications network, should be preserved with some modifications to make rates more uniform across services, carriers, and jurisdic-tions, or whether more radical steps should be taken in order to embrace the movement toward IP-based telecommunications networks.

Some industry players in the former camp argue that the IC system should be revised to fix rate inequities and thus mitigate arbitrage problems, but that termination rates (to-gether with subscriber line charges and universal service support) must be used to protect the revenue streams of high-cost incumbent local carriers. They argue that any changes to the current system that would interfere with these carriers' recoupment of investments in expensive infrastructure would threaten universal service to rural and other high-cost cus-tomers.

Proponents of a more sweeping overhaul favor a "bill and keep" system in which all car-riers would recover their network costs from their own customers. Carriers terminating calls originated by other carriers would no longer receive compensation from those carri-ers. Advocates of this approach argue that the current system does not give local carriers any incentive to reduce costs and bring operating efficiencies to their networks, because they don't have to rely on their own customers to recover their network costs associated with terminating traffic.

The most recent IC reform plan submitted to the FCC - the "Missoula Plan," so named because meetings to develop the plan were held in Missoula, Montana - purports to bridge the gap between these two camps by moving toward a more unified rate structure, but also by setting up three rate tiers for small, medium, and large local carriers. But crit-ics of the Plan argue that it fails to overhaul market-distorting universal service subsidies and, according to Vonage, fails to eliminate the "disparate, asymmetric rate regimes for compensation related to traffic termination."

What does all this mean for VoIP? So far, although a recent FCC decision (which has been appealed) requires that interconnected VoIP providers must contribute to the federal universal service fund, VoIP has not been directly affected by the IC system. This is because current FCC rules exempt IP-enabled service providers from paying access charges and, as the VON Coalition has explained in FCC pleadings, VoIP providers are permitted to originate and terminate calls as business providers "or through the use of local carriers, paying cost based rates based on reciprocal compensation."

But all this may change if the FCC adopts a new system that applies to interconnected VoIP providers. The agency is focusing on three goals for IC reform. The new rules should promote economic efficiency, which is necessary to advance competition and new technologies. Also, IC reform must preserve universal service, enabling carriers to con-tinue serving high-cost customers and to expand into unserved or underserved areas. Fi-nally, artificial regulatory distinctions between technology platforms, categories of pro-viders, and different types of traffic must be eliminated. Failure to achieve this goal would risk retaining discriminatory aspects of the present system, thus hamstringing competition and harming consumers.

Decisions the FCC makes in pursuing these goals could have a significant effect on VoIP providers. Three issues will be important. First, VoIP providers should be concerned if the FCC postpones a global overhaul and instead adopts piecemeal measures to shore up the current system. These band-aid measures could impose burdens on VoIP without any countervailing benefits. As the VON Coalition has argued, there needs to be "overall, complete reform."

Second, VoIP (define - news - alert) providers would benefit from a new system that banishes inequitable rate structures in favor of unified rates based on forward-looking costs. As Vonage has con-tended, if VoIP providers are required to pay compensation to terminating carriers, then this rate should be the same as the rate paid by all originating carriers.

Finally, if the FCC does bring interconnected VoIP providers into the new IC system, then a key issue will be whether the FCC, if it retains termination rates, ensures that VoIP providers are compensated for traffic that originates on the circuit-switched network and is terminated by VoIP providers. IT

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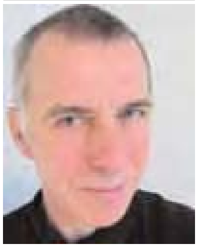
Securing Enterprise VoWiFi

By Michael Stanford



Wireless LANs are intrinsically more vulnerable than wired LANS; in the words of hacker Johnny Cache, WiFi device drivers ". . .have the distinction of exposing a connectionless layer 2 attack surface to all devices in close proximity." A survey of networking and business technology organizations released in July 2006 by Gartner showed that "60 percent of respondents do not believe they have adequate security for their wireless environment." But the explosive popularity of WiFi indicates that for many users the trade-off is worth it. And the trade-off may not be as bad as people apparently fear.

As with so many security issues, the vast bulk of real-world WiFi vulnerabilities are due to lack of basic hygiene. A wireless network that properly implements the authentication and encryption provided by 802.11i is effectively immune to casual attacks.

But the mobile working style encouraged by WiFi means that PCs must be able to connect to the corporate LAN from anywhere on the Internet. This requirement is amplified by Fixed-Mobile Convergence, with the expectation that you will be able to make a voice call through the corporate PBX no matter where you are in the world on any WiFi network. Similarly, it makes sense for a corporation to let visitors access the Internet through the campus WLAN. This implies some kind of open access facility for corporate WiFi networks. Furthermore, in this day of increasing outsourcing of IT functions, your corporate IT services are as likely to be located on a server farm in Oregon as on your company premises, so the concept of a "Local" area network is being diluted on the services side as well.

Putting these notions together, we come up with a seductive notion: all our client devices have to be hardened to the point where they can safely sit on the open Internet, and the internals of our corporate LAN have to be hardened to the point where it can handle potentially hostile devices on the premises. Our servers may not be on our premises, and our services may not be on our servers. So isn't it redundant to shelter our corporate LAN from the Internet with a secure perimeter?

...isn't it redundant to shelter our corporate LAN from the Internet with a secure perimeter?

This is the radical idea proposed by the Open Group's Jericho Forum (www.opengroup.org/jericho/index.tpl). Needless to say, it is a highly controversial proposal. But the Jericho Forum has an impressive list of members, some of whom who have already started to walk the talk. For example, in February 2006 BP (British Petroleum) reported that it had moved 18,000 of its 85,000 client PCs to this "deperimeterized" model, leaving them connected directly to the Internet even when they are located in the office. The movement is also influencing corporations that are not formal members of the Jericho Forum. Toyota Europe is on the record as an advocate of deperimeterization.

The obvious counterargument to the idea of deperimeterization is to combine the hardened clients and hardened server farms with the hardened perimeter and get the best of both worlds, with "strength in depth." But this brings us back to the realm of trade-offs. The most zealous advocates of deperimeterization point out that firewalls promote a false sense of security, that they are a barrier to rapid service deployment, expensive to maintain, and that if each network node is adequately secured, then firewalls constitute redundant system complexity.

It will be interesting to see how it pans out. Market forces are making it an increasingly urgent issue. WiFi equipped notebook computers are rapidly displacing desktops, and dual mode phones will soon add hundreds of millions of wireless clients to enterprise networks worldwide. All these clients will be highly mobile, and expect full access to corporate services via the Internet. IT

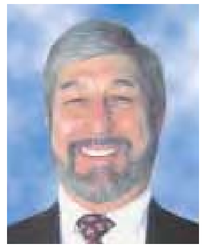
Michael Stanford has been an entrepreneur and strategist in Voice-over-IP for over a decade. His strengths are technical depth, business analytic skills and the ability to communicate clearly. Michael has founded, run, and successfully sold two software companies. The first (Lucid Corporation) developed software for hand-held computers; the second (Algo Communications) developed application software for telephony. Algo was ultimately acquired by Intel, where he subsequently spent six years as a senior manager, ending up as the Director of VoIP Strategy for the Digital Enterprise Group.

In his current consulting practice, Michael specializes in Voice over IP on wireless networks, both WiFi and WiMAX. The October, 2006 issue of Internet Telephony Magazine recognized him as one of "The Top 100 Voices of IP Communications," and the November 2006 issue of VoIP News named him one of "The 50 Most Influential People in VoIP".

Continuity Planning 101: A Continuing Educational Series

VoIP and Continuity Planning — Is VoIP the Answer for Your Organization?

By Rich Tehrani & Max Schroeder

In 2005, more IP telephony systems were shipped than TDM (Time-Division Multiplexed) phone systems. Yet despite the acceptance and thousands of installations worldwide many companies and individuals are still not clear on what they have to do to implement VoIP. Is their network ready? This is a key point as having a network that handles a complex data workload does not guarantee that it is fully ready for VoIP.

3Com has been changing the way businesses speak since it brought the first IP-PBX to market in 1998. More recently, IBM (quote - news - alert) and 3Com (quote - news - alert) have made several announcements including a March 30, 2006 announcement outlining plans to offer the 3Com VCX™ suite of IP telephony solutions on IBM's "all-in-one" System i business computing solution. The March 26, 2007 follow-up was a delivery announcement of the above bundle so it looks like 3Com is still moving technology forward.

We decided to contact Michael J. Leo, Director, Convergence Marketing for 3Com to get his views on the subject of what makes a network both VoIP-ready and in compliance with a business continuity plan.

RT: Rich Tehrani • MS: Max Schroeder • MJL: Michael J Leo

RT: Michael, what would be your simplest example of what characterizes a voice-ready network?

MJL: I can condense it down to 3 critical characteristics:

Promoting Simplicity - Simple-to-manage is a plus for any network but a voice-ready network also needs to be able to scale as the business grows which requires an intrinsically simple architecture.

Ensuring Quality - The latest codecs now position IP phones as having audio quality surpassing analog or non-IP digital phones. However, the network must deliver sufficient bandwidth to attain business-level clarity and immediacy.

Guaranteeing Security - Many companies are unaware that a VoIP network is prone to the same security vulnerabilities as their data networks. Both at the application and Infrastructure levels since the VoIP systems are typically running on common operating systems and networking protocols, including SIP. Along with the constant protection of both Application and Infrastructure it is just as important for the security component to provide resilient QoS for voice traffic. Users also need to be aware that network threats are not always coming from outside their networks. In fact, over 70% of today's treats are coming from inside their networks. You need to provide overall security protection for known and yet to be disclosed "unknown" (Zero Day Threats) for both your wired and wireless networks.

MS: Perhaps you can expand on the security guarantee aspect of VoIP in business continuity architecture.

MJL: The first step in any business continuity plan regardless of the depth of the plan is to ensure the integrity of the nucleus. For example, a failover site with the same architecture and populated with the same data as the primary site will probably generate the same results. A second consideration is that maintaining voice communications is essential in many time-critical disaster scenarios. The concept of true multi-site survivability is that all telephony features, not just the basics, should remain viable in the event of a WAN failure. Taking advantage of the additional bandwidth for VoIP creates new challenges in delivering a high level of QoS within a legacy data network. Companies need to consider deployment options and solutions including security that will continuously provide acceptable bandwidth for time-sensitive applications. Providing traffic prioritization and bandwidth shaping for VoIP traffic needs to be considered mandatory to have quality voice traffic.

RT: Once past the minimums, what advanced security measures need to be evaluated?

MJL: IP telephony voice traffic must be shielded from eavesdropping and introduces additional device types. Plus shielding attacks originating inside the network. Therefore, many organizations must expand beyond the standard perimeter defenses and desktop antivirus software. Organizations with wireless connections also must integrate Wi-Fi protections into the security mix.

Lastly, all of the above must be balanced against network and application performance, quality of service, reliability and bandwidth utilization levels. The most prudent course in migrating to VoIP is to do a full audit when still at the concept stage. Stage two is to develop a voice transition strategy that includes costs and budget constraints, the impact of possible downtime on users and interoperability with other telephony equipment.

Again, we see that planning for a disaster or business interruption can be achieved successfully, cost effectively and as an integral component of day-to-day business operations, if done properly. With today's IP solutions even the cost is within the scope of most enterprises. IT

Max Schroeder is a board member of the ECA, media relations committee chairman, and liaison to TMC. He is also the Sr. Vice President of FaxCore, Inc. (news - alert)

Rich Tehrani is the President and Group Editor-in-Chief at TMC and is Conference Chairman of Internet Telephony Conference & EXPO.

If your organization has an interest in participating in the TMC/ECA Disaster Preparedness Communications Forum, please contact maxschroeder@tmcnet.com or rtehrani@tmcnet.com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By Jeff Hudgins

Can an Asterisk-Based System Scale Out?

Zaptel drivers and Asterisk® software have come a long, long way since their initial release and continue to grow and improve every day. We now see a tremendous number of solutions being deployed that run on Asterisk software. One question that has been raised concerns the scalability of the end solution. So let's explore the concept of a straightforward VoIP gateway solution and how its scale and flexibility are addressed.

The Hardware Layer

The release of Intel's 3000 chipset is aimed squarely at the entry-level server market. It supports single-socket systems but can handle Pentium D and Celeron D as well as Intel's 3000 series dual and quad-core processors. Platforms based upon this chipset can also support up to four DIMM sockets and up to 8 GB of DDR2 memory. With all of these options, there is ample room for application developers to tune and scale their code for a variety of VoIP gateway functions.

In addition to the server component chip set, provisions are also available for plug-in third-party cards to interface to the public switched telephone network (PSTN). Industry-leading board manufacturers such as Aculab (news - alert) (http://www.aculab.com), Dialogic (news - alert) (http://www.dialogic.com), and Digium (news - alert) (http://www.digium.com) all offer driver support for Asterisk. These cards offer a broad range of protocol support and can scale up to 8 T1/E1 interfaces, if needed.

The Operating System Layer

Asterisk is a complete IP PBX software solution that runs on a variety of operating systems (OS) including Linux, Mac OS X, Open BSD, FreeBSD, and SUN Solaris. A typical Asterisk based VoIP gateway would likely be configured with a Fedora Core or similar operating system.

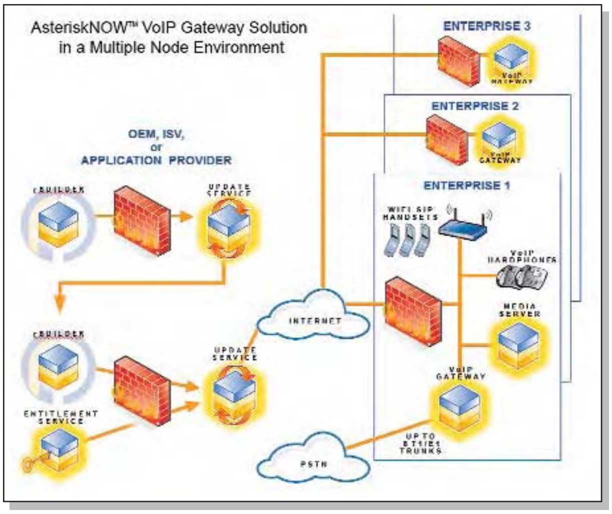
The Application Layer

In the past, developers purchased an Asterisk Developer's Kit which included a TDM board with two modules installed. Developers then built their applications working with the Asterisk platform. The recent release of AsteriskNowTM appliance simplifies server application scale-out by eliminating the hassles of installation, configuration, debug, and maintenance. AsteriskNow is a Software Appliance that contains everything you need - the Linux OS, Asterisk, the AsteriskGUI, and all the other software - to build a complete solution, such as a VoIP gateway system. The Asterisk software can be easily configured with a graphical interface. Additionally, the burden of OS kernel versions and package dependencies are eliminated.

So how was the software appliance created? Digium turned to a company called rPath (news - alert) (http://www.rpath.com) and their rBuilderTM product in order to build the AsteriskNow software appliance and Asterisk Business Edition appliance. According to Marty Wesley, Senior Director of Marketing at rPATH, "rBuilder proved to be the easiest way to build the appliances, turning what was several weeks of manual work into a couple of days. In addition, the application now contains field update capability which comes standard in the rPath Appliance Platform."

The Solution

Integrating the hardware, OS, and application layers into one platform now provides a scalable Asterisk based solution. The figure accompanying this column shows how an ISV or OEM can implement a complete turn-Key VoIP Gateway appliance based upon an AsteriskNow (rBuilder-based) platform.



Final Score

The scale-out of an Asterisk-based system is certainly achievable, but you better be working with industry players that can ensure your success.

Marty Wesley adds: "With the adoption of rBuilder, one of the first things Digium saw was a drop in the number of support calls related to problems with the installation or configuration of Linux. Previously, Digium experienced up to 40% of their support issues that were unrelated to the Asterisk application. Secondly, Digium was able to offer AsteriskNow as a free trial appliance as a lead into their enterprise-grade Asterisk Business Edition. The ease of consumption of AsteriskNow has led to intense interest and trial in the product, averaging over 5,000 downloads a week. The software appliance model opens the door for enterprising VARs to offer specific versions of Asterisk-based appliances. Asterisk can be combined with many other business systems to offer more complete business solutions."

In the end, combining a software appliance with a robust hardware appliance provides Value Added Resellers and End Users a solution they can leverage right out of the box. IT

Jeff Hudgins is VP of Engineering at Alliance Systems. (news - alert) For more information, visit the company online at http://www.alliancesystems.com.

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By Richard "Zippy" Grigonis

Diversified Technology's New Blades

I've been following Diversified Technology, Inc. (DTI) of Ridgeland, Mississippi since their FTS900V Telco System appeared in 1995. (It was a vertical rack mountable, fault tolerant, IBM compatible computer chassis designed for installation in Telco rack mounting frames used in central offices.) Unlike California-based embedded hardware vendors, DTI has its main facility in the most picturesque countryside of Mississippi, not far from the great mansions built in the antebellum South.

At the Embedded Systems Conference in San Jose in April 2007, DTI demonstrated that the CompactPCI (cPCI) form factor has not entirely been eclipsed by the newer, heftier AdvancedTCA (ATCA). DTI unveiled a high performance, low power cPCI Blade based on the Dual-Core Intel® Xeon® Processors. Designed the CPB4712, it's the latest and most powerful bade in DTI's CompactPCI line, examples of which will soon be appearing in VoIP and media gateways to industrial environments and data control centers. Such a powerful cPCI blade means that DTI customers can now upgrade their existing installations rather than scramble to do a forklift upgrade and install ATCA equipment.

DTI demonstrated that the CompactPCI (cPCI) form factor has not entirely been eclipsed by the newer, heftier AdvancedTCA (ATCA).

The CPB4712 features either the low voltage or ultra low voltage, high performance Dual-Core Intel® Xeon® processors at speeds ranging from 1.67 GHz up to 2.16 GHz with 2 MB L2 cache. The blade is based on the Intel 3100 chipset with a 667 MHz front side bus. You can load up to 4 GB of memory on two PC-2700 SO-DIMMs. Also onboard are dual 100/1000 Mbps Ethernet ports routed according to PICMG 2.16, one 10/100/1000 Mbps Ethernet port available at the front panel, one PMC/XMC site (64-bit/133 MHz and x4 PCIe), High Speed USB, and a serial port. Single piece price is $1,795. Quantity discounts are available.

The appearance of the cPCI-based CPB4712 doesn't mean that Diversified has forgotten about ATCA. Far from it. DTI recently introduced two new ATCA blades at the MVA conference in San Diego in February 2007.

DTI's CPB4712 Dual-Core CompactPCI Blade.

First up is the ATS1936 Switch Blade, a low-cost 10 Gbps Ethernet switch. It naturally complies with PICMG 3.0 R2.0 ECN002 and PICMG 3.1 Option 1 and Option 9. The ATS1936 features three AMC sites for OA&M, firewall, and encryption applications. Interestingly, Diversified provides separate control and data planes by separating the Base and Fabric networks. The ATS1936 provides 1 Gbps Ethernet switching on the 3.0 Base Fabric and the 3.1 Expansion Fabric provides 1Gbps/10Gbps Ethernet switching. Both networks support Layer 2 switching as well as Layer 3 routing and IPV6. Single piece price: $5,245.

DTI's other latest ATCA offering is the ATC6231, a dual-core Second Generation AMD Opteron™ processor-based Node Board for next-gen telecom apps such as wireless access/edge, telecom fiber transport, media gateways, softswitches, and Internet IP-based applications. The ATC6231 comes with the 2nd-Generation AMD Opteron processor Model 2210, with 2 MB L2 cache and support for up to 8 GB of memory per processor interface. It uses a high I/O bandwidth HyperTransport technology link interface, Broadcom HT2100 and HT1000 server-class chipset. There's also a standard 2.5-inch SAS micro hard drive for storage, along with 10/100/1000Mbps dual port Ethernet controllers and one AMC.1 site for user configuration (connectivity to the AMC Site is x8 PCI-Express with Common Option to SAS drive and Ethernet). The board also features 16 MB of persistent through reset SRAM for error logging and redundancy. Single price is $4,645. IT

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

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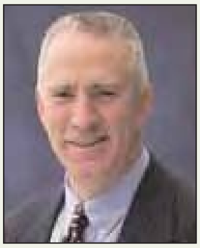
Featuring:

Level 3's Myrle McNeal

Rich Tehrani's Executive Suite is a monthly feature in which leading executives in the VoIP and IP Communications industry discuss their company's latest developments with TMC president Rich Tehrani, as well as providing analysis on industry news and trends.



Myrle McNeal, Senior Vice President, Level 3 Communications

There is no question that VoIP (define - news - alert) adoption among both businesses and consumers continues to surge. But with all the hype surrounding VoIP offerings and the constant announcements about increasing customer figures, what many might find surprising is that, in terms of pure percentages, the number of VoIP users is still quite low. For service providers, however, that presents an ongoing opportunity, though, which many are taking advantage of.

Among the chief characteristics of a burgeoning industry is a high rate of the birth of new companies, followed by consolidation, which is what the IP Communications space is now witnessing. Level 3 Communications (quote- news - alert) has, with a number of significant acquisitions in the last year-and-a-half, including WilTel Communications and Broadwing Communications, established itself among the drivers of industry consolidation.

Rich recently spoke with Level 3 Communications' Senior Vice President of Voice Service Provider Markets Myrle McNeal about the opportunities that abound in the flourishing voice services market, and specifically about how Level 3's acquisitions will benefit the company as it seeks to capitalize on those opportunities.

RT: Level 3 has completed a number of large acquisitions in the last 18 months. Has the company's direction or vision changed during that time?

MM: I'd say generally the vision has not changed. When we started the company, we stated that our goal was to build the most cost efficient network and support it with the most cost efficient operations. Our goal, in that respect, remains the same.

What has changed, perhaps, is the breadth and the capability that we can offer related to that goal. Initially, our core strengths were in IP services, and we've developed a great deal of strength in VoIP services. With the acquisitions, we are also a much, much stronger wholesale provider with regard to video and data service as well as metropolitan services. All that said though, the core expectation still remains the same.

RT: How have these acquisitions affected your ability to service your wholesale customers?

MM: In general, our ability to service wholesale customers is better than it has ever been, and there are a variety of reasons for that. First of all, we now have a metropolitan fiber service or lip service in 116 metro markets that serves about 6,100 on-net locations, which is important for a few reasons. As a wholesale provider, it gives us better reach to the enterprises that our wholesale customers are trying to reach, and business VoIP service providers, in particular, are very interested in that metro footprint. For other wholesale customers, it helps us in that it puts us a lot closer to traffic aggregation points, which, in the past, were primarily central offices, but now are increasingly also cable head-ends as well as wireless towers. There are a lot of different places that the traffic is coming in. The depth of the metro footprint is very helpful in that respect.

Also, our voice capability is much, much stronger than it has ever been. As a result of the consolidation of the Broadwing and Level3 networks, we can now offer a network that is on par with the reach and quality of the networks provided by the legacy inter-exchange carriers. The difference is that ours is also very, very capable of handling VoIP and SIP.

RT: As a leader in the VoIP business, where do you see the industry heading?

MM: In the part of the business where you and I tend to live, we take VoIP for granted - it's what everyone does. But the fact is, that the whole of the telecom industry has still not embraced or fully converted toward SIP, and there are certain parts of the business where TDM is still the primary way of doing business.

The point is that we recognize that and are working very diligently to help customers who are interested in making the transition to do so in a deliberate and low risk manner. In certain parts of the industry, like in the contact center space, many customers use IP-based hardware but, for all intents and purposes, are really not extensively using VoIP-capable networks. We're pushing things along there and helping our customers understand how to implement VoIP. We are also launching capabilities that make that transition simpler. It's easier if you can straddle the fence, which we're capable of doing: We can support customers on either platform and make it easy for them to transition.

The other trends that we're seeing are that the business implementation of VoIP is probably lagging behind the

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consumer market by a few years. If you look at when Vonage and the cable companies began their announcements about launching competing voice services, they really began hitting stride probably a year or 18 months ago, perhaps even as far back as two years or further.

On the business side, though, we believe the market is considerably more fragmented. There are no dominant players, which creates a great wholesale opportunity, or for a business- or enterprise-focused VoIP provider to win substantial market share with a great value proposition. What we are finding is that our customers go to small businesses who are used to paying extraordinarily high rates for traditional services, and they are surprised at how much money they can save and how well VoIP services work and how well they meet their needs. There's a great opportunity and good growth ahead of us in the enterprise space.

RT: What does this growth translate to, regarding your core network, and what is the specific traffic type to which you attribute the growth?

MM: We report our service revenues as core versus non-core revenues, and we expect the core revenues of the company to grow by roughly 17% this year, driven primarily by voice services. While we, obviously, are seeing a substantial uptake in the use of IP-based high bandwidth services, the revenue growth from the company, in the near term, still comes largely from the voice side of the business.

We are also well-positioned in the small business space with a very active and established business partner program that helps VARs and our other providers succeed in the SMB market. So, we're in a good position to continue to drive voice growth there as well.

RT: You've also acquired some fiber. How will that particular set of acquisitions help you in the competitively?

MM: The acquisitions of the fiber networks that we picked up from WilTel and Broadwing benefit us in several ways. First, it extends the reach of our network and provides us additional diversity and greater depth. As a result, we simply have just a considerably more in backbone assets allowing us to operate more efficiently. That is also the advantage of combining those networks, since it allows us to operate the three of them much more efficiently than we could separately. So, we're driving scale and efficiency, which ultimately results in lower costs for our customers.

There's a great opportunity and good growth ahead of us in the enterprise space...

RT: Speaking of cost efficiency, how is Level 3 helping providers cost effectively deliver voice, video, and data over converged networks?

MM: Many people predicted there would be no more investment in telecommunications, but we continue to invest very heavily in the business. We are investing heavily in technology at the transport layer, at the IP layer, and in the voice network. I would guess that if you compare that to other companies, there are probably not many companies making the same substantial investments in their landline voice networks at this point.

We tend to invest in technology, not for technology's sake, but because technology ties back to the original goal, which is to drive cost-efficient operations. When we see a new technology that allows us to deliver equal or better service at a lower cost, we are very aggressive about implementing that technology in our infrastructure so that we can continue driving the underlying prices and costs for services down over time. We believe that also stimulates demand and makes the business better for all of us.

RT: Can you give an example of an innovative application that Level 3 supports with its various services?

MM: There are a couple of interesting ones, but let me talk about one that probably is not particularly well known. One of our customers, who uses Level 3's underlying VoIP capabilities - primarily our local inbound services - where we offer telephone number service and inbound local calling, works in conjunction with a wireless partner to offer a very specific wireless service offer.

Essentially, as a combination of a wireless provider and Level 3, our customer and our customer's customer are able to offer what amounts to a national wireless offering - basically a cell phone service - targeted at senior citizens who are looking for simplicity in the way that they actually use their cell phone. So they offer a telephone that has special characteristics, like bigger buttons and easier calling plans like prepaid, as well as other services.

I don't think most folks view Level 3 as a player in that space, but we do have an application where our unique VoIP capabilities fit very well into a specialized application that is being actively and broadly marketed on a national basis.

RT: Where do you see Level 3 in the next three to five years?

MM: I think Level 3 is going to go where we have always predicted it would go; it just hasn't gotten there as quickly as we would have liked. We continue to see demand for bandwidth growing at a substantial rate, driven by increases in disaster recovery applications and increased video distribution. All of the things that consume bandwidth are on a substantial upward trend. I think that will drive substantial growth for us over the next two to five years.

If you layer on top of that the VoIP business, which, in our estimation and according to various research groups, is growing at a rate of 80-100% annually nationwide, there's still substantial growth opportunity for us in the voice business as well. We are very well-positioned for that as new parts of the market begin embracing VoIP, which is where some of our voice strengths lie and where we think we are uniquely qualified.

Hosted VoIP Service Providers

There was a time, not long ago, when the longevity of the hosted VoIP market was questioned by many in the communications industry. But that was largely on the assumption that large businesses, for the most part, had little interest in a hosted solution, opting instead, for an on-premises solution they could fine tune to their liking. And to a large extent, that assumption was accurate.

But, like all good products, it needed time, and it needed the right target audience. And indeed, there was one very large customer base that had historically been underserved. For some providers, the SMB customers were too small, yet those that were willing to approach the SMBs typically had a solution with a price tag considerably higher than the SMB could afford. So the catch was to create a solution that provided the features of an enterprise-class system, but at a price point in line with a smaller business.

Thus, the hosted VoIP market was conceived, and just like any child, it grew. Indeed, it has recently experienced a considerable growth spurt, as many service providers have realized the substantial opportunity presented by the sheer size of the SMB space.

The hosted model is particularly enticing to the SMB because it offers a low-priced way to gain IP PBX features, so even the smallest company can put on a big business face. Hosted VoIP provides effective communications mechanisms for mobile workers, ensuring they can stay in touch while away from the office. It also limits overhead, since no on-site staff is required to maintain the system, and there is no hefty infrastructure investment. The list continues, but in the end, hosted solutions are about meeting the needs of the SMB market - both cost- and feature-wise - without breaking their budgets.

The following is a listing of hosted VoIP providers, each looking to grow its share of the hosted VoIP market. We encourage you to use this list as a starting point for your search for a hosted VoIP provider. There are many differing offerings, some better suited to your particular needs than others, so be sure to contact these providers for additional details.

Accessline Communications http://www.accessline.com	**Cox Communications** http://www.cox.com	**Natural Convergence** www.naturalconvergence.com	**Sprint** http://www.sprint.com
Acredo Technologies http://www.acredo.us	**DSL.net** http://www.dsl.net	**Net2phone** http://www.net2phone.com	**SunRocket** http://www.sunrocket.com
AdvantageVoice http://www.advantagevoice.net	**DSLi** http://www.dsli.com	**O1 Communications** http://www.o1.com/	**Time Warner Telecom** http://www.twtelecom.com
Apptix http://www.apptix.com	**Exquisite Global Networking** http://www.exgn.net	**Packet8** http://www.packet8.net	**Velocity Networks** http://www.vel.net
Aptela http://www.aptela.com	**Global Phone Corporation** http://www.gphone.com	**Pandora Networks** http://www.pandoranetworks.com	**Verizon** http://www.verizon.com
AT&T http://www.att.com	**GotVMail** http://www.gotvmail.com	**PingTone** http://www.pingtone.com	**Versature** http://www.versature.com
Bandwidth.com http://www.bandwidth.com	**HIP Communications** http://www.hip.ca	**Qwest Communications** http://www.qwest.com	**VirtualPBX** http://www.virtualpbx.com
Broadcore http://www.broadcore.com	**Innoport** http://www.innoport.com	**Reignmaker Communications** http://www.reignmaker.net	**Vocalocity** http://www.vocalocity.com
Broadvoice http://www.broadvoice.com	**Inphonex** http://www.InPhonex.com	**ReVoS** http://www.revos.com	**Vonage** http://www.Vonage.com
Cablevision http://www.cablevision.com	**M5 Networks** http://www.m5net.com	**RingCentral** http://www.ringcentral.com	**XO Communications** http://www.xo.com
CallTower http://www.calltower.com	**MediaRing** http://www.mediaring.com	**Ring9** http://www.ring9.com	
Covad http://www.covad.com	**Mendax** http://www.mendax.com	**Speakeasy** http://www.speakeasy.net	

Global VoIP: Market Evolution or Revolution?

By Ofer Gneezy

There's a photo I like very much that seems to capture the current state of global VoIP. Taken by a securities analyst, it shows a frozen food section in a grocery store in Germany. In the center of the picture is a freezer case, and above it, the analyst has added an arrow and text that reads, "Frozen peas". Also there - within arms' reach of the vegetables - is a display of shrink-wrapped VoIP service packages which the analyst has highlighted with an arrow and accompanying text that simply reads, "Voice".

This rather humorous scene illustrates the profound changes that are underway in international telecommunications. In addition to seeing new types of voice services and features enabled by IP and the Internet, the entire definition of a "phone company" has been turned on its head. VoIP is achieving mainstream acceptance to the point that we're buying it off-the-shelf at the supermarket. That's quite a development considering that little more than a decade ago the entire industry was defined almost exclusively by the big incumbent telcos such as AT&T, Verizon, BT, China Telecom, France Telecom and other usual suspects. However, VoIP (define - news - alert) has spawned a new class of telephone company utilizing IP as a communications platform, the "poster child" of which is Skype. Today, Skype adds 300,000 users a day; it's a mark I doubt any traditional carrier has ever matched. As for the future, ABI Research anticipates worldwide VoIP subscription rates will grow from roughly 45 million in 2006 to more than 250 million by 2012.

It's clear that a massive migration to IP is well underway and the traditional telecom landscape has changed forever. When consumers face the choice of "Voice or Iced Peas," it seems clear that this is more than a market evolution, it's a revolution.

New Version, New Visions

Due to the rise of VoIP, traditional telcos now share the stage with names unheard-of or non-existent in the mid 1990s. An example is our company, iBasis, which in just ten years went from start-up to carrying more than 12 billion minutes of international voice minutes annually - enough to be ranked among the 10 largest carriers of international voice. Even more surprising has been the involvement of companies whose main business focus was never voice service, a diverse group capable of leveraging large brands like WalMart, Best Buy, eBay, Google, Yahoo, and Apple to become potentially powerful telecom market players overnight.

The technology and capacity of this expanding field of "emerging carriers" has grown dramatically. On the other hand, traditional carriers have experienced deterioration of their market share. Telegeography research has shown that access lines for incumbent local exchange carriers (ILECs) steadily decreased from 1Q 2003 through 3Q 2006. In less than three years, ILEC access line capacity was down roughly 12 to 20% across the board. Punctuating this, a Wall Street Journal story in January 2007 reported that Deutsche Telecom had lost two million access lines in 2006 alone.

Welcome to Voice 2.0 and Telco 2.0, a new version of telecommunications services delivered by a mix of evolving traditional carriers and emerging new players. Yet, it is from the latter group that new visions are quickly giving birth to the latest services, which are only possible through IP. When a consumer uses Google to search for Chinese food, then can simply click on a map of purveyors to call in an order without ever leaving their desktop, clearly one person's vision has made another's dream come true. Likewise, the latest version of Skype (news - alert) automatically turns every phone number on a web page into a "click to call" button for Skype subscribers. I don't know about you, but my desk phone has never done that for me. It's these types of simple to use yet revolutionary services that continue to drive further VoIP adoption.

Migration Nation

VoIP began in the deep recesses of the core of the global network and provided cost savings for international long distance providers by enabling them to bypass the

established settlement regime. During those early years, pioneers (including iBasis) developed the proprietary technologies and expertise needed to deliver the high quality VoIP service carriers and retail consumers required. Spurred by the widespread adoption of broadband Internet access, VoIP spread from international long distance through domestic long distance markets and out to the edge of the network to provide local service to consumers and enterprises.

Today, largely through aggressive and creatively viral marketing, providers ranging from cable television operators to emerging entrants such as Skype and Vonage to Internet giants like Google and Yahoo are establishing a new order for retail communications services.

The migration of voice communications to IP is clearly in full swing. Whether it's with cable operators, CLECs, independent providers, or Internet properties, the number of consumer subscribers to VoIP services is growing. But that's not the whole story; in the enterprise, IP PBXs are outselling conventional PBXs by a significant margin. Spending on IP phone systems, equipment and services is estimated to increase at an annual growth rate of 28%, and the demand for enterprise voice gateway ports is also growing.

The migration to VoIP is not a geographically isolated trend. Around the world, major carriers, including KPN, BT, Deutsche Telekom, France Telecom, Telecom Italia, and Telia Sonera are rolling out commitments to all-IP networks. Carriers of all types, from Tier One incumbents to voice over broadband providers to mobile network operators, are increasingly pursuing the benefits of VoIP.

In Japan, the government has mandated that domestic TDM networks be replaced with integrated IP-based nets; incumbent operators KDDI Corporation and NTT have said they will be "all IP" by 2008 and 2010 respectively. Already, VoIP comprises 15% of all fixed-line phones in Japan. In the U.S. it is estimated that 9% of households have some kind of VoIP service running. IDC predicts that residential U.S. VoIP subscribers will grow from 10.3 million in 2006 to 44 million in 2010. France Telecom recently stated that VoIP accounted for 40% of consumer fixed traffic in that country during 2006.

Throughout global telecommunications - whether you're looking at equipment purchases, subscriber levels, industry and government initiatives, whatever - it's clear that the reign of the "IP nation" has begun.

Evolution and Revolution

VoIP is shaking another cornerstone of the global telecommunications industry by questioning the very basis of the traditional telecom business model. Who pays whom and for what in the IP age?

For example, Internet properties like Google and Yahoo can use voice service as a cost of attracting advertising revenue. In Germany, Google and Deutsche Telecom recently battled over a related matter, and for now, Google will pay Deutsche Telecom a fee for basically using its "pipes." The Wall Street Journal also reported in February 2007 that Cingular is now sharing with Apple a portion of monthly revenues derived from users of the Apple iPhone.

Economic challenges loom, particularly at the retail level, but, they do not really threaten VoIP's prospects any longer. It is difficult, if not impossible, to put down a revolution when the groundswell is so comprehensive as to create a market supported by business, government, technology innovation, and most importantly, consumers. Traditional telcos realize that IP is the future and they are adapting, too. After all, history shows that failing to evolve often results in extinction…which reminds me of another of my favorite photos, which happens to be of a piece of furniture in my home.

When iBasis was founded in 1996, we purchased a number of time division multiplexing (TDM) switches, $300,000 behemoths that were then essential for interconnecting to established carriers. Technology advancements soon enabled us to "go switchless" and we decommissioned the units as quickly as we could. We were able to sell the first switches we removed, but by the last one, the market for used TDM equipment had disappeared. Even a listing on eBay produced "no takers."

Times have indeed changed, but even so, the TDM switch does continue to serve a purpose. As the most expensive coffee table in my home, it's a constant a reminder of the continuing VoIP revolution. IT

Ofer Gneezy is President and Chief Executive Officer of iBasis (http://www.ibasis.com), a leader in international long distance, VoIP, and prepaid calling cards. As co-founder of iBasis (news - alert) in 1996, Gneezy is a true IP telephony visionary and recipient of Pulver.com's Industry Pioneer Award. Today, iBasis carries over 12 billion minutes of international voice minutes per year, serves more than 500 carrier customers, and operates a VoIP network reaching encompassing 100 countries and over 1,000 direct routes.

Enventys turns to AltiGen

Out With the Old

Enventys is a product design and marketing firm with the goal of being a singular solution for product development. It focuses on the end-to-end process, from ideation to distribution, developing simple, yet effective strategies for launching new products or developing existing ones. Being at the forefront of leading edge product development with many prominent Fortune 100 companies as well as start-ups - Johnson & Johnson, Bank of America, Verizon Wireless, and Siemens, to name a few - it is easy to see why it also felt it needed the latest in communications technology.

Enventys was using an "old" PBX system - it really wasn't very old at only five years, but perhaps the more appropriate term would be outdated PBX system that simply was not able to meet its evolving needs.

The company's founder, Louis Foreman, explained that, while the existing telephony platform didn't have anything terribly wrong with it, neither did it offer all the features he would have liked, and which he knew were available from newer IP-based phone systems.

Considering the Alternatives

Enventys considered the hosted model, but decided it was too big - especially with its current growth rate. In addition, the company was willing to make the investment in on-premises equipment that would provide more control as needs changed.

"We make decisions based on efficiency rather than on cost," said Foreman.

So, after looking at six or seven alternatives - all the key players in the space, according to Foreman - Enventys chose AltiGen. AltiGen (news - alert) offered a competitive product at a very reasonable price, but with the features and flexibility Enventys was keen on acquiring.

In With the New

So, having decided that AltiGen was the way to go, Enventys bought two MAX 1000 systems, which together give the company room to expand to as many as 96 extensions. Currently, it has about 60 in use, but, as Foreman noted, growth potential factored into the decision to deploy an on-premises solution.

The MAX1000 is an all-in-one system that performs most of the functions many systems require two, three, or even more appliances to handle, which not only adds to the cost, but also the maintenance requirements. Each MAX1000 supports as many as 48 extensions, and include 4,500 hours of voice mail storage, an integrated conference bridge, call recording functionality, mobility solutions, auto attendant features, SIP trunking support, and more.

According to Foreman, the installation went remarkably smoothly. In fact, the only hiccup, if it can even be labeled as such, was that Enventys had to be completely wired - there had to be a Cat 5 cable to each office.

"But that's a small price to pay to have that kind of capability," consoled Foreman. "It's nice to have flexibility, and to be able to change call routing schemes and make adds and changes. For once, I feel like we're in control of our phone system."

Enventys even has two remote employees - in Asia - whom they have added network extensions for, so, despite being thousands of miles away, they can easily interact with main office personnel and gain the benefits the MAX1000 offers.

Assessing the MAX1000

According to Foreman, three of the most beneficial features of the system - and the three he uses most frequently - are call forwarding, the conference bridge, and Outlook integration.

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By Erik K. Linask

"These are the features that have proven to be time saving and efficiency boosting features for me," said Foreman. "There also are all the features that are taken for granted these days - kind of like power locks and air conditioning on a car. They just come with them. Instead, it's the new things that we're excited about."

The call forward feature, he says, is among the most important features today, since it enables forwarding to a cell phone, allowing you to react to incoming calls as though you are at your desk - even though you might be on a train, in a hotel, or shopping for an anniversary gift.

Most of the other options Enventys considered had a conference bridge available, but at an added cost. The nice thing, says Foreman, is that AltiGen includes the bridge in its system, which make it a very easy solution to like. And of course, the Outlook integration, which can display information on the monitor based on the incoming call. Foreman says that he doesn't gain the full benefits of this, since he is not in sales, but the feature enables users to instantly see all pertinent information about a caller, making interactions more pleasant and more efficient.

In addition to the functionality of the system, AltiGen's phones themselves have "a nice look and feel to them." Though people are beginning to use headsets more and more, and answering calls via their computer monitors, the phones look nice in a modern office environment and are completely functional. In fact, Enventys purchases two extra phones that employees can take with them when traveling. They merely need to plug them into a network connection, and they instantly have connectivity, as though they were still at their desk.

Though it has only been using the MAX100 system for about a month, it is evident the Enventys is patently pleased with its choice. Not only does it meet its needs with regard to features and ease of use, but it is also scalable should the company need to add a third unit. One thing is certain: this proves that there is on-premises equipment available for the SMB space that is both feature rich and affordable. IT

Erik Linask is Associate Editor of INTERNET TELEPHONY, IMS Magazine, and SIP Magazine.

By Erik K. Linask

A Case of Night and Day

Coldwell Banker Meadow Realty Goes Hosted

Darkness Falls

The more a communications system becomes a small business' lifeline, the faster the business tends to realize it is time to look for a new system. When business owners start saying things about their existing systems, like, "I'm just not happy with it," it's inevitably time to make a change.

That was just what Mike Alamia, broker owner of Coldwell Banker Meadow Realty in East Meadow, NY, thought about his old phone system. In fact, he had been through several different household name phone systems, and they simply were enabling effective communications for his 12-person real estate office. He needed greater functionality and reliability, so that he and his team could focus on selling homes, not the phone system.

"I needed more on-hold and jump lines," explained Alamia. "I didn't want anyone to ever get a busy signal when they call the office."

Daybreak: Hosted VoIP Shines Bright

Alamia proceeded to "call everybody under the sun," in search of new equipment, with little luck. His luck changed, however, when his Broadview Networks' local sales representative dropped by for a routine visit (the office was using Broadview for its Internet service). Upon hearing Alamia's concerns, the rep proceeded to introduce Mike to Broadview's hosted service, which is powered by Ottawa, Canada-based Natural Convergence's silhouette platform.

silhouette provides VoIP (define - news - alert) capabilities to end customers with no on-premises equipment to install, other then a router, an Ethernet switch (both of which are most likely in place for the LAN and Internet connection anyway), and IP endpoints. Natural Convergence focuses its solution to meet the needs of small businesses with between four and fifty seats.

Upon signing up with Broadview, the real estate company purchased a number of Mitel IP phones, which are certified for use with the silhouette platform. Alamia pointed out that he had the option of leasing or buying, but that the purchase price is low enough that, unless you are extremely tight on cash, it makes sense to buy, especially since there is often used equipment available for about $100 per phone.

"I would have surpassed the purchase price in about 10 months," commented Alamia.

Silhouette Casts a Big Shadow

So, now that he has been on the silhouette platform for about a year, he is convinced he made a decision that has had a tremendous positive impact on his business. He has six lines, plus a fax line (which is imperative in the real estate business), and another six jump lines - all for a very low initial investment. The company is truly pleased with how the system has performed: "It's got all the bells and whistles. It's incredible, truly," said Alamia.

With the new system, Meadow Realty has added call control flexibility. The phones can ring at one desk, at all the desks, or some combination of the two; on-hold messages can be easily swapped out; the company even emailed a personalized message that was immediately added to the system.

According to Alamia, the call quality, compared to his old system, is like the difference between night and day. With the old system, he claims he had trouble hearing clients, and they had trouble hearing him. Now, he says, "the call quality is remarkable. I no longer have people continually asking, 'What?'"

In addition, with realtors being frequently out of the main office, the call forwarding features now available to them are a tremendous productivity enhancer, since they are always accessible at their office number. Importantly, since there is no on-site IT specialist, Alamia says its good that the features are extremely easy to use.

"They are a joke - if I can do it, anybody can," he said. "In fact, I wouldn't know how to make it any easier."

What's more, there is nothing for Meadow Realty to maintain. There is a support system in place, with someone always available, 24 hours a day, seven days a week, which is crucial for a seven day a week business.

"I simply pick up the phone and, regardless of what it is, it gets taken care of," said Alamia.

In fact, outside of buying the phones, the only thing the firm had to do was bring in a T1 line, which Alamia says he wanted to do anyway, simply to be able to keep up with technology - which he has also now done by implementing a hosted VoIP system. Is he happy with his decision?

"I would recommend the system to a family member," he boasts. "Everything is great about it, including the rates. In fact, they were able to lower my monthly expense by about $400."

As it turns out, that fateful day, a little more than a year ago, when his local sales rep stepped in at just the right time, turned out to be a business changing experience for Mike Alamia and his small real estate office. IT

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Enterprise Peer-to-Peer Communications

Chances are that the first "peer-to-peer" (P2P) communications application you ever encountered had nothing to do with VoIP. Instead, it was probably one of those systems for exchanging files (particularly music files) over the Internet without going through a website, such as Napster, Gnutella or KaZaA. Such applications create a sort of meta-network riding atop the Internet or other IP network, converting your PC into a "node" on the network that acts as both a client and a server, enabling users to connect directly to other nodes and trade media files. (Think of a wired version of a WiFi mesh.) The plug-and-play simplicity of P2P - not to mention its low cost - makes it very tempting to adapt it to consumer and even business voice applications, particularly for smaller businesses that can't be bothered with the expense and expertise needed of deploying and maintaining a more sophisticated IP PBX.

Indeed, Frost & Sullivan's report, "North American Enterprise Peer-to-Peer Telephony Solutions", reveals that small businesses and "micro enterprises" with less than 20 users constitute 89% of the total number of establishments in North America and still remain hugely underserved by telecom solution providers. The report says that P2P telephony solutions "guarantee up to 80 percent cost savings by eliminating call servers and associated hardware". The report also states, however that P2P technology really needs the backing of IP PBX system manufacturers in order to become a mainstream enterprise communication solution. P2P technology vendors must deal with the unique challenges posed by small businesses in terms of limited technology funds and competition from managed service providers. P2P vendors must be able to field an inexpensive, quality product that can overcome security, quality of service (QoS) and reliability issues, as well as the negative publicity image that P2P technology in general received in its "file-sharing and intellectual theft" days.

The report suggests that P2P vendors should partner with service providers, desk phone manufacturers, and software application developers so as to enhance the visibility of P2P products.

The inherent simplicity of P2P VoIP can be very seductive. After all, at times it seems that just about everybody in the world with a PC has downloaded a Skype client. Skype, the brainchild of KaZaA founder Niklas Zennström, is the largest VoIP peer-to-peer system in the world. It's free too, unless you want to talk to people who still use Ye Olde public wireline/wireless networks (or *vice versa*), in which case you must use the relatively inexpensive SkypeOut or SkypeIn services.

Popular Telephony (news - alert) (http://www.peerio.com) has also had a run in this area with their serverless, patent-pending technology for peer-to-peer telephony. Called Peerio, it "marks the next most significant step in computer networking evolution since the creation of mainframes and the subsequent migration towards existing client/server architecture," if they do say so themselves. Admittedly, Popular Telephony has been actively developing scads of products and solutions in this area, such as Peerio Core, an advanced middleware for P2P networking; Peerio, a reference design for a P2P telephone device; Peerio Call Control, an XML-based implementation of key PBX features; Peerio FWT, an open-source TURN protocol implementation; Peerio Biz, a white-label P2P feature-rich softphone; and GNUP, a lightweight web-activated callback bridge and billing software package.

One interesting freebie competitor to Skype is the Global Village Internet phone service (news - alert) (http://www.globalvillage.com) a division of Boston-based Zoom Technologies (news - alert) (http://www.zoom.com), the famous analog modem maker that now designs and produces VoIP Gateways, ADSL modems, cable modems, dial-up modems, Bluetooth products, and other communications devices under the Zoom, Hayes and Global Village brands.

In the 1990s, you bought a service and you got a free modem. In the 21st century, you purchase a Zoom DSL modem and you get the free Global Village service. To be specific, you can make free calls to other Global Village subscribers, all of whom have a unique 7 digit phone number. To dial another Internet service, you must first dial a "1", then a 3-digit "area code" followed by the number on the other service. For example, to call someone with a FWD (Free World Dialup) number, you sim-

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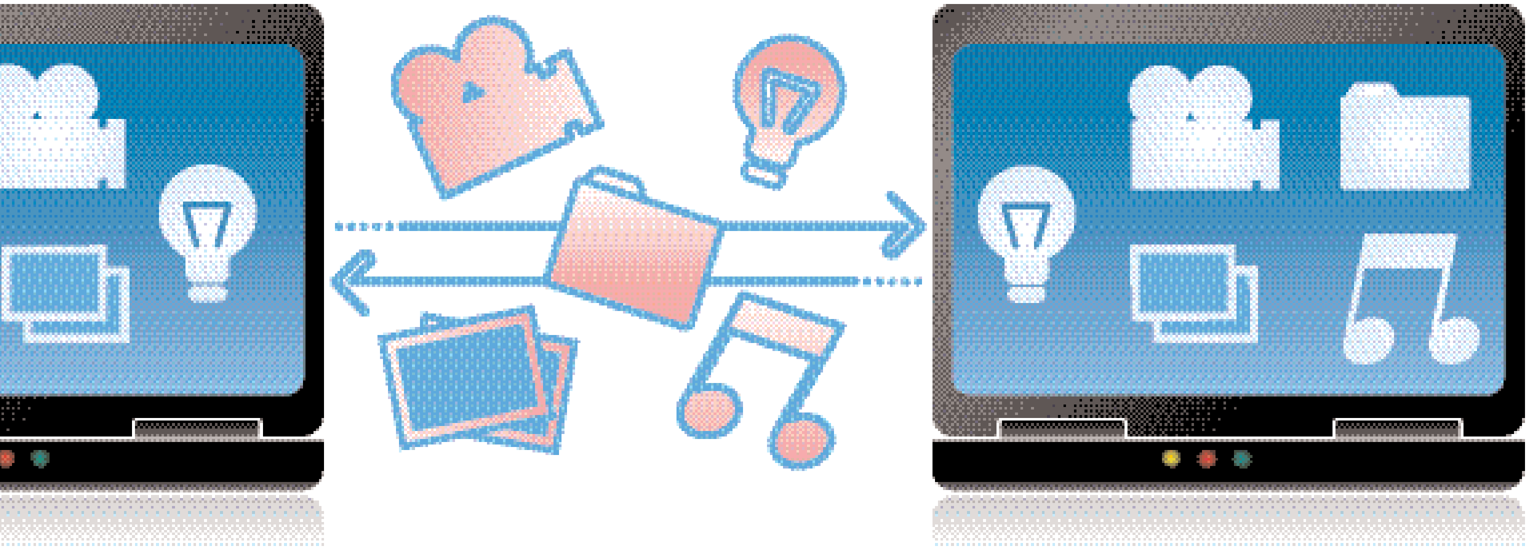
By Richard "Zippy" Grigonis



ply dial "1 393" followed by the phone number. To call someone with a SIPphone number, you would dial "1 747". For calls to the PSTN, you can opt for a plan where you pay only for calls you make, or U.S. customers can opt for an unlimited U.S. calls plan with a flat-rate monthly fee.

Speaking of SIPphone (news - alert) (http://www.sipphone.com), it's another freebie P2P service that uses the SIP (Session Initiation Protocol), the favored call control protocol for VoIP. To use it you rely on the usual suspects: either a hardware adapter or download their free softphone, a broadband connection to the Internet and, to make free in-network (SIP-to-SIP) calls, both sides of the call must have a SIPphone adapter or softphone. Naturally, you do have to pay to call users who have conventional telephones, which involves purchasing "SIP minutes". To receive calls on your SIPphone from PSTN phones, you'll need at least one Virtual Number ($12.00 for 3 months, or $35.00 for 12 months). Calls to your Virtual Number are free to you.

Commoditization and its Effects

The Frost & Sullivan Report mentioned at the beginning of this article suggested that P2P telephony vendors should concentrate on diversifying their distribution channel by reaching the mass market via both retail and online stores. Zoom has done this to some extent with their "buy a modem and get our Global Village service" strategy.

The scary part about commoditization is that, in the "race to the bottom" (price-wise), one ends up with essentially free voice service. Value-add features - what were called "enhanced services" in the old PSTN - were originally designed to keep voice customers from leaving a carrier or provider; i.e., "reduce churn". It was always assumed that voice was the real revenue-producing "killer app". Now however, as unlimited local and domestic long-distance calling makes its appearance, those new and exciting services may become the bread-and-butter of the whole telecom industry. As the "gee-whiz" aspects of IP Communications get lost in the machinery (along with electricity, plastic and LEDs), low-end P2P systems will find a home at retailers, while higher end IP PBX systems stake a claim in larger enterprises and vertical markets.

Retooling for Success

Peer-to-peer systems particularly irk those PBX makers of old that used to charge $50,000 and more for big card cages. With the writing on the wall, however, many are venturing into the field.

For example, Avaya Inc. (quote - news - alert) (http://www.avaya.com) acquired Nimcat Networks, a developer of embedded, peer-to-peer IP communications applications software such as its nimX embedded software for enterprise IP phones. Nimcat's technology boosts the endpoints' intelligence and eliminates additional hardware, such as call processing and application servers, thus simplifying installation and slashing start-up costs.

Avaya Labs has been working on both a system architecture and enabling technologies so that distributed enterprise mobile workers can link up in a hybrid peer-to-peer network over long-range and short-range wireless connections at the application layer.

In the meantime, Avaya at the 2006 VoiceCon show debuted its peer-to-peer SIP solution, one-X Quick Edition. Quick Edition delivers voicemail, auto-attendant functionality and sophisticated call management features including Call Forward, Conferencing, Park, Page and Retrieve, allowing even small businesses to appear to be a large, professional organization. Remote branches and teleworkers can be linked together via Quick Edition, yet the system allows for central management. Adding additional phones is as simple as connecting them to the network. Since each Quick Edition phone backs up the others' features, the failure of one phone will not affect the others.

Whenever one of the "big boys" enters the peer-to-peer VoIP arena, they tend to avoid "pure" P2P technology, since they want to be able to leverage existing legacy PBXs and other equipment already on the customer premise. That's no problem since SIP allows you to use IP as a pure transport layer, so you can create a centralized call system, a distributed server model, a peer-to-peer model, or a hybrid of any and all of these.

For example, SessionSuite IP Telephony Enterprise Edition by BlueNote Networks (news - alert) (http://www.bluenotenetworks.com) enables interactive IP communications within enterprises or extended communities, enabling the extension of voice and video services to teleworkers, remote offices, mobile users and nomadic "road warriors". The high availability aspects of the product allow enterprises to maintain normal business operations during weather emergencies, natural disasters, security crises, and health emergencies. It's not quite "pure" peer-to-peer in that a server is necessary, but it's less expensive than a full-blown, old PBX system.

BlueNote's SessionGateway is used to bridge VoIP and PSTN/PBX infrastructures, allowing VoIP users to place and receive calls with users of the PSTN or legacy PBXs. More than two dozen soft phones and hard phones are supported. The software even allows an organization to build its own Internet-facing telephony system, 'a la Skype, so the company's customers and partners can be part of the company's pseudo-local exchange, along with the teleworkers and remote offices.

Iwatsu (news - alert) (http://www.iwatsu.com) has also adopted a stance somewhere between the old world PBX/communications server and a radical P2P system. Their Iwatsu Enterprise 3.0 Communications Server expands to 1024 ports and easily deals with any of the four dominant communication protocol schemes (SIP, VoIP, TDM, and H.323), and it operates as a media bridge gateway that converges and transmits both voice and data traffic. The system supports P2P communication so that IP phones can connect to each other directly over the network, bypassing central system resources. Any Iwatsu ADIX system can be upgraded inexpensively to an Iwatsu Enterprise 3.0.

Even standards such as Megaco/H.248, a joint effort of the ITU and IETF that defines the operation of media gateway controllers and media gateways, supports both peer-to-peer communications and centralized communication systems.

Security, Security and More Security

Since quality of service appears to be less and a less of an issue as more and more bandwidth and advanced networking monitoring technology becomes available, the most pressing issue in the peer-to-peer world will undoubtedly remain security. Ironically, whereas centralized communication server architectures are susceptible to denial of service (DoS) attacks, SIP-based systems running in a distributed, peer-to-peer network are far more resilient, since there is no single target that can become the principal point of failure.

Of course, only time will tell, as millions of business owners flock to their local store and return with the latest "telephone system-in-a-box" special offer. IT

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SIP's Role in Enterprise IP Telephony

Session Initiation Protocol (SIP) is an Internet Engineering Task Force (IETF) initiative for managing the handshake procedures for beginning and ending real-time communications between IP network endpoints.

SIP is a text-based protocol, similar to HTTP and SMTP, for initiating interactive communication sessions between users. This makes SIP easy to troubleshoot, enables fast application development and presents a stable framework for establishing interoperability between devices, applications, call controllers and gateways. SIP is used to enable human-to-human communications that might include voice, video, chat, interactive games and virtual reality.

Since February 1996, SIP has developed a substantial industry infrastructure and momentum to encourage and promote its use, including over 130 IETF drafted initiatives influenced by SIP and the technology-promoting SIP Forum.

> The IETF framework for SIP offers a rich set of standards for authentication and privacy...

SIP is designed to perform session setup independent from the communications flow. End devices speak to each other directly using whatever application they have available. This delivers a greater degree of flexibility, failure recovery and scalability since the network maintains no state information about the end devices.

Let's take a close look at some attributes behind SIP's growing adoption.

SIP is Simple

Using the IETF fundamental of technology reuse and the proven value of simplicity, the SIP message set is a simple construction - six messages that appear in clear text to facilitate call setup. Clear text allows for easy troubleshooting and avoids complex software interactions and other processing that affects interoperability. The six messages are Invite, Trying, Ringing, OK, ACK and Bye.

SIP is both elegant and practical. Its base assumption is that all SIP endpoints and elements exist in the IP environment, an arena already equipped with standard mechanisms to handle packet transport priorities, privacy and other required services. These value-added services do not require specification in the SIP framework. Conversely, in legacy environments, such as the public telephone network, the use of HTTP is neither integral nor readily available.

This building block approach on an IP base is unique to IETF initiatives. The results are nearly trivial protocol definitions such as SIP in contrast with older session control or interface protocols such as H.323 or Q.SIG from the International Telecommunications Union (ITU).

SIP is Secure

There are tradeoffs between interoperability and cost when considering security. Simple standards make it easier to interoperate within a multi-vendor network, but they also expose the network to potential abuse. This vulnerability, however, can be effectively addressed by strong authentication and privacy services that do not interfere with the primary business benefit of standard-based solutions and also enable a long investment life through interchangeable vendors, services, applications and devices.

The IETF framework for SIP offers a rich set of standards for authentication and privacy, including secure SIP, secure RTCP, and secure RTP. These capabili-

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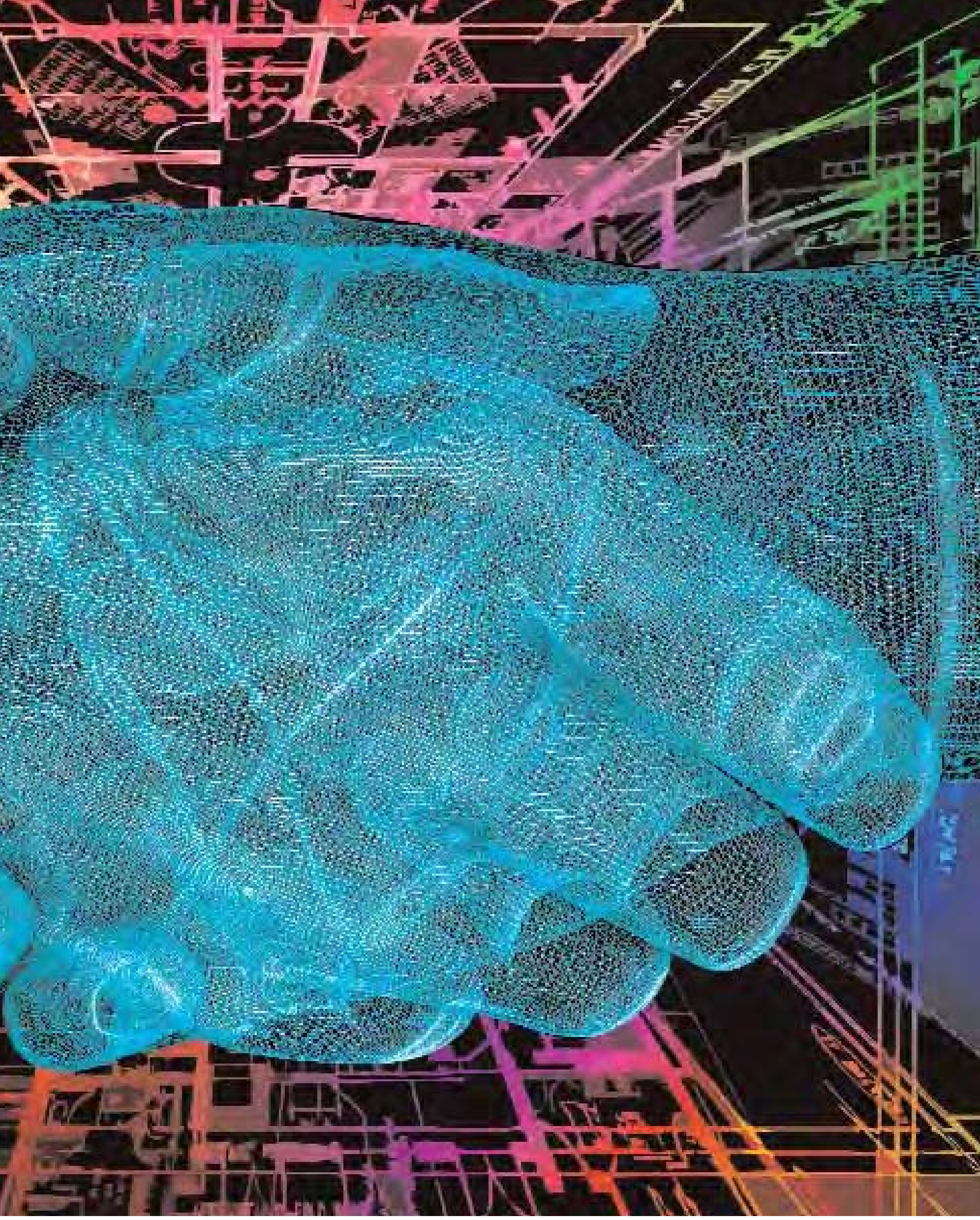

By Pat Rudolph

ties leverage IETF proposals for the use of standard implementations such as Transport Layer Security (TLS) for robust session privacy service and Secure/Multipart Internet Mail Extensions (S/MIME) for session control packet privacy.

Most vendors implementing H.323 offer no such options. Instead, they choose to implement proprietary derivatives to facilitate rudimentary forms of privacy in first-generation IP-PBX devices. This approach is typical of the PBX vendor solutions brought to market during the past two decades in which proprietary digital signaling protocols were implemented on endpoints and PBX fabric. The result of these decisions was nominally better security but considerably more vendor-specific lock-in that failed to provide many useful features - such as call forward and hold - at a standard service level.

SIP is a Standard

Despite being a standard, SIP implementations do vary. Some vendors, most notably TDM-based PBX vendors and first-generation IP telephony vendors, offer SIP interoperability as an interface into their otherwise non-SIP telephony system. This approach helps the vendor maintain account and feature control. They can charge extra for SIP support for only the most rudimentary features while offering advanced premium-priced feature options with their proprietary call control protocol and implementation.

Another approach is using standard SIP among endpoints such as IP phones, video cameras, call controllers and gateways to the public telephone network. This can enable a robust and inexpensive, vendor-neutral environment capable of rapidly delivering a portfolio of applications for presence, conferencing, contact center, messaging and mobility services.

SIP in the Enterprise

SIP has become a foundation for building innovative services aimed at enterprise. For example, SIP trunking services allow two SIP-empowered enterprises to communicate using RTP from endpoint to endpoint without the need for a gateway between them. The growth of this category of services is helping reduce costs by lowering the load on packet-circuit gateways and reducing regulatory and tax burdens.

SIP trunking also improves audio quality since bandwidth allocation can be negotiated end-to-end instead of end-to-gateway-over-digital-to gateway-to-end. Fewer digital hops with more bandwidth enable wideband audio quality, as well as SIP-initiated video conferencing.

As demand for IP telephony and SIP services and implementation continues to accelerate, the public telephone network faces change. A mainstay of the global economy for the past ten decades, it is losing ground to a network of worldwide IP communications integrated with simple, secure, standards-based, applications-rich implementations and services.

There can be little doubt of the important role that SIP will play in facilitating this transition and enabling powerful enterprise applications that reduce cost, improve user productivity, and strengthen customer interactions. Some vendors will be slow to appreciate and take advantage of this opportunity. They may focus their energies on attempting to retrain the transition rather than exploring its possibilities.

Organizations should look to partner with companies that embrace innovations and standards. They are a foundation on which a company builds business-enhancing solutions. The question is no longer why, but rather when SIP will be integral to every business' future. IT

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QoS Comes First

Anticipation of eventual widespread IMS (IP Multimedia Subsystem) rollouts, and even the current migration to IP among many networks, are changing the way service providers, operators, and solution vendors approach the issue of QoS. Granted, there is no lack of skeptics who question the inevitability of IMS adoption, but it really has the potential to answer many demands of both consumers and service providers. As long as IMS is deployed properly, we can assume it will account for a large portion of the communications market. For those specialized "best-of-breed" vendors who intend to live up to that classification, now is the time to ensure that a solution is available for operators to guarantee premium user experience from Day One of their IMS rollout. In the process of developing an appropriate solution, it has become clear that the new network and OSS architecture is only one of many changes involved. Ensuring QoS over IMS requires not only new software, but a new strategy.

...service assurance for IMS must be a proactive and customer-centric operation.

Before the IMS network structure even enters the picture, the first issue to deal with must be the new market conditions and business model, and the goals of all players involved. Consumers are essentially looking for variety and flexibility. Naturally, this entails more varied subscription plans and service bundles, and the ability to tailor these options to fit their own personal needs. Furthermore, consumers expect flexibility in network access - both in terms of method (mobile, wired, or FMC) and location (roaming ability) - while receiving the same QoS as on their home network.

For providers, IMS first and foremost provides the ability to answer the consumer demands listed above. By bringing standardizing and centralizing control for all communications services as well as access methods, service providers will have the ability to rapidly deploy new content and services to keep pace with demand. In addition, the IMS structure allows operators to accomplish this over a converged IP network without redundant "silo" infrastructure, improving overall network efficiency and reducing costs. More importantly, IMS can enable service providers to build a new business model. With margins from legacy services flattened through tight competition, providers can leverage IMS to leave the role of low-value voice commodity and become a more high-value, one-stop-shop communications service provider.

However, providers must be careful in the way they pursue their IMS rollout. Dependable QoS is central to the IMS business model, and any shortfalls in that respect can bring major setbacks in IMS adoption. Currently, many providers are planning to initially focus almost exclusively on provision and billing technology, while viewing service assurance as a priority for later stages. It can be very tempting to many service providers to prioritize their resources in this way, with the goal of being able to begin service deployment and billing as soon as possible to grab early market share. If QoS is not ensured from initial rollout, consumers may be slower to sign on, or the high-value image of IMS may be tarnished, reducing it to another low-value commodity.

So, if service providers need to guarantee QoS from day one, how can this be accomplished? Many service assurance technologies and methods which are currently deployed over legacy networks are not sufficient for the needs of an IMS provider. First and foremost, service assurance for IMS must be a proactive and customer-centric operation. That is, the goal is no longer simply reducing repair time following service degrada-

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By Duby Yoely

tion or a lapse in network availability; if subscribers are paying for a high-value service, they expect better. In order to achieve the full value of IMS, providers will need a strategy to implement automatic corrective action and prevent potential degradation before the user experience is affected.

One cornerstone of a proactive service assurance strategy which has recently seen increased adoption among operators is the concept of service impact analysis. Essentially, this analysis tool allows operators to take any actual or potential network event, and map forward its effects over various users or services. Once this information is available, the OSS can issue recommendations to preserve QoS among the affected users. This means that rather than simply correcting problems according to network element, they can be addressed in a way that will have maximum impact on user experience.

In fact, this service impact analysis has another implication, which also applies to silo networks as well as IMS: a more business-oriented approach to QoS. Granted that any operator must suffice with finite resources, the information provided in this fashion allows an operator to prioritize their network resources for maximum impact on revenue and reputation, in addition to overall user experience. For example, if an operator sees that a given network event users of different subscriber classes, it will be possible to dedicate resources to first guarantee that premium SLAs are upheld, and then to address other priorities in a more deliberate and controlled manner.

Another important aspect of service assurance for IMS networks is to provide a unified network view for "end-to-end" service assurance. Under legacy silo networks, it was sufficient to simply monitor a series of disparate Element Management Systems or Network Management Systems, which would indirectly result in satisfactory service. However, the complexity of IMS networks would overwhelm this method, with too many variables contributing to the actual service quality. A preferable approach would be to manage operations across multiple domains under a single platform, and in particular to unite fault and performance management operations, thereby providing a more coherent picture of the network and connecting user experience more directly to network management.

A true end-to-end service assurance solution does more than just unite fault and performance management. It also captures a network view beyond its internal operations, providing data that reflects the real service quality as experienced by the end-user. In IMS networks, this can be accomplished by leveraging data contained in Diameter CDRs. Diameter CDRs were developed to provide data for both policy and charging purposes and therefore in IMS networks the service assurance system should be able to take advantage of this resource as well. In fact, even among currently deployed IP networks, the more advanced service assurance systems have managed to incorporate data from a variety of sources, such as SNMP agents on network resources, testing probes, EMSs and NMSs, and various xDRs, such as IPDRs and SIP CDRs, yielding positive results. In both cases, this data can be aggregated and correlated to measure any number of customizable KPI/KQIs, which directly reflect the user experience.

Any service assurance strategy for IMS networks must also include an increased emphasis on traffic monitoring, trend analysis and planning. Among legacy networks, the silo structure kept traffic simple enough that almost all network faults were hardware-related in their root-cause. However, as previously mentioned, one of the key benefits of IMS is the flexibility for subscribers to access and use their network services wherever and however they wish. This new level of flexibility, combined with the increased bandwidth demands of many services, magnifies the impact of traffic on overall network performance. Even in today's pre-IMS environment, we already see that congestion accounts for a significant portion of service degradation.

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Operators need a proactive strategy to maintain QoS in the midst of such complex traffic; retroactively managing the network to "unclog" congestion after service degradation is already felt will not be enough for the IMS subscriber. In this case, historical data on past traffic patterns must be analyzed through advanced correlation techniques in order to anticipate traffic-related faults before their effects reach the end user.

Finally, an important feature of any service assurance solution among IMS networks must be bidirectional mediation, which can almost give the effect of an automated Network Operations Center. Given the complex network operations that will fall under centralized control following IMS rollout, engineers' manual corrections may be too slow, even if aided by OSS recommendations. Once the data from service-impact, traffic, fault or performance monitoring systems reveal potential service degradation, the ability for the service assurance OSS to initiate automatic corrective action can greatly enhance end-user experience. In fact, even if such corrective action is not yet necessary, such an automated solution could trigger complementary active service monitoring to pinpoint the malfunctioning phase within the whole IMS service delivery chain, adding to the proactive nature of the operation.

Premium service quality is an integral part of the user experience that consumers expect from IMS. A major incentive for service providers to deploy their own IMS networks is to offer a high-value, all-inclusive communications service. If concept of IMS is to truly succeed, and avoid becoming another commoditized bit-pipe, then it must be provided with a high enough quality to justify its high value. If service assurance is left for a later priority, providers may still deploy IMS networks, but its market share may be narrower or simply at a lower value. Only a proactive and customer-centric service assurance solution, incorporated in the initial stages of rollout, can uphold QoS at a level which will allow successful market adoption under a high-value framework for IMS networks. IT

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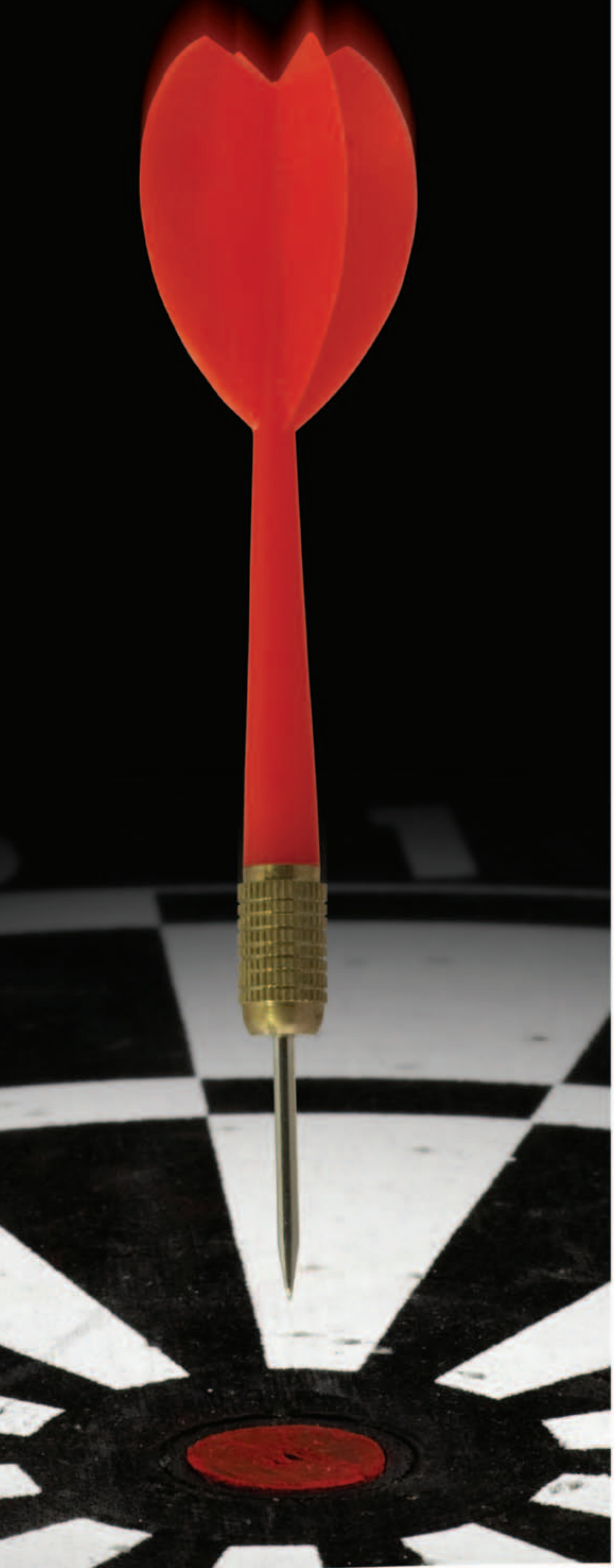

Delivering QoS for End-to-End VoIP Service Quality

Advertising buzz terms for voice over IP such as "it's the network" and the "sound of a pin drop" underscore the public's interest in voice call quality. One provider of over-the-top VoIP services even claimed recently to provide "business class" quality.

While such an achievement is admirable, it is nearly impossible to make such claims regarding QoS without the ability to manage the subscriber's broadband connection. At a minimum, the entire network must be engineered for end-to-end quality of service for the application being provided.

Quality of service for advanced IP applications can be measured first by the subscriber's experience with the application (be it voice, video, or gaming) and then by the application's specific transport flow specifications such as latency, jitter, and bandwidth. For example, for a high-quality VoIP phone call, the latency and jitter must be tightly controlled (approximately < 1/100 second) so that VoIP data packets can be appropriately reassembled by the application. For real-time video such as IPTV, sufficient bandwidth is required to transmit the data packets, and the jitter must be less than the buffer capacity.

Table 1 shows examples of applications and their associated resource requirement (e.g., flow specifications).

Taking this concept one step further, the intention is to quantify the quality of subscriber experience - called the QoE - and correlate the result to the necessary flow specifications. In the world of VoIP, the QoE is oftentimes defined by a mean opinion score (MOS), a number from 1 to 5 determined by surveys of actual human interaction with the media. Today, numerous tools allow service providers and subscribers to electronically determine the quality of a broadband connection via its associated impact on MOS.

Application	Real-time	Flow type	Typical data	Latency	Jitter
VoIP	Inelastic	UGS	16-96 kbps	< 10 msec	< 2 msec
Videophone	Inelastic	rtPS	300-700 kbps	< 10 msec	< 2 msec
Shooter game	Inelastic	rtPS	< 20 kbps	< 50 msec	<10 msec
Bulk file transfer	Inelastic	nrtPS	10 kb – 10 MB	< 250 msec	
E-mail	Elastic	BE	User defined		
Web video	Inelastic	DS	750 kbps	< 5 sec	
SDTV video	Inelastic	DS	1.5-2 Mbps	< 1 sec	
HDTV video	Inelastic	DS	4-5 Mbps	< 1 sec	

Table 1. Application-Specific Network Resource Requirements.

In addition to the MOS score, subscribers are sensitive to service uptime and busy signals, which also contribute to quality of service. Service uptime is a reflection of reliability: The standard is called five 9s, or 99.999% and is best defined by the availability of network elements such as routers and softswitches. Therefore, while an over-the-top VoIP service provider may be able to guarantee the reliability of its own softswitch, it cannot make any commitments regarding the availability of the broadband network's routers.

On the other hand, while a busy signal may indicate that the network is down, it may also be a good sign that the service provider is ensuring call quality by blocking calls when network capacity is reached. Service providers design networks with the Erlang traffic engineering model in mind to comply with their subscriber service level agreements (SLAs). The Erlang model accounts for available network capacity and the resources required for each call (as previously defined by MOS). The SLA can be managed either one of two ways: by over-engineering all the network resources, and/or by dynamically enforcing the SLA using a policy-based approach to network management.

This returns us to this article's original premise that it is nearly impossible to make QoS claims without the ability to manage the subscriber's broadband connection. The ability to over-engineer the network or apply a policy-based approach to manage SLAs is available only to a VoIP or content provider with access to the network.

By Jay Malin, Ph.D.

One other approach - albeit lacking a service guarantee - is the implementation of a more robust audio and video CODEC that minimizes the network resources required to deliver the service. For the service provider, the choice comes down to cost.

In the case of the over-engineered network, the cost is raw broadband bandwidth, which for an MSO means splitting a node or, in a wireless network, adding a sector (or spectrum). Sometimes, this can be performed easily; however, the cost, in a world of proliferating bandwidth usage - is prohibitively expensive. Adding capacity without reserving resources for walled garden applications means that service providers who "own" the network not only cannot guarantee the QoE, but also cannot competitively monetize the value of their plant upgrades because the "all you can eat" bandwidth is available even for over-the-top content.

On the other hand, by dynamically managing the allocation of network resources for premium advanced IP services, service providers can both monetize the value of the premium service and most efficiently exploit their capital plant. Policy management combines admission control and real-time, parameterized resource allocation in order to manage the delivery of IP applications such as VoIP (define - news - alert) and video. The figure below shows how a policy management architecture can be implemented for a broadband VoIP service. The admission controller enforces the traffic engineering model by comparing the number of concurrent application sessions against the network's capacity. If the resources required to ensure the application's performance comply with the SLA, the application (e.g., voice call) is blocked. This function can be reasonably performed in most IP networks. (See Figure 1.)

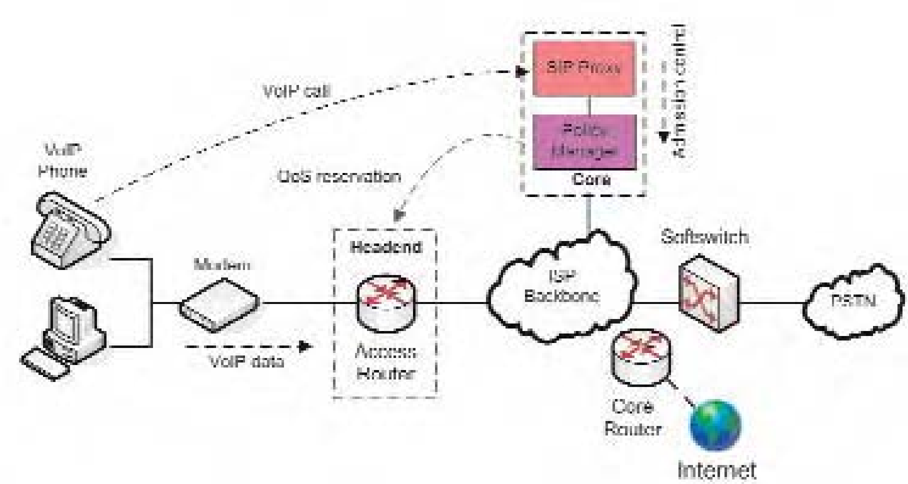

Figure 1. Policy Management Architecture.

While many networks are being upgraded to add real-time bandwidth (and latency and jitter) allocation capabilities, cable and WiMAX stand out above the rest. The standards committees overseeing these networks have incorporated dynamic, parameterized QoS capabilities into their specifications. This means that when an application such as a voice call is admitted by the admission controller, the precise resources are allocated at the applicable router regardless of whether it is a CMTS or base station. Application-specific data packets are scheduled and handled in the network routers based upon both priority and resource requirements. Similar types of support are being added to DSL aggregation routers, WiFi access points, and 3G packet gateways.

In an alternate, less precise approach, data packets are marked by network elements for prioritization throughout the network. As long as it is combined with an admission control function and the markings are coordinated across all applications, the SLA can be reasonably enforced. However, because any packet can be marked by virtually any subscriber, this approach is not completely trusted and is not nearly scalable as the former approach.

QoS is most often addressed in the access network due to the competition for network resources. In the access network, many subscribers contend for limited network resources, thereby causing the congestion predominantly responsible for poor QoS. While the backbone is not completely immune to resource limitations, it is over-engineered to accommodate significant traffic. Network management technologies such as MPLS and DiffServ can be mapped to access network QoS policies to facilitate an end-to-end managed service.

End-to-end quality of service can be delivered only when the application provider has access to the network. In order to reliably deliver a high-quality IP service such as VoIP, service providers must incorporate an adequate traffic engineering model into a policy-based admission control engine. By offering a premium delivery service to subscribers who cannot be otherwise provided by a third party, service providers can monetize the value of their network while managing subscriber satisfaction. IT

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Hosted IP Telephony Solutions

Small and medium-sized businesses (SMBs), particularly 100 users and below, have for the past several years dramatically streamlined their operations to remain competitive. For businesses running on paper-thin margins, there are few new projects that their small IT groups can possibly take on, such as VoIP. Such organizations are turning more and more to hosted IP Communications solutions. The benefits of outsourcing VoIP (or even a full-blown unified communications package) can be greater than simply saving some money.

Many of these services are critical to organizations. We all know of enterprises with email systems that go down on the last few days of a quarter, which usually portends that the email system will be outsourced by the end of the next quarter, since email has become too important to place in jeopardy.

For companies of just about any size, an SLA [Service Level Agreement] from a service provider will always be better than whatever uptime a company can create for itself, unless they spend money lavishly. Even the smallest service provider has multiple redundant systems with automatic failover, and on top of that it's probably using the latest and greatest hardware, which the small business can't or won't have available. Moreover, service providers generally do joint tests with most of the partners they deal with to deliver their applications to customers. Microsoft, for example, has defined a fairly rigid test for hosted Exchange. VoIP vendors such as BroadSoft have similar tests as well.

Hosted services will over time turn into commodities. (After all, anything you can sell on a line-by-line basis is destined for mass-marketing.) Look at web hosting today, or email. They are all commoditized and are provided at very low cost. However, what makes things such as hosted VoIP or hosted Exchange interesting is that you can build an ecosystem of value-added tools around them. This gives the enterprise additional productivity-enhancing features and it allows the service provider to maintain a much longer product lifecycle and it can therefore sell the service(s) at a premium for a longer time.

Product vendors have taken stock of the ripe hosted services market and have altered their product lines to facilitate connections between service providers, SMBs and "vertical" organizations. For example, early in 2007 ZyXEL Communications (news - alert) (http://www.zyxel.com), the world's fourth biggest DSLAM vendor, announced a new SMB VoIP Integrated Access Device and SMB VoIP router that can bridge the move to VoIP for growing businesses. Both products enable multi-line VoIP service through one of two options: by hosted VoIP services or by integrating with an existing analog PBX system. Both of these devices are designed to ease deployments by service providers.

ZyXEL's VOP1248G a 48-port POTS/VoIP line card has a media gateway that converts analog voice to VoIP, thus eliminating the need to install special VoIP phones or an analog telephone adapter (ATA) on customer premises. This new POTS/VoIP line card works in conjunction with ZyXEL's IES 5000 and IES 5005 chassis-based Multiple Service Access Platforms (MSAP) and is targeted to telcos looking for a easier way to offer a VoIP architecture to customers. ZyXEL's IES 5000 and IES 5005 MSAPs provide support for ADSL 2+, G.SHDSL.BIS and VDSL 2 line cards and are designed for telcos wanting to offer multiple services over the same box.

A Firm Foundation

Determining just how inexpensive, feature-laden and easy-to-use an IP service is depends primarily on the quality and scope of the platform used by the service provider. One of the first widely deployed and well-known delivery platforms in this area is Ensim Unify by Ensim (news - alert) (http://www.ensim.com).

Co-developed with Microsoft, it enables service providers to centrally create, control, and deliver hosted IP and application services. The latest version of Unify includes support for Microsoft Hosted Messaging and Collaboration (HMC) and Windows-based Hosting (WBH) Technology, support for the Microsoft Connected Services Framework (CSF) 3.0 , a New Resource Selection Policy Feature, and it provisions many services such as BroadSoft and Blackberry that are not available as part of the HMC solution. There's also the built-in ability to add custom services via an SDK.

Francois Depayras, Vice President of Sales and Marketing for Ensim, says, "Ensim is a software developer that provides software solutions for service providers. We really focus on the provisioning and automation of services. For example, if you want to deploy a Voice-over-IP line, there are literally 100 steps for a manager at a service provider to deploy such a line. Our technology completely automates that process through templates so that the service provider can create silver, gold, and platinum packages for different types of VoIP lines, and then enable those completely automatically. That can happen straight from the storefront to the end customer, or at least if it's a large enterprise, it can be done with an agent over the phone, and with one click of a button you know that a service has been provisioned from A through Z without any mistakes. If any errors somehow occur, there's an automatic rollback and accompanying notification to the administrator."

"So, in a sense, we offer something of a service delivery platform. Granted, not to the full extent of a large service delivery platform, but we really do focus on the automation of services provisioning," says Depayras.

"The heart of our platform takes care of the monitoring, metering, the service resource management, and so forth, to ensure that the application will be run as service by the service provider," says Depayras. "We also offer a pre-built control panel or self-care management tool at delegated levels, so the service provider can now have access to a portal that offers them a wide range of choices where they can do such things as create different types of mailboxes, VoIP lines and so forth, and really manage all of their next generation services. The reseller also can also access the same control panel, but based on their role, they have different capabilities available to them. Our control panel can be shared with the enterprise administrator and the end user. So the goal really is to make sure that the users should not have to call the service provider everything time something strange happens, they should be able to self-manage fairly efficiently, and that's done through providing a graphical user interface [GUI] that's very user-friendly, rather than a command line interface, for example, which can be very complex."

"Ensim's on-demand platform offers pre-built connectors for about 15 applications today," says Depayras. "We have connectors to BroadSoft, Siemens OpenScape, Microsoft Exchange, IBM SharePoint. We create much more than VoIP service - we can create a unified communications package. One Ensim Unify is set up at a service provider, it's very easy for the provider to add a connector or remove a connector, thus making it very easy to test the market with new services at minimal expense. The same interface and provisioning system can be used."

"As for where this industry is going," says Depayras, "I firmly believe that many companies decide to outsource solutions as important to everyday business as VoIP or email, since there's no sense in keeping some of it in-house, unless it's necessary for there to be some kind of tight integration to an HR form system, for example. Outsourcing everything at this point makes a lot of sense."

More and more, enterprises of 250 employees and below really don't have the IT resources in-house to accomplish VoIP on their own...

"Why a provider should deliver hosted VoIP is fairly obvious in terms of where the market is going," says Depayras. "More and more, enterprises of 250 employees and below really don't have the IT resources in-house to accomplish VoIP on their own. Thus, there's a wave of outsourcing to the largest telecom providers as well as regional service providers, that essentially transforms them into an IT resource for small enterprises. They ensure that the SMBs have strong SLAs, strong archival systems, and so forth. For a service provider, then, the market certainly exists. But service providers have the bad habit of building a platform for each service they deliver, which is extremely inefficient at the end of the day, and which in turn forces them

5 Tips for Selecting a Hosted VoIP Provider

by Joel Maloff, VP of Products, GlobalTouch Telecom (http://www.globaltouchtelecom.com).

Selecting a hosted VoIP applications provider is not as simple as it appears. Understanding how your prospective providers compare will help you make an informed and hopefully successful decision. Here are five suggestions. . .

1. Feature Suite - If you are seeking to offer VoIP services from a hosted applications provider, you have already realized that building such systems yourself or purchasing the systems from others is an expensive proposition. What features and services are important to you? Are you interested in providing consumer VoIP services? Pre-paid and post-paid? IP PBX services? IP Trunking? Do you need assistance in handling Direct Inward Dialing (DID) numbers, E911 and CALEA? Some vendors offer hundreds or even thousands of features. Which ones do you really need and which are you and your customers willing to purchase? Before you start comparing hosted PBX vendors, know what features and services you want. Prepare a list of required and "nice to have" features.

2. Underlying Technology - Not all VoIP systems are the same. Some offer better voice quality but require greater bandwidth. Others are deigned to offer superior voice quality but consume less bandwidth and therefore are available at lower cost. Some will support video calls whereas others may not. Some may be more secure than others. Ask questions and demand answers from your prospective vendors.

3. Support Services - What support services can you expect from your provider? Are you expected to provide Tier 1 support with the vendor offering Tier 2 and 3 support? What response times can you expect? What reports will be available to you? These are critical areas in managing and growing your business.

4. Scalability - Can your vendor continue to grow with your needs, both in terms of capacity as well as new features as they emerge? This may be a direct result of the underlying equipment used by the vendor. If it is an open architecture and designed to support emerging standards, you may be better off than a closed proprietary system that may not be interoperable with future developments.

5. The Real Cost - Make sure you consider not only the set-up and monthly recurring costs but also any additional costs that might arise. These include bandwidth fees, custom development charges, updates or new features, support service "overtime," and additional personnel that you will need on your staff to coordinate with your vendor. Creating a "pro forma" that looks at all anticipated costs, the timing of these costs, and how they relate to your expected pricing and revenue generation models will help you stay ahead of the challenges.

Using a Hosted VoIP Applications Provider can be an excellent and timely choice. Clearly articulating the details in advance can make the experience all the more rewarding!

GlobalTouch Telecom, Inc. offers a vertically integrated VoIP platform. The company designed and developed every aspect of its technology from the ground up creating a one-stop single vendor VoIP solution. The product thus comprises an all-inclusive, private label (white label) offering for carriers, MSOs, Resellers, PTTs, ILECs, ISPs, CLECs and marketing companies. The platform can be completely customized and rolled out in 60 days or less.

to spend a tremendous amount of time on R&D to develop functionality that's already available off the shelf through companies such as Ensim."

"Hence, the goals for a service provider today is four-fold," says Depayras. "The four most important things a service provider should look at when going to market with a hosted VoIP solution, or even any kind of next-generation hosted service is, first and foremost, the time-to-market. It's easy to say you want to get a VoIP solution out quickly, but it's more than that. It's being first to market or as quick to market as possible, for the application or service that the provider wants to launch. But it's also about being the first-to-revenue and getting to generate revenue as quickly as possible. The system needs to be up and running targeting small-to-medium sized enterprises, which requires resellers because modern enterprises are geographically dispersed and very fragmented. Some enterprises are vertical, others are geographical in nature. It's important to have as wide a range of salespeople and resellers to be able to go after them."

"Fortunately, a provider can be first-to-market with our Ensim platform and it can be used to plug in any additional services, using the same functionality that's built in, utilizing the same back-end and control panel or self-care management tool," says Depayras. "Thus, there is very little training of support or operations, and there's no explanation that needs to be given to existing customers because we're basically just adding functionality over time. So, it's hosted VoIP the first day, and then delivering additional services on the VoIP line such as E911, and then three months later you can add collaboration capabilities through SharePoint, for example."

"Second, there's the ability to have a channel-enabled strategy," says Depayras. "The SMB market to me is really the El Dorado of markets. We know that it's a $12 billion market today, and everybody knows there's a pot of gold lurking there, but it's extremely difficult to reach it, simply because the fragmentation of the industry and that SMBs are really geographically dispersed and each vertical has specific needs and requirements. It's

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very important for the service provider to affiliate itself with resellers that can cater to existing customers."

"Third, decreasing the cost of the operation is absolutely crucial," says Depayras. "It's fair to say that automation of provisioning and self-care of end users adds a dramatic aspect to all of this. Our approach usually decreases the cost of operations by at least 50 percent. Many of our existing customers are now able to place their IT or operations people on other, more important projects since everything doesn't have to be managed manually - it's all now automated. For example, 50 or 60 percent of support calls are people saying 'I've forgotten my password'. For anybody who has a support organization in the U.S. or Europe, that phone call is going to cost between 10 and 20 dollars or Euros. That's a huge support cost just for something as trivial as a password."

"When a service provider looks to launch something such as hosted VoIP or messaging or collaboration, the platform is obviously crucial but it's also just as important to have an ecosystem of tools around the solution that are really created to enable and empower the end user to do things on their own, such as to self-configure their VoIP phone or self-configure their mobile device with Exchange," says Depayras. "All of these little tools involve a little click on a self-care management section of a web page, rather than calling the administrator or service provider."

...decreasing the cost of the operation is absolutely crucial...

"The fourth aspect focuses on extensibility," says Depayras. "Looking at hosted services today, some are extremely successful, some not that much. Some are perhaps ahead of their time, the marketing message may be the wrong one, some people still don't really get why they should outsource. It's always an expensive gamble for a service provider to create a platform to provide hosted services. For example, the appropriate infrastructure needs to be built. The provider may decide to build its own provisioning or control panel and management center, and so forth. We've generally seen service providers launch next-gen services on their own over a 12 to 18-month period. That's a long time. But using tools such as our Ensim Unify software, customers have been launched in as little as ten days, and the average is about a month-and-a-half. We can quickly and inexpensively provide everything in that last mile between the service provider and the end user: the control panel, management tool, automation and provisioning." IT

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

Streamlining Service Creation

Service Creation is another one of those overly broad terms that covers everything from provisioning to the ability under IMS [IP Multimedia Subsystem] for somebody with a bright idea to create yet another possible "killer app" service.

Service creation and provisioning at telcos and cablecos always had a sort of 'garage shop' ambiance. Now, however, some genuine, rigorous "standards and practices" are finally permeating the industry; these will enable a sort of "service factory" approach to assembling services and provisioning them quickly and inexpensively.

In particular, one should take note of the Product and Service Assembly (PSA) Catalyst Initiative, led by a consortium of companies such as Axiom Systems (news - alert) (http://www.axiomsystems.com). The PSA is a collaboration of vendors and service providers who are creating an IT reference architecture that 1) streamlines the next-gen network product/service lifecycle, 2) bridges the service creation gaps between OSS, BSS and service execution, and reduces the cost of service/product production.

Service creation and provisioning... always had a sort of 'garage shop' ambiance.

Simon Osborne, Axiom Systems' Senior Technical Architect, says, "Service creation means different things to different people. I guess the most confusion about service creation involves the execution plane - creation in the bundling and product offerings. The PSA initiative is one that Axiom and many other partners have been working towards within the umbrella of the TMF [TeleManagement Forum]. PSA is a reference architecture to support the rapid assembly of services. You could argue that it's service creation in a bundling/order fulfillment-type metaphor. The PSA architecture is aligned to the TMF in terms of the way they model their architecture through the eTOM [Enhanced Telecommunications Operations Map] and interfaces associated with TAM [Telecommunications Applications Map, developed to provide a high-level decomposition of apps into functional groups, so that similar, reusable functions can be grouped together and reside in well-known service boundaries] and things like that."

"We're using that landscape as inspiration," says Osborne. "To support the PSA initiative, there's a PSA Catalyst project in two phases that showcases the ideas, deliverables and benefits of the PSA initiative for telecom operators and service providers as they move forward. The first phase was demonstrated at TeleManagement World Americas 2006 in Dallas, Texas, in December 2006 and the second phase of the PSA Catalyst was demonstrated at TeleManagement World in Nice, France, in May 2007."

"These Catalyst events helps establish a set of vendors working within the PSA architecture," says Osborne, "which is a reference architecture of the federation and cooperating catalogues within the architecture, all focused on 'How do I the service provider discover what my network can do? And how do I bundle and present that in a logical model and an aggregation model, ultimately propagating that into the order handling and CRM systems to allow customer orders to be fulfilled against them?'."

"The deliverables from the first phase of the PSA Catalyst are an industry-standard API to allow the participating members within this federation of catalogs to promote their capabilities, all the way up the stack," explains Osborne. "Equally, we now understand the rule dependencies, the rule sets, and the relationships between and among those building blocks so that validation can take place and service assembly can rapidly take place."

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By Richard “Zippy“ Grigonis

Osborne goes on: “The second phase of the PSA initiative is that we’ve taken onboard some new vendors and we’ve got more service providers as well, and QinetiQ, one of the world’s leading defense technology and security companies, who are all providing valuable use-cases around their infrastructure and their service play.”

Osborne admits that, “We’ve acknowledged that perhaps in the first phase there was a bias toward the service delivery or fulfillment aspects of this architecture. So, in recognition of that we’ve brought in Convergys as a partner to understand the emphasis that billing places on this same componentized architecture when subject to service component assembly and service modeling. At the product catalog layer, we’ve brought onboard TIBCO. Even Microsoft is now onboard.”

“The whole value of the PSA architecture is to allow multiple vendors to be able to plug into it by using the common interface and API,” concludes Osborne.

Hitting “Bedrock”

JacobsRimmell (news - alert) (www.jacobsrimmell.com) is an operations support system (OSS) vendor that started when Founders David Jacobs (CTO) and Phil Rimell (chief architect) left Reuters’ IT department nearly ten years ago. Among other things, it’s now the principal service fulfillment provider for cable’s 800-pound gorilla, Comcast.

JacobsRimmell’s Vice President of Marketing and Strategic Alliances, Joe Frost, says, “We’re known for our OSS and provisioning abilities in the MSO space. Comcast is our biggest North American cable customer. We’re deeply involved in Comcast’s ’Bedrock’ platform. At the moment we do all of Comcast’s data, IP data and application provisioning technology for Comcast VoIP or ’digital voice’. We’ve just started to move the digital TV customers onto the Bedrock platform too.”

Bedrock is Comcast Corp. new multi-service operations framework, created in part to deal with the wider range of services made possible by IP. Indeed, Bedrock enables automated end-to-end service fulfillment and provisioning of all Comcast services - video, VoIP, data and eventually wireless. Tests indicate that in terms of the high-speed Internet service alone, for example, Comcast was able to reduce time-to-install by a third thanks to Bedrock, from 12 minutes to about eight.

A key partner in development of the Bedrock platform has been U.K.-based JacobsRimmell. As JacobsRimmell’s Joe Frost says, “For the last four consecutive quarters, Comcast has set records for the number of new subscribers they’ve been signing up. It’s something like 10,000 new voice users a day going through our platform. It’s quite a successful implementation.”

“Basically we see three primary markets,” says Frost. “First is the Cable business as usual. Second is the commercial services market which encompasses both cable and wireline telcos. The third market is the information management space. That really reflects the fact that most of the operators, specifically the incumbent tier-1 telcos, are trying to implement a lot of new matched or blended services - or whatever you want to call them - using our methodology, along with the expectation of new technologies. In that respect, most of them have recognized now, particularly while they are in lab trials, that the technology is capable of all sorts of cool things, but in order to be able to develop and then offer services that may be very timely or ’seasonal’, their whole operational infrastructure needs a significant upgrade.”

Frost elaborates: “The operators realize they must have far better control and visibility of their operational data; that’s the data that involves the subscriber’s identity. It’s not just the telephone number or bill payer’s street address any more; it’s the actual user that is entitled to access the service and in particular it’s the context in which the user sits - they can take their services over any network, any access technology, any device. After all, that’s the whole premise behind IMS [IP Multimedia Subsystem]. One then maps the entitlement of those new products and services in real time, because many subscribers, particularly the younger ones and business types, tend to frequently move around the landscape. They change how they take their services frequently, and they use different types of devices to partake of those services. ”

Operators can’t afford to send out engineers when you’re looking at a potential revenue of perhaps $30 a month...

“The second trend I mentioned, commercial/business services, is a field in which we made our first announcement recently,” says Frost. “What we saw there was an opportunity to develop an off-the-shelf provisioning or fulfillment platform. We at JacobsRimmell have observed a massive ’land grab’ going on right now among the Tier-2, 3 and 4 telcos and the big cable companies who see a huge opportunity in taking revenue away from the wireline incumbents and delivering managed voice services to small and medium businesses [SMBs]. It’s a market the incumbents are not as focused on as they perhaps should have been all along. But one of the biggest drawbacks is that, following business services, particularly business VoIP services, in this market, you see that operators’ rates are dropping incredibly quickly. It’s almost become a commodity offering. Because of this, the operational system must be far more efficient and there’s a much wider range of equipment needed, particularly customer premises equipment.”

"Today's trend is that operators deploy SIP [Session Border Control]-based user equipment at the customer premises," says Frost. "That ranges from the standard desktop handset through the small and medium office IP PBX, such as the Linksys Voice System 9000 SIP PBX that retails for less than $400. One of the big issues for the operators is that these devices don't just pop out of the box and start working immediately. They need an IP address, they need their features configured, and so forth. So what we've seen from the operators and vendors alike is a need to be able to deploy more complex equipment across a far wider range of customer premise sites, and have everything work the first time. Operators can't afford to send out engineers when you're looking at a potential revenue of perhaps $30 a month for a phone service. The equipment and service must be configured configured and dispatched from the operators' warehouse or a pre-configuration location."

"That's why we've put together an off-the-shelf provisioning package called QuickStart Business VoIP," says Frost. "It's a full-functioned, end-to-end fulfillment or provisioning platform. It handles all of the office equipment, softswitch, VoIP switching infrastructure, the messaging infrastructure and so forth, and handles all of the order management. It also deals with all of the SIP-based customer premises equipment in one application so that the call agent or field technician needs just one screen from which to work. It handles the whole process from assigning a new customer, taking an order and assigning products to that customer, and then configuring the physical infrastructure or equipment that delivers the service. It's a real rapid implementation off-the-shelf type of solution not normally found in this industry today. We've seen an opportunity here - customers simply haven't been able to configure and adapt their equipment quickly enough with conventional techniques, because it's been too complex to do using existing tools."

All in all, it appears that the days of being able to pick up a phone and watch a field technician visit your premise and install equipment and services is coming to a close. IT

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Delivering Reliable Quality of Service

Quality of Service (QoS) has been the bugbear of IP Communications since its very beginnings. Fear over call quality (or lack thereof) slowed the adoption of IP by both providers and customers. Various techniques were proffered to maintain voice and video quality: overprovisioning of bandwidth, dedicated bandwidth (private IP networks), prioritization of realtime voice and video packet traffic, router protocols to signal the location of network congestions points, and so forth. Today, however, many providers and experts claim that even the public Internet offers more-than-acceptable QoS characteristics for voice, video and multimedia.

Ajay Joseph, Vice President of Network Architecture and Engineer for iBasis (news - alert) (http://www.ibasis.com), says, "We are a VoIP company with a presence in roughly 120+ countries. We have a wholesale business where we sell to carriers and a retail business where we sell prepaid disposable and non-disposable calling cards. Carriers connect to us with both IP and TDM methodologies. On the retail side, we use the wholesale network to terminate the voice calls. All of this is transported via IP across a core network. Interestingly, we use the Internet quite successfully to transport voice."

"Interestingly, we [iBasis] use the Internet quite successfully to transport voice."

"Some of the challenges that iBasis initially encountered years ago involved the public Internet," says Joseph. "The quality of service was affected by unpredictable congestion points that would appear. We would have to take carriers in and out of service, depending on the quality of the network. Maintaining the system was a very manual, time-consuming effort, and it just didn't scale."

"So we put several things in place," says Joseph. "First, since we used the Internet, as a policy we tried to minimize the amount of public and private peering that we use across the network. That's because, if you look at the Internet, there are many various ISPs that connect to each other. These ISPs are connected through peering points, which allow for bilateral connections between themselves, or they could also go through what are called public peering points. If congestion occurs at these peering points, it gets taken care of by the ISP, which means it's out of our control. One of the decisions we made was to design an virtual IP network so as to avoid both private and public peering points. So, between all of our POPs [Points of Presence] we have a virtual backbone. We don't own the whole physical network, but we connect to ISPs as customers and we then dynamically define routes using BGP [Border Gateway Protocol] the IP routing mechanism, so as to keep the iBasis traffic away from the peering points. All of the peering that does happen is through the iBasis cloud itself, not through these public and private peering points."

"Now, if any kind of capacity upgrade must take place," says Joseph, "we at iBasis manage it, not the ISP. We've been running this for at least seven years now, and the quality we get using this technique is very good, very clean and clear. The packet loss is close to zero. And packet latency is low too."

Joseph adds, "We've also observed two interesting phenomena: First, the general quality on the Internet has improved tremendously over time. Second, the price of bandwidth has dropped in terms of dollars per megabit-per-second. What all this means is that our original decision to use the Internet was a good one. It reduces tremendously our cost of building out the network."

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By Richard "Zippy" Grigonis

"So we've got millions of calls entering the network at any given point," says Joseph, "and all those calls come into the network and have to reach their respective destinations. By the time a call reaches the far end across the IP cloud, it could have gone through congestion points anywhere in the world. Fortunately, in my shop we have a software development group that develops the software that handles the routing of the calls in the network. Given that we don't actually own the whole pipe, we probe every endpoint in our network which rides atop the Internet, and we probe the quality of the connections on a fairly frequent basis. The results of these probes are fed in real-time to the routing system. Depending on the quality of the IP pipe towards the far end, if the quality is very bad, then the call will not be routed there - it automatically gets taken out of the routing system. So, from an operations point of view, we don't have to do the manual activity of removing a call from routing because the network quality is bad. That works pretty effectively."

"Additionally, we've made other modifications," says Joseph. "Let me give you an example. Let's say a call originates in New York and it needs to terminate in Malaysia on the other side of the world, but the quality of the Internet between New York and Malaysia happens to be bad at that time. However, the Internet's quality from New York to Los Angeles is found to be good, and it's good from LA to Malaysia. So in real-time we're also looking at the different paths across the Internet cloud between all of the different endpoints and we figure out which is the best path through which to ultimately terminate the call. If the path from New York to Malaysia is bad, but New York to LA is good and LA to Malaysia is good, then we'll just force the call to travel via LA instead of directly to Malaysia. That maximizes the probability of completing the call, as opposed to saying 'Oh the quality is bad, we won't take any calls at this location for a while'."

"We also use session border controllers inside the network for transcoding and quality purposes," says Joseph, "and we've been running the system very successfully for about seven years now. Millions of calls traverse our network at any given point. We buy routes from our providers and they give us rates and coverage - such as to Malaysia. We have systems that look at the quality of the call that terminate across providers and we have threshold values that are formulated based on the provider, the coverage, and so forth. Based on a particular threshold, if a particular provider does not 'behave' well for a particular route, the provider gets shut out and goes back into testing. We have a testing system and a 'scrubbing system' that looks at the quality of the calls that terminate through a provider and if something's wrong, different routes are used instead of the provider in question."

"Thus, our quality verification methodology is quite extensive; it goes through the IP cloud up through the session layer and all the way up to the application layer," concludes Joseph.

At the heart of maintaining QoS are the related activities of testing and monitoring.

One major company in the field, Psytechnics (news - alert) (http://www.psytechnics.com) has helped promote the more encompassing term, Quality of Experience (QoE) to describe what its voice, video and multimedia solutions do. Psytechnics recently published a report (March 2007) wherein they applied their testing expertise to evaluating the voice quality of a pre-release of Microsoft Office Communications Server 2007 and Microsoft Office Communicator 2007 desktop VoIP solution. Their results revealed that Microsoft software-generated calls deliver superior voice quality to a single-purpose IP phone. As Psytechnics reports, "These findings show that the quality of Microsoft's offering is high enough to allow companies to integrate voice communications with PCs, which could eliminate the need to purchase expensive IP phones."

Psytechnics also is now heavily into the testing and QoE of the burgeoning IPTV industry. Psytechnics' Vice President of Product Marketing, Benjamin Ellis, says, "Two things have been keeping us really busy at Psytechnics. One concerns enterprises adopting VoIP, running into problems and then having us help them out. The other concerns IPTV, an area now becoming populated with many providers, all of which have now gotten over the excitement of finally get it working at all, and are now interested in getting IPTV to work well and efficiently."

"If you had asked me ten months ago, I would have said that the ways you provide a good quality of experience to voice and video were somewhat similar," says Ellis. "But recently they've both diverged. The real difference between voice and video actually has to do with where the content comes from. That sounds dumb, but for VoIP, the users are providing the content. Much of what we do there is looking at the 'total speech quality', which involves things such as looking at a waveform. With IPTV the content is less of an issue, but it is still an issue, and we do some decent things to maintain quality. But especially when the provider receives the video files, they do some assessment on them to check that they are of sufficient quality. Once you've taken care of the content quality, as it were, you don't have much to worry about it since once the video file is on the server it doesn't get corrupted. When it gets translated across the network as packets, what we do is what we call 'ingestion' to check that it still is of broadcast quality."

"We're also into ensuring content delivery to set-top boxes, and making sure people actually have a picture," says Ellis. "About a year ago, people were very focused on network tools, checking that the QoS was configured correctly to deliver the IPTV and those sorts of things. Then there was a huge fracas over what I was call 'signaling metrics' or how quickly a customer can change channels, and how long does it take a channel to come up on the screen once you've changed channels - those sorts of things. There are some reasonably-sized providers in Europe and they've realized that these areas they were most concerned about aren't actually in the problem space. Once you've deployed IPTV, all of those factors remain fairly constant."

"When you deploy IPTV, however, you get interactions in the network that you didn't expect, between different subscribers for example, or having network elements not doing what they were meant to do," explains Ellis. "It comes down to how well you're actually delivering the stream. That's great for us as Psytechnics, since that's the thing we major in measuring, by looking at the stream on a packet-by-packet basis and working out how well a packet gets delivered and if it didn't get delivered well, then were there impairment issues of a sort that the subscriber will actually notice? Or is it something that would simply be compensated by the set-top box? These are kinds of things that we examine and help resolve."

So it appears that while QoS will always be a cause for concern, both the modern Internet and QoS techniques should allay most of the perennial fears held by consumers, businesses and other organizations. IT

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

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Enterprise Mobility

Motorola's acquisition in January 2007 of the "wearable computer" company Symbol Technologies called attention to the increased mobility of enterprise employees who nevertheless still demand quick and easy access to business-critical information. Motorola says its new enterprise mobility business will provide "cutting-edge, end-to-end products and services, coupled with application-specific solutions from a vast channel partner network." The term "enterprise mobility" encompasses everything from the integration of cell phones into a corporate PBX system to vertically-oriented solutions involving the quick delivery of productivity-enhancing information to people in the field, the factory, the warehouse, at cash registers and at patients' bedsides.

Everyone, it seems, is getting into the enterprise mobility industry.

President and CEO Kathy Zatloukal of the aptly-named company, MobileAccess (news - alert) (http://www.mobileaccess.com) says, "Our infrastructure provides the most cost-effective approach for a total cost of ownership [TCO] for enterprise connectivity in buildings. In other words, a single investment allows the enterprise to enjoy the benefits of cellular-based applications,

The new spectrum... will really create the true wireless revolution within enterprises.

more premised-based applications such as wireless LAN, and it can also support public safety-type of applications. In particular, it supports applications used by verticals - one example would be wireless patient monitoring in a healthcare institution. There's a multitude of building information-type of applications that can ride on this infrastructure or even some security and surveillance applications. Basically, it's all about the TCO for a multitude of wireless applications in the enterprise."

"Most people believe that the wireless revolution has already happened," says Zatloukal. "However, if you follow what spectrum has been allocated and what new spectrum is scheduled to be allocated, there's really only a small portion of the spectrum available for applications that's actually been licensed. The new spectrum that will be hitting the market is actually what I think will really create the true wireless revolution within enterprises. If you follow the annual growth rates of fixed voice, fixed data, mobile voice and mobile data, you'll see that, for network operators in most developed countries, mobile data now makes up 20 to 30 percent of their revenues. It has started to make a significant shift in terms of where the focus will be of those particular providers of wireless solutions. A lot of this data right now is based on messaging traffic as well as early-on Internet-type services."

"When I reach out to CIOs and their enterprises, current mobile devices are creating an awareness and interest in mobile data," says Zatloukal. "These are devices such as notebooks, PDAs and smartphones. Some of these can handle applications where there are video, voice and data, and then there's the issue of the number of 'bands' supported. Complexity is definitely occurring. That's a trend in the marketplace that is creating the momentum around mobile voice and data. I believe what will actually create the desire and action by CIOs to really embrace it for enterprise applications - possibly even to the point of substitution for some wireline connectivity - will be when some 4G-type aspects appear, such as what you find in the advanced IP network architectures. Then you get into modulation technologies that really allow the performance and cost factors for the operators to meet an 'intersection point' on a graph that makes sense - for example, with OFDMA, or the Orthogonal Frequency Division Multiple Access modulation technology. Then you start to see a platform evolve that is truly ready for mobile data applications."

Over at Tango Networks (news - alert) (http://www.tango-networks.com), their

By Richard "Zippy" Grigonis

technology can fuse the features of your PBX with the mobility of using carrier wireless networks without resorting to expensive, customized handsets or having to change user behavior. Corporate IT departments can use Tango Abrazo to manage mobile phones as they would any other corporate asset, thus eliminating excess calling costs, productivity inefficiencies, inadequate call security, unavailable content monitoring and unnecessary legal exposure.

Al Leo, Vice President of Business Development and Sales, says, "We connect companies' mobile phones to the corporate PBX. Thus, those mobile phones integrate into the corporate-wide PBX-based network. It gives a user a single number, a single voicemail box, access to PBX-based features which are more robust and easier to use than the features typically associated with a cell phone, such as call forwarding, abbreviated dialing and conference calling. We extend the corporate dialing plan and corporate least cost routing algorithms out to that end user too."

"So, we give the end-user some productivity, acceptability and enhancing capabilities," says Leo.

Leo continues, "For the enterprise IT management team, we enable them to manage mobile voice and integrate it into the rest of their solution and business applications. They can now offer productivity, accessibility and enhancements to their end users. It also gives them the ability to manage the costs and functionality of mobile voice that they typically don't have in today's telecom environment."

"The final thing that our solution does is to keep the wireless carrier or integrated operating company front-and-center in the value chain with respect to enterprise mobility," says Leo.

"Our solution will work with any mobile phone," says Leo. "It's architected so that we put a network element, the Abrazo-C in the carrier's network and then we propose that enterprises that want to take advantage of this solution should then purchase the Abrazo-E as a CPE-based solution that becomes part of their PBX-based network. This approach allows us to work with all mobile phones, and there's no need for dual mode phones or client software on the phone. We basically eliminate the handset as a barrier to entry. Enterprises are transiting from TDM to IP and we've able to integrate with any type of PBX, leveraging the investment made in them."

"Additionally, we give the corporation the ability to create a mobility policy and enforce it as part of the solution," says Leo. "It's not just a piece of paper or expense report. We have a rules engine in the Tango Abrazo E that allows enterprises to define how and when people can use their mobile phones."

Location-Based Services

A discussion of Enterprise Mobility usually brings up the not-so-tangentially related subject of location-based services.

At NetMotion Wireless (http://www.netmotionwireless.com), John Knopf, Director of Product Management, says, "We don't offer what you'd call a traditional application. We offer software that's deployed by the IT or security department - it's basically a VPN, but it's a VPN that is specialized to ensure that mobile workers remain highly productive and that all of the data they're transmitting over the wireless networks is secure."

"Our particular spin on location-based methodologies is that our VPN has a policy module associated with it allowing us to change both the VPN's behavior and the things to which the user is allowed access, based on their location," says Knopf. "For us, a 'location' means various things: the physical IP address assigned to the user's network card, or a particular network interface used by the employee - they could be on a wireless LAN versus a carrier cellular data network. The user could even be associated with a particular SSID [Service Set Identifier, a 32-character unique identifier attached to packet headers sent over a WLAN that acts as a password when a mobile device connects to a base station]. We have a working knowledge of all those things and we can dynamically change what the user has access to and change the behavior of our VPN most of the time to make people more productive and to ensure that they're getting the best service from their data carrier and for their specific applications."

"So, we're not a traditional location-based application, where a user says, 'I'm in a certain location. What are the associated services around me such as hotels and restaurants?' Instead, we focus quite heavily on making the worker productive based on their location," concludes Knopf.

Remote Access

One of the earliest technologies to make workers mobile was remote access. It's modern descendant is still going strong today.

At Aventail (news - alert) (http://www.aventail.com), Director of Product Marketing Chris Witeck says, "We're fundamentally a remote access control company. We examine the different ways a user potentially can gain access to information and applications on the network, as well as really increase the functionality of mobile devices as a market driver. But then, the definition of mobile driver devices includes many things. Just over the last two years we've expended effort into developing, in particular, for traditional PDA and phones and to allow users to control access to those devices. But we also take into account the fact that another trend has been going on for a longer time - the fact that laptops are much more cost-effective these days, and you're increasingly capable of putting other things on your laptop that traditionally were in the realm of the mobile device, such as a VoIP softphone."

"The convergence of voice and data requires IT to look at remote access differently," says Witeck. "There are more and more use-cases where users can work remotely; there are more devices and connectivity is available at higher bandwidths and at more locations. People are also outsourcing their supply chains and are working more with people who are in a different location. Aside from these factors, companies used to have a 'voice team' that would buy phones for employees, and then there was a 'data team' handling data connectivity and network access reliability. Put voice on the same device, and you start to see the IT guys get more involved in the voice decisions and the voice guys get more involved in the data decisions. That changes things to a certain extent within an organization."

"From our perspective, this all raises the strategic profile of remote access quite a bit," says Witeck. "Much of this is being driven by the fact that increasingly functional devices that can handle both voice and data are appearing. We see a big surge in interest in phones or PDAs becoming remote access platforms, and that's driven by Windows. Palm was an early vendors to deploy a functional PDA device, and Blackberry was an email-only device. But given that IT guys are interested in both data and voice, interest is driven with devices that can be increasingly functional for both access to voice-based services and to specific applications."

Let's hope the world's employees don't become peeved by these gizmos that enable them to be reached by anyone, anytime, anywhere. IT

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

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CommuniGate Update

By Greg Galitzine

CommuniGate Systems has taken another step towards the next generation of Internet Communications with its recent announcement of Pronto!, a Rich Media user client based on Adobe Flex 2 technology.

According to Jon Doyle, vice president of business development at CommuniGate, (news - alert) "Rich Media is synonymous with "interactive multimedia," and Pronto brings this new medium together with email, IM, and scheduling."

In fact, to hear Doyle tell it, much of the credit for the widespread adoption of today's business productivity tools needs to be given to the consumer sector.

"The first two pillars of Internet communications over the last 10 years have been email and websites, and one could argue consumers drove email into widespread adoption as the primary medium for business communications with all the CompuServe and early Hotmail types of services."

But email is so "web 1.0" isn't it?

Building on this theme, Doyle believes that IM and chat are also gaining momentum in the business world thanks to widespread adoption of these technologies in the consumer market.

"Pronto is all about unification of these standard forms of communications in a client that is secure and far more portable than a fixed desktop client, said Doyle."

We've written about the CommuniGate Pro IP communications platform in the past and how the platform integrates applications such as scheduling with the SIP engine, and presence with IP phones (i.e., from Polycom) and IM clients. Pronto is the Rich Media client that presents all these applications in a single, easy to use and flexible user interface.

According to the recent announcement from CommuniGate, Pronto talks to CommuniGate Pro via the XIMSS API (XML Interface for Messaging, Scheduling, and Signaling).

This new API is was introduced to CommuniGate Pro in version 5.1 last summer, and it was designed to enable rapid development of lightweight clients and interfaces with web and XML skill sets. The idea is that XIMSS will allow developers to design UIs, build portals, interface with cable modems, or link to external applications and services with no need to use complex protocols.

CommuniGate also recently announced results of a massive scalability profile, where the CommuniGate Pro SIP Farm running on the IBM System z mainframe was tested at IBM's lab in Montpelier, France.

According to a company statement, the System z mainframe demonstrated excellent overall scalability as the number of Linux virtual machines tested was increased to a maximum of 20 running the CommuniGate Pro SIP Farm Dynamic Cluster under z/VM, the mainframe virtualization technology. As the number of virtual machines was increased, the throughput increased near-linearly. z/VM enabled very efficient utilization of mainframe resources while meeting throughput objectives with a growing workload.

I spoke to Bill Reeder, Linux System z9 Architecture and Strategy, at IBM, about the relationship between IBM and CommuniGate.

CommuniGate is an IBM ISV business partner. The companies have been working together for about five years with IBM hosting applications, such as CommuniGate's Stalker, on System z hardware.

When asked about the significance of the scalability profile, Reeder responded, "These tests are very significant in today's marketplace, as telcos are undergoing a rapidly increasing need to host VoIP (define - news - alert) to their customers. Scaling to 25 million users on one server platform raises the bar and creates a good solution for those customers."

Regarding IMS, Reeder believes that the ability to scale to 25 million users while maintaining a small server footprint in an energy efficient manner means the solution should be very appealing to carriers.

"Keeping in mind that the scaling of services during the test was linear with the number of processors, those customers can scale as their demand for VoIP services increases. When one considers the scaling possibilities hosting IMS services on System z, along with management costs, and a lower cost per square foot for energy consumption, the TCO story can be very compelling."

The future portends some exciting news for CommuniGate. Doyle told me that the proposed summer 2007 release of version 5.2 will be all about expansion of mobility.

"We are working tightly with Adobe on Flash Lite for handsets such as the Blackjack," said Doyle.

"Our strategy is to expose all the Rich Media capabilities of the platform to mobile handsets. Subscribers will be able to share music, update blogs, or hand off conversations on their GSM network to the IP (WiFi) network seamlessly." IT

Greg Galitzine is Group Editorial Director for TMC's IP Communications Media Group.

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