VoIP Monthly Newsletter

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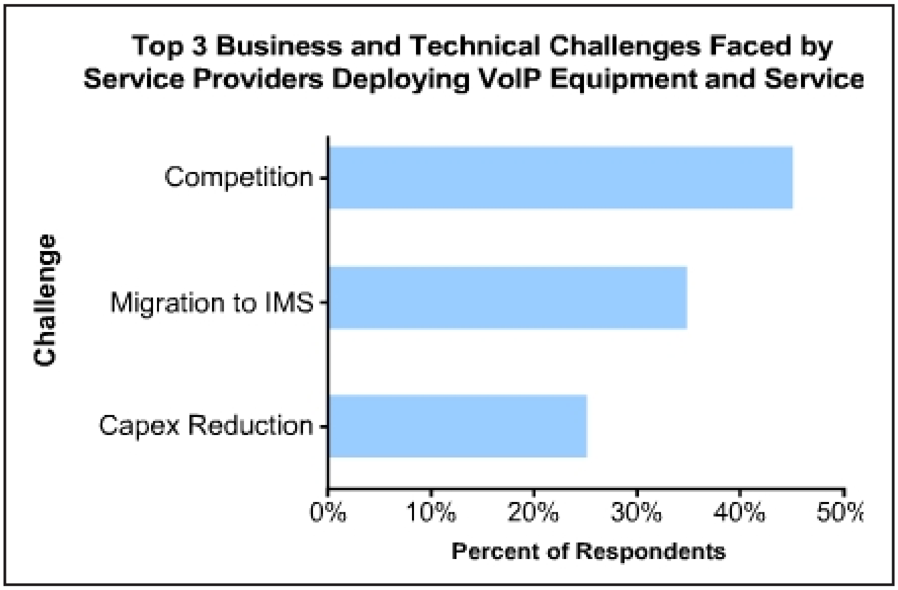

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SERVICES

Deutsche Telekom (ICSS) and VocalTec announce solution for international VoIP interconnection

VocalTec Communications Ltd., a global provider of carrier-class multimedia and voice-over-IP solutions for communication service providers, and International Carrier Sales & Solutions (ICSS), the wholesale arm of Deutsche Telekom, have launched an innovative solution for global VoIP Interconnection.

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After successfully completing test and evaluation procedures in a live network environment, the Essentra EX solution was accepted by Deutsche Telekom ICSS and is commercially available to enable IP interconnection of Deutsche Telekom's international VoIP network to other VoIP networks/affiliates.

With this innovation, Deutsche Telekom ICSS can market reliable international VoIP interconnection services to its customers. Wholesale providers and Internet Service Providers (ISPs) can now use the Deutsche Telekom international network to carry their VoIP traffic and benefit from assured QoS, security, and optimal routing plan management. This solution is part of the global network for voice minutes termination.

"We required a field-proven VoIP Interconnection solution that will enable us to interconnect to other VoIP networks in a cost-effective manner and with the VoIP oriented flexibility of our VoIP services," said Helmut Angst, vice president, ICSS, Deutsche Telekom. "VocalTec's solution satisfies all of our technical requirements, particularly in the reliability and interoperability required for interconnection with other VoIP networks and enables us to also quickly address innovative customer requests and projects."

"We are proud to expand our long-term cooperation with Deutsche Telekom ICSS, and to become its provider of VoIP Interconnection solutions," said Yosi Albagli, president and CEO at VocalTec. "Having our Essentra EX solution installed in one of the world's top telecommunications companies' network, clearly positions our product as a leading solution in the market. We look forward to collaborating with Deutsche Telekom ICSS and to support its future growth."

Installed with numerous Tier 1 service providers worldwide, Essentra EX offers a cost-effective solution for the interconnection of disparate VoIP networks. With full support for SIP-to-H.323 interworking, Essentra EX is a highly interoperable solution allowing service providers to quickly and efficiently meet their business objectives of customer acquisition, reducing OpEx, and rapid return-on-investment, while ensuring quality-of-service, reliability, and security.

Talkster powers free international and long-distance calls for GTalk2VoIP

Talkster (www.Talkster.com), the company delivering to online communities instant mobile voice and text communications, officially launched its integration with GTalk2VoIP, offering callers worldwide free international, long-distance, and group conference calls. Talkster's partnership with GTalk2VoIP extends the company's ad-supported free calling service from 34 countries to cover the entire globe.

The partnership between Talkster and GTalk2VoIP allows callers from any country in the world to use Google Talk to make free calls to or from anywhere else in the world. Now, if the country you are calling to or from is not included in Talkster's network of more than 30 countries worldwide, you can simply use Google Talk to call the Talkster number provided to make your portion of the call completely free.

Especially beneficial to areas with strictly regulated VoIP and telecom practices, such as India and United Arab Emirates, Talkster through GTalk2VoIP will let people communicate freely between mobile phones, landlines, and PCs, even in countries not currently covered by Talkster's ad-supported free calling service.

"In Africa, the Middle East and other places with restrictive VoIP laws, Talkster for GTalk2VoIP offers free calling without any red tape," said Talkster COO and co-founder James Wanless. "Our mission is to bring ad-supported free telephony to the world, yet Talkster is constantly receiving requests for countries where access numbers aren't yet available. It was important for Talkster to find a way to serve the global community, and our integration with GTalk2VoIP gives anyone in any country a way to talk to family and friends all around the world without incurring long distance and international calling charges."

GTalk2VoIP is a free and publicly open voice gateway for major instant messenger clients. It makes it possible for people to use instant messenger services, including Google Talk, to call people on regular phones, without requiring any additional software, establishing new accounts, or paying any further fees.

"The idea behind GTalk2VoIP is not only to provide outgoing calls from GoogleTalk to ordinary telephone numbers, but make them affordable while still providing the highest possible call quality," said Ruslan Zalata, founder of GTalk2VoIP. "Our partnership with Talkster brings even more value to what we are offering our users, because now they can use GTalk2VoIP to make completely free unlimited international calls to any phone, landline or mobile, with no registration or credit card required. This is like the holy grail for many of our users who are spending a fortune on long distance and international calls."

When away from their computers, callers can use Talkster's Free World Dialing with GTalk2VoIP from their mobile phones by downloading Talkonaut, a free mobile client. Callers can download the Talkonaut software at www.Talkonaut.com. The Talkonaut mobile client works over GPRS, 3G, or Wi-Fi, allowing callers to use their data plan if they choose. With Talkonaut, callers can use Talkster's Free World Dialing with GTalk2VoIP to make free international, long-distance, and group calls directly from their mobile phones for free.

All Talkster calls are ad-supported, so users don't pay long-distance or international calling charges, and can talk as long as they like for the cost of a single local call. Callers listen to a 10-second audio ad prior to the start of their call and their talk time is never limited or interrupted. The ads are localized with relevant offers and coupons, bringing the caller something valuable in addition to reducing their phone costs and providing excellent call quality.

"This partnership is just one example of how the Talkster ad-supported voice platform can be leveraged by third parties to offer free calling to customers," added Wanless. "Talkster's platform and open API makes it simple for any online site, from social networks to online classifieds to easily offer users free ad-supported calling as a value added service with no risk or infrastructure investment. Talkster also makes it simple for these companies to monetize their offerings through advertising revenue sharing, making the Talkster platform an easy way for brands to build both value and revenue."

NEW PRODUCTS

Cbeyond and Microsoft offer small businesses a complete IP-based phone solution

Cbeyond Inc., an IP-based managed services provider to small businesses, announced the availability of its BeyondVoice with SIPconnect service preconfigured on the Microsoft Response Point IP-based phone system. BeyondVoice with SIPconnect helps to simplify the installation of services for small businesses using Response Point Service Pack 1, which is now generally available.

"We selected Cbeyond as a preferred provider because of its high-quality service," said Xuedong Huang, general manager of Microsoft Response Point. "Combining Cbeyond's service with the Response Point phone system extends the benefits of our easy-to-use phone system with reliable VoIP service to small businesses in key metropolitan areas across the U.S."

Cbeyond's BeyondVoice with SIPconnect service is an integrated package of local and long-distance voice, high-speed Internet, and mobile services that uses the SIPconnect specification to interoperate with IP-

PBX phone systems like Microsoft Response Point.

Cbeyond helped to pioneer the Session Initiation Protocol (SIP) trunking industry standard, called SIPconnect, which institutes a common protocol among digital voice providers worldwide. SIPconnect eliminates the need for a VoIP gateway on the premises, improves voice quality, and creates a strong foundation for personalized applications and rich media services. Microsoft Response Point customers using Cbeyond's service will be able to take advantage of digital voice service based on SIPconnect.

"Our strategic alliance with Microsoft allows us to provide a complete solution to our joint customers that is both reliable and cost-effective," said Rob Consolazio, vice president of marketing and business development for Cbeyond. "By leveraging the tried and tested SIPconnect specification for full featured telephony, small businesses can take advantage of extensive calling features provided by service providers such as Cbeyond who have a reputation for delivering premium customer service."

Cbeyond channel partners selling the combined solution are able to take advantage of Cbeyond's generous compensation plans and have access to extensive training and resources. Through Cbeyond PartnerAccess (cbeyondpartners.net), the company's channel partners can manage their customer accounts, download sales collateral, create on-demand marketing materials, view account reports, and peruse product information at their convenience.

"With the combined Microsoft and Cbeyond solution we can quickly configure and install an IP-based phone system to our small business customers," said Olivier Havette, Cbeyond channel partner, Microsoft Gold Certified partner, and CEO of The Net Doctor. "The ease-of-use and service reliability offers our customers a complete solution that is cost-effective and reliable."

Earlier this year, Cbeyond fortified its relationship with Microsoft by integrating the Hosted Microsoft Exchange solution into its service offering, which is the first hosted service based on Microsoft Exchange Server 2007 that is fully integrated with mobile access from a single provider. For more information about Cbeyond and its small-business communications services, visit www.cbeyond.net.

Vanu Inc. and Mavenir Systems introduce dual-mode wireless network solution with VoIP core

Vanu Inc., a developer of software radio solutions for cellular operators, and Mavenir Systems, a provider of service delivery network technology, introduced a new GSM and CDMA network solution using existing or planned Class 5 VoIP switch investments. Current service providers and new entrants can now use the same VoIP switch, billing system, and service delivery platform to seamlessly support convergence services over both wireline and wireless devices. The all-IP architecture of the solution, coupled with the flexibility of a software-based Radio Access Network (RAN), enables operators to gracefully transition to future wireless standards, such as UMTS, LTE, and beyond.

With the integration of a Mavenir mOne Convergence Gateway, a wireline VoIP switch can now perform the functions normally provided by a standalone Mobile Switching Center (MSC), including call control, mobility, messaging, voicemail, and supplementary services standard to GSM and CDMA handsets.

"This 3GPP standards-based approach allows VoIP carriers to provide wireless service without a large upfront investment in a new wireless-only switch. Now carriers can capitalize on their current network investments to accelerate their entry into the wireless market," said Payam Maveddat, Mavenir's VP of product management. "New entrants with greenfield deployments also benefit from a VoIP switching infrastructure, which is significantly lower cost than traditional circuit switching and supports a wider range of services and applications."

With the switching infrastructure in place, the wireless network is completed through the use of the Vanu Anywave software radio to implement the entire RAN as portable application-level software running on standard processors and operating systems. Vanu Anywave is the only RAN product to simultaneously support multiple standards (GSM and CDMA) on the same platform. This unique capability allows carriers to capture new roaming revenue or support legacy standards while adding new ones. New wireless standards are readily available as software downloads rather than costly hardware upgrades. Carriers also benefit from backhaul cost savings because an all-IP backhaul enables wider choice of transport, including DSL, cable, and microwave.

"Vanu is committed to addressing the needs of small service providers across North America and the Caribbean," stated Bryan Martin, director of sales for Vanu Inc. "Our innovative wireless infrastructure solutions have helped our customers expand their service offerings, open new revenue streams, and grow their customer base. We are very excited about our partnership with Mavenir and the opportunity to provide an integrated, comprehensive solution for current wireline and wireless operators. Our collaborative solution will provide companies with unused spectrum assets a low-cost entry point into the wireless market."

Snom klarVoice delivers the next generation of VoIP sound

Snom technology AG, a developer and manufacturer of advanced voice-over-IP phones for the commercial and residential markets, announced the next generation of voice: snom klarVOICE. This wideband handset, which can be adapted to all snom VoIP telephones, captures more than twice the spectrum of voice frequencies as standard phones, enabling phone conversations with greater clarity and richness.

"Telephones have remained unchanged for so long, most people have no idea what limitations they have lived with," explained Dr. Michael Knieling, executive VP of marketing and sales. "But VoIP also lays the groundwork for a revolution in the quality of voice we communicate with!"

The new snom klarVOICE handset works with the codec G.722. This codec is able to shrink the bit rate of the voice channel down to 12.65Kbps, offering excellent quality. In standard narrowband VoIP calls, the voice signal is sampled at 8,000 times per second, resulting in an effective voice pass-band of about 200 to 3,300Hz. The new wideband handset offers a doubled sample rate, providing an effective pass-band of 50 to 7,000Hz.

Use of snom klarVOICE results in a much higher-fidelity voice call, more akin to talking to someone in the same room rather than over a phone.

The snom klarVOICE handset, which can be adapted to any existing snom 3xx series VoIP phone (snom 300, 320, 360, 370) using snom's latest firmware release (Version7.1.33), is available for an MSRP of US$32.50. Snom 3xx series are business-class, SIP VoIP phones and feature a global executive design and styling with a large, high-resolution greyscale display screen, programmable function keys, and advanced business calling features.

PARTNERSHIPS

BroadSoft and Fonality team up to offer managed IP PBX for service providers

Fonality, a provider of open source phone systems, and BroadSoft Inc., a provider of VoIP application software, announced that the two companies have certified their products to work together, enabling service providers to market and deploy Fonality to their small- and midsized-business (SMB) customers with confidence. BroadSoft, which provides VoIP applications and SIP trunking to seven of the top 10 and 13 of the 25 largest carriers worldwide, has completed certification of Fonality trixbox Pro and PBXtra with BroadWorks, BroadSoft's VoIP application platform for fixed-line and wireless service providers. BroadSoft has also joined the Fonality Authorized Certified Ecosystem (FACE).

BroadWorks offers a range of carrier-grade applications that includes hosted PBX, unified communications, mobile PBX, business trunking, and residential broadband. Fonality products include a family of open source-based, hybrid-hosted IP PBX offerings tailored for SMBs. By certifying the products together, the companies offer service providers a complete line of hosted and premises-based unified communications offerings for customers of all sizes.

"By collaborating with Fonality, we're giving service providers a way to go to market quickly with fully integrated, market-tested offerings for smaller companies," said Leslie Ferry, vice president of marketing for BroadSoft. "Service providers can now take new products to their customers with the confidence that comes from knowing that both companies' solutions will work together seamlessly."

"BroadSoft is the dominant VoIP platform deployed by service providers and MSOs," said Chris Vuillaume, vice president of business development and channels at Fonality. "Their certification is an important stamp of approval for Fonality products and expands our market opportunity to include Tier 1 and Tier 2 providers."

Fonality business phone systems are designed for modern workplaces, accommodating companies that have a mix of office, mobile, and home-based workers. Fonality solutions support both VoIP calling and traditional phone lines, allowing a smooth transition for businesses upgrading their calling services. Its patented, hybrid-hosted architecture allows employees' identities to be maintained as they travel between work, home, and hotels.

Fonality products, when paired with the HUD presence software, provide a unified view of instant messaging, email, and calling for all fixed and mobile workers.

Costco Wholesale Business Centers to sell Sotto Wireless office telephone service

Sotto Wireless (www.sotto.com) announced that its all-in-one wireless and office phone communications service, Sotto@SOHO, will be sold through Costco Wholesale Business Center stores at the Fife, Washington, Costco store. Other participating stores will include Costco Business Centers in Lynnwood, Washington; Hayward, California; and Phoenix, Arizona.

Sotto services combine the savings of hosted VoIP with the freedom of cellular telephony, allowing small businesses to communicate like big companies at a fraction of the cost. Subscribers get the mobility of cellular along with the services traditionally associated with an office PBX — extension dialing, call transfers, and an automated attendant — all with one phone number that follows them anywhere they do business.

Sotto@SOHO is for businesses of up to 10 employees working anywhere — at a central office, remote and home offices, or on the road. All Sotto phones place and receive calls over VoIP for a low fixed monthly rate when in range of a self-installed Sotto Wi-Fi Gateway. Away from the office, Sotto's Business Anywhere Nokia smart phones switch to the GSM cellular network.

"Costco Wholesale Business Centers offer an array of services important to business owners," said Richard Chavez, senior vice president, Costco Wholesale. "Sotto@SOHO is targeted to businesses that are focused on value for what they spend. We're pleased to add Sotto@SOHO to our lineup."

With the growth of mobile-phone services as the norm for businesses of all sizes, and often the only phone service for startups and small businesses, Sotto sought to give people the ability to pick up calls from the office as well as the road. Users have one number, one phone, and one voicemailbox that work the same, no matter where they are. In addition, users do not need to worry about racking up minutes charges on their cellphones since they have unlimited minutes through the Sotto Gateway. Business owners retain control of their most valuable asset, customer communications, through Sotto's innovative Service Manager, which gives administrators access to detailed usage and calling records on business lines.

"Businesses can't afford to miss calls, but many can't afford to pay for the increasingly high costs of multiple telephony options — as anyone with both land lines and cell phones knows," said Rod Nelson, CEO of Sotto Wireless. "Sotto@SOHO gives users the benefits of both services — the mobility of cell phones and the permanence of office systems — for one easy-on-the-budget price. Plus it keeps ownership of customer communications in the hands of the business owner."

Costco will sell Sotto@SOHO starter kits containing two Nokia E51 smartphones and a Sotto Gateway for $399. A $699 kit includes two Nokia E61i smart phones, a Sotto Gateway, and a Polycom IP320 desk phone with Office Attendant. Service plans start at $49.95 per month for unlimited nationwide VoIP calling, plus cellular bundles as low as $49.95 for 1,000 voice minutes. Sotto cellular bundles are shared among all of the business's users, never expire, and only need to be purchased again after they're used up. As a special introductory offer, Sotto is providing 90 days of free wireless VoIP service, 1,000 cellular voice minutes, 20MB of data, 100 text messages, and a 60-day money-back guarantee.

Sotto@SOHO is also available through Sotto's online store at www.sotto.com, in addition to the Costco Warehouse Business Centers in Fife, Washington, and soon in Lynnwood, Hayward, and Phoenix.

Dialogic and Broadvox team up to deliver SIP trunking service

Dialogic Corporation, a global provider of world-class products and technologies for media and signal processing, and Broadvox LLC, a provider of integrated managed VoIP services to SMB, enterprise, and carrier customers, announced the interoperability of the Dialogic 2000 Media Gateway Series and Broadvox GO! SIP Trunking. The combined service and hardware solution enables SMB and enterprise customers the ability to deploy the flexible and cost-effective SIP trunking service with legacy PBX and key telephone systems.

Broadvox GO! SIP Trunking products provide significant cost savings of up to 70 percent when compared to traditional telecom services. Customers receive unlimited local calling with discounted toll-free, nationwide, and international calling. Service options include E911, DIDs, and local number portability, allowing businesses to keep their current phone numbers.

The Dialogic 2000 Media Gateway Series (DMG2000 Series) provides a SIP trunk interface to the Broadvox service with T1/E1 trunk interfaces to the PBX or hybrid key telephone system. The DMG2000 Series is noted for its ease of use through Web-based configuration interface and reliable 1U rack mount appliance design. A range of TDM interface densities from single T1/E1 to quad T1/E1 and "stackability" provides a competitive SIP trunking interface for businesses with 50 or more users.

"The arrival of SIP trunk service, delivered by providers such as Broadvox, is similar to the value proposition and market disruption seen with deregulation of the US long distance services market in the 1980's" noted Jim Machi, senior VP of marketing at Dialogic.

"This time it's even better for business since the cost advantage covers both local and long distance service. To date, most of the market has focused on introducing SIP trunks with the purchase of an IP-PBX. Our DMG2000 Series of appliance gateways are a cost-effective way to open up the much larger market — the installed base of SMB and enterprise customers that continue to get good service from their existing telephone system, but want the benefits of SIP trunking now."

Dialogic recently introduced two new models to the DMG2000 Series with failover relays that further enhance their use with SIP trunk services.

The failover feature enables the gateway to switch back to conventional PSTN service in the event of service disruption over the broadband IP connection. This can provide extra assurance to customers eager to take advantage of SIP trunks, while keeping a traditional PSTN route available as a backup or alternate calling route.

"The Dialogic 2000 Media Gateway Series is a terrific complement to the Broadvox GO! SIP Trunking service. Interoperability with Dialogic means that Broadvox VARs can offer any medium to large SMB or enterprise customer an easy to deploy communication solution by replacing or supplementing a legacy PSTN service with SIP trunks without making a major IP-PBX investment. The ability to insert the gateway between the existing phone system and the SIP trunking service delivers maximum benefit with minimal disruption to the legacy infrastructure. Plus, it provides savings on monthly telecom costs while providing the foundation to move toward an all VoIP SIP-based system in the future." said David Byrd, VP marketing and sales at Broadvox.

To make it easy for interested VARs to try out the DMG2000 series, Dialogic has announced a Starter Kit Program for the Dialogic Media Gateway Series, which provides training, support, and a generous discount on initial gateway purchases. More information about this program is available at www.dialogic.com.

Comcast and Vonage form collaboration to address network management and better meet customer needs

Comcast Corporation and Vonage Holdings Corporation announced a collaborative agreement to address the reasonable network management of Internet services. Comcast committed to work together with Vonage to ensure that network management techniques are chosen that effectively balance the need to avoid network congestion with the need to ensure that over-the-top VoIP services like Vonage work well for consumers.

"This agreement helps Vonage to ensure that customers have the best possible Internet experience," said Louis Mamakos, Vonage chief technology officer. "Although we're competitors with Comcast, this understanding helps our two companies work together to balance the needs of network management with consumers' ability to freely access the services, applications and content of their choice."

"This collaboration with Vonage, and our outreach to many key participants in the Internet community, demonstrate that we are committed to provide network management solutions that benefit consumers and competition," said Tony Werner, Comcast chief technology officer.

This is the latest in a series of announcements related to Comcast's network management practices that demonstrate the company's commitment to ensure its customers' ability to use any application or access any content they choose while avoiding network congestion situations that could affect the consumer experience. In March, Comcast announced it would move to a protocol-agnostic network management approach by the end of 2008, and tests on this approach have already begun. Comcast has announced other collaborations with BitTorrent Inc. and Pando Networks, as well as participation in the P4P

Working Group organized by the Distributed Computing Industry Association (DCIA). Comcast has also participated in the IETF Workshop on P2P Infrastructure, and will continue to collaborate in the IETF with other ISPs, P2P providers, and others on technologies related to network management and P2P application development.

Junction Networks launches channel program for OnSIP PBX service

Junction Networks, host provider of the OnSIP PBX service, has launched an OnSIP Authorized Agent program aimed at developing a nationwide channel of qualified resellers. Starting today, Junction Networks is opening the door for system integrators in the fields of data, networking, and telecom to qualify for a new revenue stream by promoting its business communications service.

"At Junction Networks, we are committed to delivering VoIP to businesses in a way that makes sense technically, breaks the barriers of traditional phone service and levels the playing field economically," said Junction Networks president Rob Wolpov. "That's why OnSIP is designed in-house, using the best available open technology components; is priced with no expensive user, extension or 'seat' licenses; imposes no commitments, minimums or contracts; is simple to set up and manage; and is loaded with the features you would expect from a $25,000 phone server."

Recognized as one of the "Top 15 VoIP Players To Watch" by Channel Reseller News, Junction Networks has been selling business-class voice service, PBX switching, and enhanced services directly to small to medium-sized businesses since 2004, amassing more than 4,000 customers.

In exchange for joining in its sales effort, Junction Networks is offering its authorized agents a share of ongoing revenues from two sources: highly profitable monthly services such as auto-attendants, voicemailboxes, conference bridges, and hunt groups, and per-minute calling activity to and from the PSTN. The agent program also includes training, technical support, an extensive online knowledgebase, marketing materials, RFP assistance, and access to leading phone vendors.

For those selling technology services to SMBs, the OnSIP Agent program represents an opportunity to add remote offices and new hardware to a support business. Junction Networks' service is particularly friendly to phone sales, since it does not charge for OnSIP per extension, but by usage.

Revenue percentages will scale with agent level. In addition, OnSIP agents stand to profit by establishing service contracts and retainers with clients to procure, install, configure, and manage phones on an ongoing basis. Junction Networks encourages agents to do so, as such contracts are an additional revenue opportunity that fits well within existing revenue models for resellers.

Wolpov sees his potential OnSIP agents in three major types of existing consultancies: data network specialists, IT consultants, and telecom consultants. Since the service is based on IP, those who install, support, and manage routers, firewalls, ISPs, and LANs are well positioned to offer their customers a money-saving telephony proposition.

"To IT consultants, OnSIP is simply another IP service to add to your portfolio," said Wolpov. "And telecom consultants know better than anyone how traditional PBX and key systems are being replaced en masse by IP-based solutions. For them, OnSIP is the continuously profitable, low-maintenance answer to their existing customers' VoIP migration question."

Those interested in becoming authorized agents for OnSIP must demonstrate technical competence and an ability to sell, implement, and manage Internet-based services. Junction Networks' agent specialists will also assess applicants' qualifications based on go-to-market plan, target customer base, and current business model. Applicants are encouraged to assess OnSIP, in turn: All OnSIP features are free to try for 30 days, and prospective agents can receive $10 of free calling services to make and receive calls to/from the PSTN.

For more information on the Junction Networks Authorized Agent program, please e-mail agent@junctionnetworks.com.

Teledex announces new IP cordless guestroom phones

Teledex, a worldwide provider of guestroom telephones for the hospitality industry, announced the release of a new line of VoIP DECT cordless phones. The Teledex iPhone SIP NDC series phones are SIP-compliant cordless phones designed for the specific deployment requirements of the global hospitality industry.

Using the latest DECT technology for base-handset communication, and SIP protocol for IP-PBX integration, the new Teledex SIP NDC series phones offer flexibility and features heretofore unavailable for the worldwide hospitality industry. Based on the Teledex iPhone SIP ND series corded IP phones — which enjoy a wide range of out-of-the-box interoperability — the new IP cordless phones come ready for integration with many popular IP-PBXs, to enable universal deployment across a hotel corporation's entire operating portfolio. Sharing the same attractive case design as the other models in the Teledex iPhone line, the SIP NDC series enables a consistent look and feel for every hotel chain, regardless of telephony infrastructure.

Teledex CEO Ron Lesniak, Ph.D, commented, "In listening to our extensive customer base, it became clear that there was a huge void in the market for a cordless IP phone. Virtually all the major new-build hotels and resorts — and increasingly, existing hotels — are going IP, and they need a cordless IP phone for these installations. There are significant challenges in producing an IP cordless phone in a design that's small enough for the tight space considerations that are so important to this industry, and I'm proud of the team that has worked on this project."

Available in single- and two-line configurations, with up to 10 programmable guest service keys, the Teledex iPhone SIP NDC series is due to ship in the fourth quarter of 2008. Pricing for the new series of phones has not yet been established. Interested parties should contact sales@teledex for pricing information.

BroadLight and AudioCodes to deliver advanced VoIP services over GPONs

BroadLight, a supplier of GPON semiconductors and software, announced that it has joined forces with AudioCodes, a provider of VoIP technologies and Voice Network products, to deliver high-performance VoIP solutions over GPONs. By designing the AudioCodes' VoIPerfect engine to run seamlessly on BroadLight's BL2348 GPON Residential Gateway System-on-Chip, BroadLight will enable service providers to offer advanced VoIP services over high-bandwidth GPONs.

"By complementing our industry-leading GPON processors with AudioCodes' widely deployed VoIP technology, equipment companies can easily migrate from external DSP to the BL2348 integrated DSP and offer feature-rich toll-quality voice services while maintaining their existing voice application software," said Doron Tal, vice president of business development and product management at BroadLight. "GPON's speed and QoS performance enables high-speed triple-play services with uncompromising VoIP quality."

"BroadLight and AudioCodes both provide field-proven technology which has a high rate of deployment in the marketplace today," stated Shaul Weissman, vice president of VoIP processors at AudioCodes. "The combination of our respective technologies and market leadership will provide customers with powerful and feature-rich VoIP technology over GPON."

The GPON technology enables the usage of high-definition voice in the residential gateway, providing customers with unprecedented voice quality. The addition of AudioCodes' VoIPerfectHD software to the BL2348 will enable service providers to excel the PSTN voice quality and offer additional value-add services on the same platform.

The BL2348 GPON RG SoC provides the performance, integration, and functions required for the new breed of cost-effective residential gateways connecting the high-speed digital home to FTTH services. The BL2348 uses its field-proven PONRunner network processor to perform bridging and routing functions with throughputs of 1Gbps. Its embedded GPON MAC, integrated VoIP DSP, Ethernet switch, and interfaces for Wi-Fi and USB creates a cost-effective single chip solution in a 19x19mm PBGA-441 package, working at industrial temperature range.

VoIPerfect is AudioCodes' underlying, best-of-breed, core media gateway architecture for voice-over-IP technology products and systems. AudioCodes' VoIPerfect platform provides the main technology building blocks of AudioCodes' entire product line. The VoIPerfect architecture comprises feature-rich VoP (voice-over-packet) DSP software and highly optimized media streaming embedded software, integrated PSTN signaling protocols and VoIP standard control protocols, provisioning and management engines, and additional features enabling carrier-grade quality and high-availability. The VoIPerfect architecture components are available in AudioCodes' products at various levels of integration — from chip-level products through PMC modules and ATCA/PCI/cPCI blades, to high-availability and nonhigh-availability analog and digital Media Gateway platforms.

FINANCING

Prominent Kuwaiti businessman to fund $9 million joint venture for international launch of i2Telecom's MyGlobalTalk

I2Telecom International Inc., a developer of patented and innovative high-quality voice-over-Internet Protocol products and services, announced that it has executed a memorandum of understanding (MoU) for a strategic joint venture with Raed A.H. Rajab (the "investor"), a prominent businessman in Kuwait, to expand the company's operations into a number of international markets by the end of this year. Through established sales channels, the investor will provide $9 million in funding for operating and marketing expenses (the "capital commitment") incurred by the joint venture during the next 12 months involving the launch of i2Telecom's MyGlobalTalk and other telecom offerings into these markets.

The joint venture will establish wholly owned subsidiaries in each country and transfer exclusive rights to the company's technology to that country in exchange for the capital commitment. The company will consolidate the operating results of the joint venture in its financial results beginning with the third quarter. Revenue from the joint venture is not included in i2Telecom's previously announced guidance.

Upon execution of a definitive agreement subject to customary closing conditions and receipt of a $1 million initial capital contribution, the company will issue to Mr. Rajab 1 million restricted shares of i2Telecom common stock as an equity incentive.

The company may, at its discretion, issue additional shares of i2Telecom restricted stock in lieu of distribution of 20 percent of the joint venture's profit.

Such shares will be issued at 95 percent of market value of the company's shares on a quarterly basis as profits are earned. I2Telecom will retain 100 percent equity ownership and an 80 percent profit participation in each wholly owned subsidiary that is established by the joint venture.

MyGlobalTalk, which recently won Internet Telephony Magazine's and Unified Communications Magazine's "Product of the Year 2007" awards, places Internet telephony in the hands of every cell-phone user, independent of wireless carrier technology, handset manufacturer, or type of wireless carrier voice/data plan involved.

"We are very pleased to have been selected by Raed A.H. Rajab as his telecommunications partner in this rapidly developing region of the world," stated Paul Arena, chief executive officer of i2Telecom International Inc. "We believe that our ongoing success in the international marketplace reflects not only the fundamental economic benefits of mobile VoIP, but also the added value of our intellectual property. Working with such successful businessmen as Mr. Rajab in these growing markets, we will provide significant cost savings and improved communications capabilities to the customers and employees of organizations around the world."

The joint venture benefits i2Telecom shareholders by providing $9 million to finance operating and marketing expenses that will enable the company to expand its international infrastructure and establish scale in these growing foreign markets.

I2Telecom's proprietary platform can easily be integrated within a combination of hardware, software, and cellular phones for next-generation global telecommunications solutions.

The platform consists of a new architecture for the company's VoiceStick, MyGlobalTalk, Digital Portal communications and microgateway products, innovative applications built to Session Initiation Protocol (SIP) standards, global VoIP network, global Public Switched Telephone Network (PSTN) interconnect network, and geographically dispersed network operation centers (NOCs).

"We selected i2Telecom because of its advanced features and technology, the Company's ability to provide unparalleled service in remote areas of the countries we are targeting, and its excellent reputation for supporting global partners," stated Raed A.H. Rajab. "We see this as an opportunity to make an investment in mobile VoIP technology that can provide an above average return on our capital. i2Telecom's products are very easy to deploy and an ideal complement to SMEs, Internet Cafés and Hotels in numerous countries in the Middle East, Asia and elsewhere. We believe i2Telecom is positioned to emerge as one of the leaders in this region of the world, and we intend to put our full effort behind the joint venture with i2Telecom to make this happen."

Businesses and consumers can substantially reduce long-distance and international telecommunications costs by using i2Telecom's easy-to-deploy VoIP solutions that seamlessly interface with existing mobile-phone connections without having to change wireless carriers or voice/data plans currently in place.

CONTRACTS

Acme Packet safeguards True Corporation's VoIP network

Acme Packet, a provider of session border control solutions, announced that True Corporation, Thailand's only fully integrated communications service provider and the largest broadband provider in the country, has deployed Acme Packet Net-Net session border controllers (SBCs) for wholesale peering interconnects, IP PBX trunking, and residential VoIP. The Net-Net SBCs are deployed in Bangkok using SIP and H.323 signaling protocols. Acme Packet's multiservice architecture maximizes service reach and security at the access and peering interconnect borders of True's VoIP network.

Acme Packet's SBC provides an extensive feature set to protect and optimize

True's VoIP network. Denial-of-service (DoS) attack prevention, topology hiding, and access control lists keep unauthorized and malicious traffic from breaching the network's core. To maximize service reach, adaptive hosted NAT traversal eliminates the roadblocks to incoming calls created by premises-based NAT/firewalls. Acme Packet's equipment is field-proven with most VoIP products and has features that can solve many interoperability issues, enabling True Corporation to maximize its service reach with peering partners. Acme Packet helps optimize network efficiency and cost with the Net-Net SBC's least cost routing (LCR) feature, which enables selection of the most appropriate peering partner based upon various metrics, including cost, time of day, and available bandwidth. "In order to deliver our VoIP services that match our demands and our customers' expectations for security and quality, we needed a session border controller that fits our requirements for interoperability, as well as network security and control features," said Dr. Jay Jootar, assistant director, head of VoIP business at True Corporation. "With Acme Packet, we are able to deliver secure, advanced IP communications to residents, businesses and wholesalers, utilizing the Net-Net SBC's multiservice architecture that protects both our peering and access borders."

"True is bringing IP interactive communications to the Thailand market and our SBC plays a key role in making its service offering secure, high quality and scalable," explained Seamus Hourihan, vice president of marketing and product management for Acme Packet. "True recognizes these characteristics as essential to being successful with its new service offerings."

VocalNet selects Edgewater Networks' EdgeMarc and EdgeView for its hosted PBX solution

VocalNet Inc., a provider of next-generation integrated voice, data, and managed services, and Edgewater Networks Inc., a provider of VoIP networking and security products, announced that VocalNet Inc. has selected Edgewater Networks' EdgeMarc for their Hosted PBX solution, DigitalVoice.

The EdgeMarc will be deployed by VocalNet at customer premises as a demarcation point for their DigitalVoice offering. VocalNet recommends the EdgeMarc to customers and uses the quality-of-service (QoS) and call quality monitoring capabilities to ensure DigitalVoice service quality. For VocalNet, the EdgeMarc combined with the EdgeView VoIP Support System provides powerful troubleshooting and circuit monitoring capabilities that are demanded in the enterprise space. This powerful combination allows VocalNet to sell a high-quality service at a reasonable price while enabling them to scale their operations and business to keep up with the demands of their customers.

"We chose the EdgeMarc because of its VoIP compatibility, attractive troubleshooting capabilities and general user friendliness," said Annette Warren, VP operations, VocalNet Inc. "The EdgeMarc Network Services Gateways with the EdgeView system allow us to manage and deliver the QoS that our customers demand in a hosted PBX solution. These products ensure a good customer experience with reduced installation and problem resolution times."

In addition to QoS capabilities, the EdgeMarc allows VocalNet to offer integrated telephone adapters and Wi-Fi capabilities to their customers as part of the bundle. That, coupled with the overall user friendliness of the EdgeMarc, allows VocalNet to provide the services and quality at the price and convenience their customers look for.

"We are very excited about the new partnership with VocalNet as they introduce our simplified solutions to their DigitalVoice offering," said David G. Norman, chief executive officer of Edgewater Networks Inc. "We believe our

EdgeMarc solutions allow service providers to simplify their offerings and scale their businesses. VocalNet's bundling of the EdgeMarc coupled with the EdgeView platform is an exciting offering to the market at a price the customer can afford."

The EdgeMarc is a part of a complete solution from Edgewater Networks that also includes EdgeConnect managed power-over-Ethernet (PoE) switches, the EdgeView VoIP Support System, and EdgeView Reports server. EdgeView is a troubleshooting and setup tool that allows service providers to proactively manage and support their communications services. EdgeView Reports is an advanced yet easy-to-use reporting tool that provides valuable VoIP performance information to network planners, operators, product managers, executive management teams, and end users.

BUSINESS

VocalTec receives $9.2 million net from the sale of 11 of its 22 patents

VocalTec Communications Ltd., a global provider of carrier-class multimedia and voice-over-IP solutions for communication service providers, announced the closing of the Patent Purchase Agreement (PPA) for the sale of selected patents to Karo Millennium J.P., L.L.C. as previously announced on May 28, 2008. The transaction was successfully brokered by IPinvestments Group of Atlanta, Georgia.

Pursuant to the agreement, VocalTec sold 11 patents and certain patent-related rights, out of the company's portfolio of 22 patents. With the consummation of the transaction and the payment of all transaction-related expenses, including payment to the Office of the Chief Scientist of the Israeli Ministry of Industry (OCS), Trade and Labor, VocalTec retained net proceeds amounting to approximately $9.2 million.

VocalTec is granted a geographically unlimited, nonexclusive license to use the sold patents and other patent-related rights in connection with the development and marketing of its products.

In addition to the 11 patents sold, VocalTec retains a patent portfolio comprising 11 additional patents as well as several trademarks, including "Internet Phone" trademark and Internet domain name.

Yosi Albagli, VocalTec's president and chief executive officer, commented, "We are happy to have successfully concluded this transaction with Karo Millennium J.P., L.L.C. We believe the proceeds from this transaction will significantly boost our capabilities to continue executing our growth strategy."

Cordia's Brazilian subsidiary granted a VoIP license by the Brazilian Telecommunications Agency

Cordia Corporation, a global communications service provider of traditional CLEC and VoIP technologies, announced that Cordia Comunicações S.A., its subsidiary in Brazil, has been granted a Serviço de Comunição Multimidia (SCM) license by the Brazilian Telecommunications Agency (Anatel). SCM is a fixed telecommunications service that allows the emission and reception of multimedia information such as voice, data, and image using any means. The license allows Cordia to offer broadband Internet telephony (VoIP) and value-added services, including its Magellan service line, internationally and domestically throughout Brazil, targeting small and midsized businesses and residential customers.

Peter Collins, president of Cordia Comunicações, stated, "The soft launch of our VoIP offerings in Brazil, through a resale arrangement with another carrier, was met with great enthusiasm in Brazil. Customers are looking for a way to avoid the unusually high cost of traditional landline telecommunications in Brazil. The savings sought by the Brazilian consumer along with the rapidly growing levels of broadband penetration in Brazil gives us a great opportunity for growth. The license allows us to provide voice, data and other value added services over our own network, reducing our reliance on others and reducing costs associated with selling another carrier's service. With the issuance of the license we will commence a full commercial roll out of our product by increasing our internal telemarketing efforts and expanding our sales channels through the use of agent sales and other media outlets."

Mr. Collins continued, "Our long term objective is to deploy our own next-generation wireless IP network nodes utilizing WiMAX technology that are capable of providing mobile voice, data and value added services."

Mr. Dupré, Cordia Corporation's chief executive officer, stated, "Since the soft launch of VoIP service in the fourth quarter of 2007, our Brazil subsidiary has grown to more than $1 million in annual revenues today. With more than 190 million people and a need for communications infrastructure investments, Brazil is in the top 5 of the world's most important communications markets." Mr. Dupré further stated, "The SCM license gives us a ground floor opportunity to play a central role in the deployment of IP based communications networks and services in Brazil."

Voxitas VoIP services certified to work with the Interactive Intelligence all-in-one IP communications software suite

SIP trunking services from Voxitas (formerly NetLogic), a business-class VoIP national service provider, have been certified to work with the Interactive Intelligence all-in-one IP communications software suite to help customers more easily and cost-effectively support distributed locations.

"Our SIP trunking services, combined with the Interactive Intelligence software, give customers advanced contact center automation and enterprise IP telephony applications minus the cost and complexity of managing and growing a distributed infrastructure," said Matt Siemens, vice president of sales and marketing for Voxitas.

"Our customers are increasingly turning to SIP trunking to meet their voice-over-IP scalability requirements," said Joseph A. Staples, senior vice president of worldwide marketing for Interactive Intelligence. "The Voxitas services met our stringent voice quality standards so joint customers are assured the best service delivery and performance on the market."

Also certified to work with the software are the Voxitas TrueComposure and TrueDirection products, which deliver disaster recovery and automated call distribution services.

A separate marketing agreement enables Interactive Intelligence to use Voxitas SIP trunking resources for the purpose of education, training and demonstration.

The Interactive Intelligence all-in-one IP communications software suite offers single-platform architecture with inherent multichannel processing to deliver comprehensive applications minus the cost and complexity introduced by multipoint products. For more information about the company and its products, visit http://www.inin.com.

LiveVox awarded patent for carrier-grade hosted VoIP dialer solution

LiveVox Inc., a provider of hosted dialer solutions, announced that the US Patent and Trademark Office has awarded a patent for the company's carrier-grade, hosted-VoIP dialer solution.

The patent solidifies the company's position as the first to deliver a VoIP dialer without using the public Internet and validates the technological advancement of the LiveVox solution. Because the solution directly accesses the carrier backbone using a Session Initiation Protocol (SIP) interface, LiveVox can offer hosted dialing services with unlimited capacity, higher quality, and lower cost than premises-based or other hosted dialing solutions. Other hosted solutions use either the public Internet or a hybrid of VoIP and analog protocols.

"Patenting our hosted, VoIP dialer demonstrates that LiveVox is a true innovator in the market and the first to offer a comprehensive product that addresses our clients' current and future needs at the lowest cost in the industry," said Louis Summe, chief executive officer, LiveVox. "The purpose of innovation is to deliver better results for our clients.

This platform provides optimized dialing strategies, better agent productivity and increased right-party connections. Today's economic climate dictates that collectors implement solutions that allow them to resolve debt more quickly, which often means having to contact more people to find those able to pay."

With LiveVox, call centers with heavy outbound requirements, such as credit and collection organizations, can meet growing volume needs without incurring costly per-line fees. LiveVox has replicated the key features of the traditional dialer and delivered them in a hosted solution, allowing call center and collection organizations to consider new strategies, such as replacing hardware-based dialers, to boost productivity without purchasing additional infrastructure or software licenses.

Junction Networks' VoIP service to be offered preconfigured for Microsoft Response Point

Microsoft Corp. has selected Junction Networks as a preferred SIP trunking and gateway service provider for its Microsoft Response Point small-business phone system. The Response Point Service Pack 1, generally available today, will feature services from Junction Networks preconfigured for simple account activation and maintenance.

"Small-business people who want their PBX in-house and IP-based have another important decision to make in VoIP service providers," said Rob Wolpov, president, Junction Networks. "They can realize significant telecom savings, flexibility and location independence, but only if the service is reliable, and voice quality comparable with the PSTN. We view Microsoft's choice as a major validation of our service quality."

Microsoft chose Junction Networks for the provider's strict adherence to SIP standards, its commitment to customer satisfaction, and its established phone service for small to medium-sized businesses. Response Point customers will enjoy the convenience of Junction Networks' simple account setup process: The SIP-based voice service from Junction Networks is preconfigured in the Response Point Administrator software. A small business needs only a working Internet connection to acquire service and lines in minutes.

"We thoroughly tested the service from Junction Networks and are pleased to align with them to deliver Response Point as a complete VoIP solution for small businesses," said Xuedong Huang, general manager of Microsoft Response Point. "With our focus on ease of use, and Junction Networks' high-quality SIP trunking and gateway service, small businesses will experience all the benefits of the magic blue button for voice dialing with reliable VoIP service."

Response Point is sold with unique "blue-button"-equipped IP phones from Aastra Technologies Ltd., D-Link Corp., and Quanta Computer, and employs speech recognition that allows users to perform basic and advanced phone functionality with spoken commands.

Junction Networks is offering Response Point customers access to a special promotion that waives the $9.95 monthly service charge and supplies two free phone numbers, one of which is toll-free, for the first 30 days.

The new account will include a credit of $7 toward inbound and/or outbound calling. Junction Networks does not require a long-term contract or commitment, and there are no penalties for cancellation at any time.

Junction Networks recently launched its own authorized reseller program, in addition to supporting Microsoft's broad network of resellers, to help it reach and service small-business customers with Microsoft's offering and its own trunking/gateway service.

Telchemy announces over 30 million units of VQmon shipped

Telchemy Incorporated, a provider of voice-over-IP and IPTV fault- and performance-management technology, announced that over 100 equipment manufacturers have successfully integrated Telchemy's VQmon software into a wide range of products, such as IP phones, VoIP gateways, routers, session border controllers, probes, and analyzers.

Over 30 million units of VQmon have been shipped, representing an annual growth rate of over 200 percent. This number does not include an estimated 100 million additional shipments by a number of large ISPs who license VQmon on a subscription model basis.

"Telchemy's market leading position is based on solid technology, a deep understanding of VoIP and IPTV performance and an excellent product," said Olga Yashkova, an industry analyst with Frost & Sullivan. "They have a clear vision of the future of multimedia performance management and a proven ability to execute this vision."

The VQmon product family incorporates the following:

- VQmon/EP, ideal for IP Phones and Gateways to provide real-time VoIP performance measurement and reporting. VQmon/EP generated QoE reports can be analyzed using a "collector" such as Telchemy's SQmediator.
- VQmon/SA, designed for Routers, Probes and Analyzers to provide in depth analysis of VoIP traffic and IP performance
- VQmon/SA-VM and VQmon/HD, integrated into Routers, Probes, Analyzers, and IP Set Top Boxes, providing a rich set of real time IPTV and IP Videoconferencing performance metrics.

VQmon products are compact, reliable, efficient, and highly portable software agents that are integrated into a wide range of software and hardware platforms, producing accurate real-time, actionable, performance and QoE metrics for IP voice and video traffic. VQmon is unique in its ability to model the impact of time varying impairments on user perceived quality, which has been shown to materially improve the accuracy of QoE metrics.

"We are pleased that VQmon has achieved such a strong market position, deployed through our OEM customers into Carrier, Cable and Enterprise networks internationally," said Alan Clark, president and CEO of Telchemy, "We work hard to ensure that we provide the great technology and support that enables our customers to be highly successful and to enable network operators and network managers to provide reliable, good quality services to their users and subscribers."

MARKET INTELLIGENCE

IP telephony holds strong in the face of unified communications hype, IDC says

Despite escalating buzz and hype around the unified communications (UC) solutions from Microsoft and IBM, these solutions have had minimal impact on the growth of IP telephony lines, which recorded shipments of 30.9 million in 2007, a new study from IDC reveals.

"By the end of 2007, Microsoft and IBM shipped their first versions of unified communications solutions. So, IP telephony vendors no longer had to speculate about what effect the desktop-based collaborative environments with IP telephony would have," said Nora Freedman, a senior analyst in the Enterprise Networks group at IDC.

Cisco, Avaya, Nortel, and Siemens were the leading four vendors in the worldwide IP PBX market based on their market results for end-user revenue in 2007, IDC's report shows. In the IP PBX market, Cisco gained the most market share, while Alcatel-Lucent lost the most market share relative to 2006. The other IP PBX vendors had nominal changes in market share year over year.

For IP phones, Cisco maintained its dominance in the market as the leading vendor in terms of both hardware desktop IP phone shipments and end-user revenue. As seen in the IP PBX market, Alcatel-Lucent suffered the greatest decline in year-over-year market share, based on end-user revenue for IP phones.

IDC's research reveals that some potential threats to the IP PBX market include desktop collaborative environments, open source IP PBX, and hosted VoIP services. UC mobility solutions are the primary threat to desktop IP phones.

IDC recommends the following actions in these markets:

- For IP PBX vendors, IDC believes it will be crucial to solidify their relationship with Microsoft and IBM, putting further emphasis on communications expertise and solutions as well as building an ISV ecosystem to help software developers build applications on their telephony platform.
- For IP phone vendors, it will be important to assess how the adoption of UC software clients may diminish the importance of the desktop phone. In addition, IDC recommends examining how IP phone vendors can incorporate videoconferencing capabilities and/or integrate with full-room telepresence solutions.

IDC's study, Worldwide IP PBX and Hardware Desktop IP Phones 2007 Vendor Shares, (IDC #212377) highlights the shipment and revenue market shares for enterprise IP PBX (pure and hybrid) systems, IP telephony lines/seats, and desktop hardware IP telephones for 2007.

Digium ships 4 millionth port to support demand for Asterisk phone systems

Digium Inc., the Asterisk Company, announced delivery of the 4 millionth Digium port for connecting telephony systems based on Asterisk to communications networks. The number is an indicator of the rapid rise in popularity of Asterisk, the world's most widely used open source telephony platform, for managing voice communications and, increasingly, integrating voice with corporate data. Digium has become the preferred source for the interface cards — each of which contains one or more ports that connect phone lines to communications networks — based on quality, flexibility, and performance.

Asterisk allows organizations of all sizes and industries to decrease cost of ownership and improve control over voice communications. It offers all the features of expensive corporate phone systems and the ability to be implemented in a variety of ways and comes at a price that's easy to afford. Alongside Asterisk, Digium offers high-quality interface cards that connect the phone system to different types of networks, including analog telephone service (POTS) and digital E1/T1/PRI or BRI. The company has, since 2001, focused on delivering a wide range of options to satisfy the needs of organizations working with different types of networks around the world. Such concentration has allowed Digium to become the vendor of choice for hardware to support Asterisk-based applications.

Core to Digium's success is its global network of 25 distributors and more than 300 resellers. Ashley Furniture, the top-selling brand in home furniture in North America, purchased the 4 millionth port for their Switchvox SMB system from Digium reseller and online retailer

.e4 Technologies, which in turn received the board from Digium distributor NETXUSA.

"We're a cost-conscious company and we want to purchase products getting the best possible value," said Brad Hermsen, network services manager at Ashley Furniture Industries. "At the same time, we can't compromise on quality and features in our phone systems, so we looked to .e4 Technologies when we selected the components of a phone system that really supports the way we do business."

Michael White, principal of Traverse City, Michigan-based .e4 Technologies, said, "Our unwavering focus on customer satisfaction is at the root of .e4's rapid growth, and we've found that Digium hardware and software help us deliver solutions that make our customers very happy. We're proud of our work for Ashley Furniture because it met their specific business needs. The breadth and quality of the Digium product line help us deliver this kind of top-quality yet cost-effective solution to customers every day."

NETXUSA offers Digium voice-over-IP products and services to independent resellers, who in turn service end users. Tom Boone, NETXUSA's CEO, commented, "Digium has been a strong partner since 2001 and we've seen demand for their products grow at an increasing rate as more companies look to open source and to Asterisk. Digium's support for our resellers and for us impacts our bottom line."

Digium delivers a range of hardware and software for use with Asterisk, including cards, Asterisk toolkits, and turnkey systems that integrate voice and data using Switchvox software and Digium appliances. Digium's cards are protected by a five-year warranty, and all of Digium's commercial hardware and software is covered by its Exceptional Satisfaction Program (ESP). As a part of the ESP program, customers not satisfied with the performance of their Digium hardware can return it for a full refund.

"In addition to designing hardware for organizations using any variety of types of networks, Digium has focused on creating and delivering the open source telephony industry's most partner- and customer-friendly programs," said Danny Windham, CEO of Digium. "The quality of our products is equaled only by the quality of our programs, which together have allowed us to hit the 4 million port milestone."

Details on Digium's hardware are in a new white paper available at www.digium.com.