The SIPconnect Interface Specification:

A Standards-based Approach to Direct IP Peering for IP PBX and VoIP Service Provider Communications

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1.0 Executive Summary

Today's Voice over IP (VoIP) communications systems offer customers a wealth of advanced features, as well as the ability to easily add new capabilities as requirements change and evolve. IP PBXs in particular are positioned to grow and potentially overtake traditional TDM-based PBX deployments in 2005. Moreover, VoIP service providers are rapidly expanding their networks and offering new services to keep up with this growth.

One of the major drawbacks of today's next-generation communications architecture is its reliance on offloading packets to traditional TDM networks at various points during a transmission. While this is necessary in many cases to interface with analog equipment at termination points, in some instances a pure end-to-end IP connection is possible. Equipment manufacturers and service providers have largely reached a consensus that the Session Initiation Protocol (SIP) is the best method for sending voice packets over IP networks.

But in today's network architecture, when an IP PBX interfaces with a service provider's network using SIP, important information is generally stripped out of the transmission and only basic functionality is achieved. Intelligent end-user identification information can be lost, and network issues such as quality of service (QoS) and application layer security are not consistently addressed. IP PBXs from different equipment manufacturers use varying methods to interface with service providers' networks. Consequently, it is difficult if not impossible to currently preserve and extend vital call and application-related information and thus leverage the power of next generation applications from one IP PBX to another.

VARs, Interconnects and other channel partners tasked with installing new IP PBXs for their business customers also face major challenges in managing the PSTN interconnection and providing necessary security measures, tasks that often require performing custom configurations on a customer-by-customer basis, significantly adding to the cost and complexity of their deployments.

The draft SIPconnect Interface Specification launched by Cbeyond Communications with support from vendors Avaya, BroadSoft, Centrepoint Technologies, Cisco Systems, and Mitel Networks, addresses this problem.

Driven by Cbeyond Communications service requirements, SIPconnect defines one method for interconnection between IP PBXs and VoIP service provider networks. SIPconnect specifies a reference architecture, required protocols and features, and implementation rules necessary for seamless peering between IP PBXs and VoIP service providers. Cbeyond is making the document public in order to gain industry-wide approval of the interface and service.

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2.0 Business VoIP Communications: New Choices and Options in Today's Communications Marketplace

2.1 Leading Trends in the IP PBX Market

The IP PBX market has been like an express train in recent years, with revenues set to approach the $10 billion mark this year. 2005 is actually expected to be the turning point for this communications equipment segment, when revenues will likely surpass those of traditional, TDM-based PBXs. And by 2008, 55 percent of all available PBXs are expected to be IP enabled.[1]

The rapid growth of this market has created a unique opportunity for businesses that wish to upgrade and converge their communications network infrastructures and deploy exciting new IP-based features and capabilities. This type of converged infrastructure enables them to save money on recurring charges for maintaining analog phone lines and the transport costs associated with PSTN traffic.

Of course, IP PBXs aren't the only train in town for business communications solutions. Traditional, TDM-based key, hybrid and PBX systems are still widely used, and new hosted IP PBX services are also starting to make inroads into the marketplace. VoIP-based solutions typically offer a wealth of features not available through traditional systems including desktop integration for presence-based applications, simple Web-based system management, and a converged network infrastructure that eliminates separate wiring for phone and data traffic.

The rationale for choosing an IP PBX is extremely compelling for customers. And having such a communications system in place enables VoIP service providers to offer a complementary host of enhanced features to their customers like wireless/wireline integration, soft-phone support, teleworker/remote office applications, and click-to-dial capabilities.

The challenge for service providers, however, is to be able to extend these next generation capabilities beyond the enterprise and over the wide area network, to some other remote end point. In order to extend these features in the most efficient and seamless manner, service providers need to find a way to directly connect, or peer, to IP PBXs on the customer premises.

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[1] Sources: Synergy Research, Frost & Sullivan, CompTIA

2.2 Today's IP PBX and VoIP Service Provider Interconnections: Unrealized Opportunities

In a typical business setup, an IP PBX is located on the customer premises. It is the interface to the customer LAN, enabling the use of IP phones, PCs, audio and video conferencing devices, wireless equipment and various other communications endpoints. An IP PBX can also serve as the interface to the PSTN, enabling the essential conversion of IP data packets to traditional analog or digital signals and vice versa, although this is typically accomplished through the use of a separate, adjunct IP telephony gateway, which can reside on the customer premises or within a VoIP service provider's network (see Figure 1).

In any case, all packets must be converted through this gateway, which inevitably introduces delay, or latency, into the transmission. In addition, the gateway typically does not perform any prioritization of voice packets over data packets, which can result in a negative impact on voice quality when some packets are delayed in reaching their destination due to network congestion.

Moreover, as voice packets are converted and routed over TDM networks, advanced IP-based communications signaling information and features can be stripped out of the transmission. Indeed, one of the main drawbacks to using broadband telephony services for business use is the inherent lack of quality of service (QoS), especially when traffic is transported over a "best effort" network such as the public Internet.

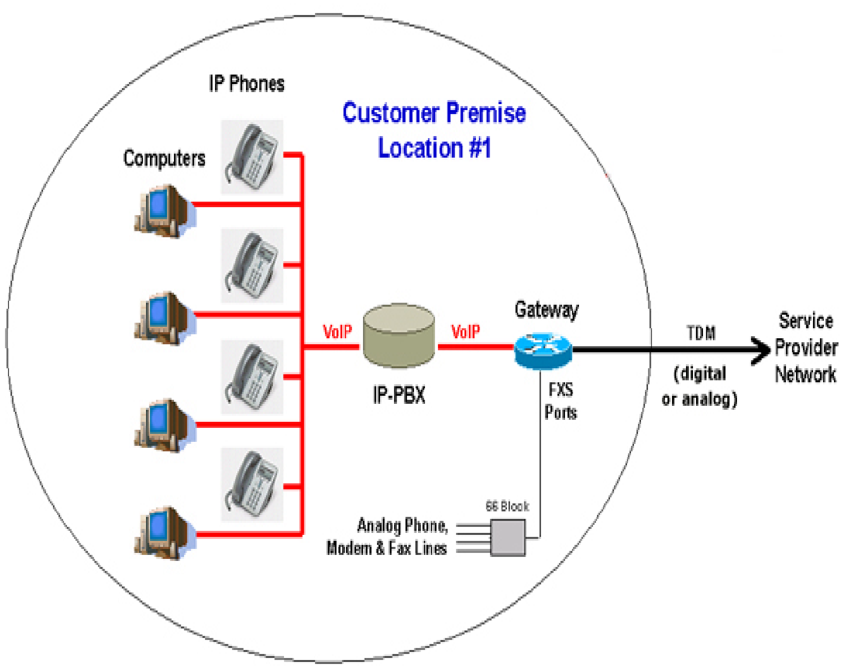

Figure 1: A Typical Customer Premises Setup using an IP PBX and VoIP Gateway

As we can see, routing VoIP traffic over TDM networks is at best a "Band-Aid" approach to next-generation IP-based communications in that it introduces a number of problems. It's inadequate for supporting the full capabilities of VoIP as important signaling information is lost, which ultimately limits the ability of service providers to deliver the rich set of features enabled by IP-based communications.

In short, TDM routing of VoIP traffic is a limited approach to next generation telephony. A much more efficient and cost effective approach, which also enables the full capabilities of packet-based communications, would be to enable IP PBXs to peer, or connect directly with VoIP service providers, eliminating the need for gateways and TDM traffic routing altogether (see Figure 2). Of course, to accomplish this, the equipment and service providers must utilize common standards so that direct IP peering may be accomplished.

To this end, the adoption of the Session Initiation Protocol (SIP) can not only help unify a number of protocols associated with VoIP, but also enable direct packet peering from a compliant IP PBX to a compliant VoIP service provider, and beyond that to another compliant IP PBX. However, there is not currently a well-defined methodology for applying SIP to the specific scenario of connecting service provider networks with IP PBXs. A standard method for interconnection that builds on SIP and other VoIP protocols, developed and approved through an industry initiative, would enable direct IP peering – revolutionizing next-generation communications.

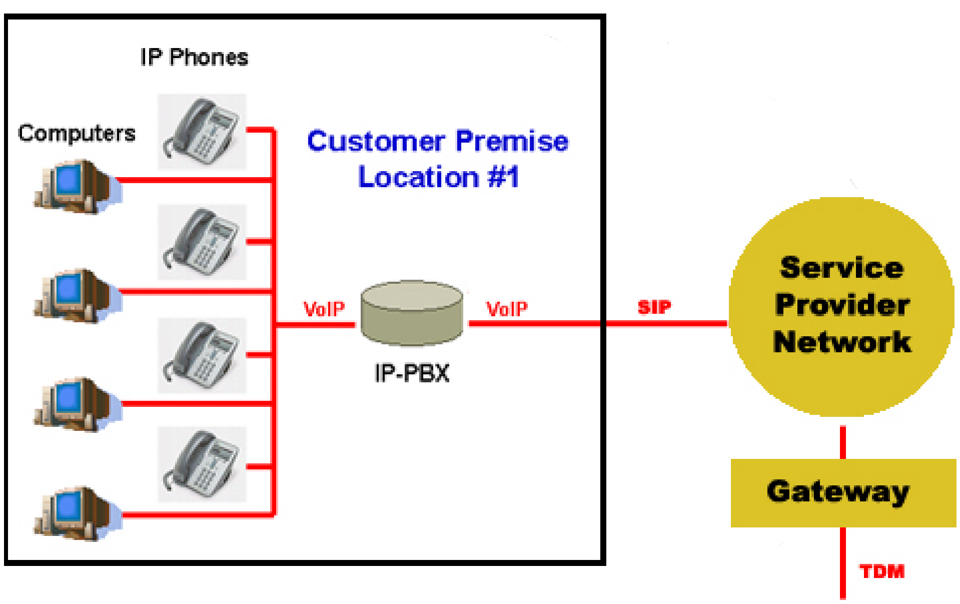

Figure 2: Direct IP Peering between IP PBX and VoIP Service Provider

3.0 The Session Initiation Protocol (SIP): An Industry Standard

3.1 An Overview of SIP and its Popularity in the VoIP Community

Session Initiation Protocol (SIP) is a text-based protocol designed to initiate, modify, and terminate interactive communication sessions among users, including voice, video, chat, and other types of multimedia applications. The current proposed standard is overseen by a working group within the Internet Engineering Task Force (IETF, www.ietf.org).

In the early days of IP telephony, the H.323 protocol was widely deployed by VoIP equipment manufacturers and covered all types of call services, from basic signaling to call control features to conferencing. As SIP gained popularity over the last few years, however, it began to replace H.323 as the protocol of choice among equipment vendors.

SIP covers basic signaling, user location, and registration, allowing support for other features via separate protocols. This makes it a lighter weight and more efficient protocol than H.323, and one that is decentralized within the network – pushing intelligence to end points like phones, phone systems and wireless devices.

3.2 The IP PBX/Service Provider Interface Challenge: Why SIP Alone Is Not Enough to Create a Workable Interconnection Approach

SIP is now considered the logical choice for routing a variety of IP traffic, and the protocol already defines ways for accomplishing interconnection between SIP-enabled IP PBXs and service providers' SIP-enabled networks. However, these methods do not address the entire scope of interconnection such as required features, security and quality of service issues, and do not offer a common, predictable solution for retaining intelligent end-user identification throughout a network transmission.

A straightforward, single industry-approved approach for accomplishing this interconnection does not yet exist. Having an industry accepted method of interconnection that builds on the fundamentals of SIP would allow interoperability among SIP-compliant IP PBXs and service providers, enabling direct IP peering.

3.3 How SIP Can Be Enhanced for Direct IP Peering Between SIP-Enabled IP PBXs and SIP-Enabled VoIP Service Providers

Currently, there are two basic ways that SIP can be expanded to meet the needs of direct IP peering. A baseline set of signaling and media implementation rules to guide interconnection efforts must be established between SIP-enabled service providers and

SIP-enabled IP PBX deployments. When the protocol is used for this purpose, certain functionality gaps must also then be addressed. An enhanced solution would define a method for service providers to interface with an IP PBX as a single logical entity, while also maintaining the ability to deliver personalized, enhanced features to individual end users at specific PBX extensions.

4.0 The Benefits of Direct IP Peering for IP Communications

4.1 A Competitive Edge for IP PBX Manufacturers

As with any healthy market, competition abounds in the IP PBX space. A large number of products are available, each addressing the different needs of a diverse group of customers. Being compliant with SIP and other popular protocols is a huge benefit for IP PBX manufacturers, allowing their equipment to interoperate as well as communicate with gateways and other equipment within the network.

Virtually all major players in the IP PBX arena boast support for SIP. But most vendors haven't given much thought to the benefits of direct IP peering. Since their equipment doesn't sit in the larger "network cloud," why should they be concerned with that type of interconnection?

The answer is simple: Direct IP peering is a huge "value add" for businesses and for service providers alike – those purchasing and interconnecting with IP PBXs. When your product supports an industry-accepted standard that addresses quality of service and security issues, reduces equipment and transport costs, increases features and functionality and eliminates the time needed to set up a proprietary interface to your IP PBX, you have a sizeable competitive edge.

4.2 Improved QoS and Security for Service Providers

For service providers, the benefits of direct IP peering are equally compelling. By boasting an industry-accepted standard for connecting with customer premises equipment, they can effectively position themselves years beyond their competitors. This type of interface enables them to offer higher quality services with advanced features tailored to IP PBX users.

And the ability to forge relationships with IP PBX vendors based on this interconnectivity can go a long way toward winning customers and establishing new relationships with the various distribution channel entities, including interconnects, system integrators and VARs.

4.3 New Features and Cost Savings for Business Customers

Business customers are the end users who will ultimately benefit from direct IP peering. They are faced with the challenge of setting up an affordable yet feature rich communication system. Many have perhaps been hesitant to deploy a VoIP-enabled solution because of quality of service issues and fears about deploying "new" technologies. And many of the advanced features enabled by IP-based communications systems – presence applications, video conferencing, one-click dialing – may not be sufficient to persuade a customer to give up their old phone system. There must be practical reasons for change.

Businesses that peer directly with their communications service providers eliminate the needs for expensive TDM gateways, and increase the efficiency with which they use local access facilities.

An additional and easily overlooked feature enabled through VoIP-based communications solutions is the ability to provide Direct Inward Dialing (DID) capabilities with lower cost service offerings.

This is a potentially huge and immediate selling point for most small business customers. While many small businesses are unable to afford a full T1 or PRI line with which DID is normally provided, direct IP peering enables VoIP service providers to offer multiple direct phone numbers through a single connection.

This means a small business customer can use multiple phone "numbers" within their business without requiring the recurring expense of separate, analog lines or expensive digital circuits for this purpose. The fact that individual employees can have their own, distinct phone numbers within a small business instead of extensions off of a main number is a huge benefit taken for granted at larger businesses.

And, of course, the presence-based applications and other enhanced end-user features that can be enabled are an added bonus. Direct IP peering extends these benefits for customers, enabling end-user information to be carried across the network intact to other VoIP-enabled destinations.

4.4 Benefits for VARs and Interconnects

One of the biggest challenges for value added resellers (VARs), Interconnects and other channel partners tasked with installing IP PBXs for their customers is the PSTN interconnection -- specifically, using gateways to interconnect to an ILEC or CLEC service provider. Quality of service problems, including latency and echo, inevitably arise due to the necessity of performing VoIP to TDM conversions, and these channel partners often find that they need to perform custom configurations on a customer-by-customer basis, significantly adding to the cost and complexity of the deployment.

The ability of an IP PBX to directly peer with a VoIP service provider offers VARs and Interconnects immediate relief from such quality of service issues. When a VoIP service provider handles the interconnection to the legacy TDM world, and effectively manages the QoS associated with this interconnection, not only are gateways unnecessary at customer locations, but the time-consuming task of custom configuring and troubleshooting such equipment is also eliminated.

Additionally, a direct IP peering arrangement also allows a number of important security-related functions to be "off-loaded" from the customer premises to the VoIP service provider network, including issues relating to NAT traversal (to allow seamless SIP connectivity through network firewalls) and other security concerns including denial of service attacks.

5.0 The SIPconnect Interface Specification

5.1 What is SIPconnect?

The draft SIPconnect Interface Specification, driven by Cbeyond Communications requirements, defines a common set of implementation rules for those who desire to interface a SIP-enabled IP PBX with a SIP-enabled VoIP service provider. It specifies which VoIP protocols must be supported, provides guidance in the areas where the protocols leave too many options, and identifies a baseline set of features that should be supported by PBXs and service providers. It's important to note that SIPconnect is not intended to be a new protocol; rather, it is being presented as a recommended set of interoperability guidelines for the interconnection of a SIP-enabled IP PBX to a service provider's SIP-enabled VoIP infrastructure.

5.2 Implementing SIPconnect: How Existing IP PBX Manufacturers Can Support the Specification

As mentioned earlier, one of the most important goals of direct IP peering is the ability for service providers to interface with an IP PBX as a single logical entity, while also supporting and delivering enhanced features to individual end users. SIPconnect supports this concept through a requirement that the IP PBX associate with a service provider using one or more high level "parent" accounts. To preserve the ability to deliver features on a per subscriber basis, the call control server may also support lower level "child" accounts.

These accounts always inherit the contact information and security credentials of their parent, but preserve the ability to provide services unique to each account.

SIPconnect establishes norms for connectivity well beyond simple population of SIP headers. It offers the following benefits:

- **Universal Approach**. SIPconnect fills a current void in the industry by providing clear instructions for IP peering between IP PBX and VoIP service providers. This will accelerate adoption and reduce development costs for PBX developers and service providers.

- **Customer Cost Savings.** Direct IP peering makes VoIP gateways unnecessary, and extends the benefits and savings of VoIP communication systems (i.e. DID and conferencing) into the network and beyond to compliant destinations.

- **Transparent Feature Transport.** Individual end-user information is passed from the IP PBX to the network intact (and with application layer security intact) rather than being lumped into one account. This enables important information for presence and other user-based applications to travel through the network to terminating IP PBXs without being stripped out.

- **Quality of Service.** Important transport layer issues are defined, including: QoS configuration, echo cancellation, method for DTMF relay, packetization rates, codec support, and dealing with fax and modem traffic.

- **Security.** Service providers act as the public interface (for DNS queries, etc.) for communicating with SIP devices via their SIP proxies (i.e. a session border controller). This enables them to add security at the application layer for customer communications.

Most importantly, these recommendations use and build upon guidelines published in RFCs dealing with the SIP family of protocols. The SIPconnect guidelines are intended to compliment (and potentially guide implementation) of these RFCs, and not to replace them in any way.

5.3 Case Study: BeyondVoice with SIPconnect - A Managed Services Provider Voice and Broadband Internet Package from Cbeyond Communications

As the pioneering service provider behind SIPconnect, Cbeyond Communications is the first service provider to offer a service based on SIPconnect. The company is a Managed Services Provider offering integrated IP voice and broadband Internet service packages as well as IP-based applications for small businesses. Cbeyond is now offering an enhancement to its BeyondVoice series of services with support for direct IP peering with IP PBXs using the SIPconnect Interface Specification. *BeyondVoice with SIPconnect* enables Cbeyond small business customers to connect an IP-based phone system directly to Cbeyond's VoIP network - without requiring an investment in VoIP gateways or the recurring expense of traditional TDM-based voice services such as analog lines or PRI circuits.

Cbeyond is also planning a series of enhanced features specifically designed for SIPconnect users. These will include an online management portal that enables individual users behind an existing IP PBX to access a Web-based personal communications tool to place and manage real-time calls. *BeyondVoice with SIPconnect* packages start at $495 per month and support 1.5 Mbps to 4.5 Mbps of Internet access and anywhere from five to 48 active calls. A customer using SIPconnect will also enjoy the benefits of multiple DID phone numbers, while choosing the appropriate BeyondVoice package based on the actual number of active or simultaneous calls they anticipate making at a given time.

5.4 The Future of IP Peering for IP Communications: Opportunities for Industry Input

In addition to Cbeyond Communications, a number of leading IP PBX equipment vendors have been instrumental in developing the SIPconnect Interface Specification, including Avaya (www.avaya.com), BroadSoft (www.broadsoft.com), Centrepoint Technologies (www.talkswitch.com), Cisco Systems (www.cisco.com), and Mitel Networks (www.mitel.com).

Cbeyond is making the draft document public in order to gain industry-wide approval of the interface and service. The draft SIPconnect Interface Specification is available on the SIPconnect website at www.sipconnect.info, and input and discussion from those who will utilize direct IP peering are vital to the further development of the interface and expansion of SIP deployment.

To learn how your company can become part of the SIPconnect initiative, and to access the complete draft SIPconnect Interface Specification and related documentation, visit the SIPconnect website listed above.

6.0 Conclusion

Next-generation network solutions that utilize gateways and other equipment to enable IP networks to interface with TDM networks have been essential in the evolution of VoIP and enhanced IP communications. These bridging solutions between legacy networks and new IP networks have contributed to the growing deployment of IP PBXs and other next-gen communication solutions.

Now that VoIP-based communications solutions like IP PBXs are being deployed to customers at breakthrough rates, and VARs, Interconnects and VoIP service providers are increasingly looking for more efficient and cost effective ways to serve their customers, a better method of interfacing IP PBXs with service provider networks is necessary.

In fact, the ability to transport the intelligent end-user information associated with enhanced features from a customer premises through the network and beyond to another customer is vital for the overall growth of the IP telephony marketplace. Direct IP peering enables this next generation interconnection and the SIPconnect Interface Specification offers a revolutionary starting point for equipment manufacturers, service providers, and VARs and Interconnects.

Indeed, the ability of an IP PBX to directly peer with a VoIP service provider offers VARs and Interconnects immediate relief from the need to troubleshoot quality of service issues, and allows a number of important security-related functions to be handled by the VoIP service provider network, including NAT traversal for seamless SIP connectivity through network firewalls.

The next step is up to the industry at large. Ideally, more equipment vendors and service providers alike will contribute to the development of SIPconnect, resulting in a working standard that will benefit the entire industry.

The success and momentum of VoIP and SIP is undisputed, and direct IP peering is the next plateau. The true power of IP communications depends on leaving behind legacy interfaces and services that reduce new technology to outdated common denominators.

For More Information

To learn more about the SIPconnect Interface Specification, visit www.sipconnect.info.

About Robins Consulting Group

Robins Consulting Group (RCG) is a leading marketing communications consultancy providing an array of marketing and other services to the IP telephony industry. Marc Robins, an internationally recognized authority in the field of IP telephony and emerging new communications technologies, founded RCG in 2003. Prior to RCG, Mr. Robins served as vice president of publications and trade shows, associate group publisher and group editorial director at TMC, publisher of the trade magazines *Internet Telephony*, *Communications Solutions*, and *Communications ASP*, and producer of the *Internet Telephony* Conference & EXPO trade shows, for which he also served as chief architect and conference co-chairman. For more information about Robins Consulting Group services, call 718-548-7245 or e-mail robinsconsult@optonline.net. This white paper was commissioned by Cbeyond Communications.