SIP connect

The SIPconnect Technical Recommendation:

An Industry-Accepted Approach to
Direct IP Peering for IP PBX and
VoIP Service Provider Communications

SIP FORUM

1.0   Executive Summary

As enterprises and service providers migrate services to VoIP, a new demand has emerged for IP peering, or an end-to-end VoIP connection, *between* enterprises and services providers. 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 have gained strong traction in the marketplace, and by most accounts IP PBX deployments have now overtaken traditional TDM-based PBX deployments. Moreover, service providers are rapidly evolving their networks using VoIP technology to improve efficiency and deploy new services.

To that end, a new method of connecting VoIP networks to other companies and service providers is needed. Companies are running into the limitations of using traditional TDM technology (such as trunking gateways) to make their links. When PRI or analog connections are used to connect to service provider networks, the features that can be supported between the networks are limited to the least common denominator of features that can be supported on the PRI or analog line connecting them to each other. The quality of the media connection is also diminished as traffic is converted from VoIP to TDM and back again between networks.

Equipment manufacturers and service providers have largely reached a consensus that the Session Initiation Protocol (SIP) is the best protocol choice to use in the interconnection outlined above. However, choosing SIP alone is not enough. As a large family of IETF RFCs, support of SIP rarely implies support of all RFCs or even all functionality of a particular RFC. Further, there are often multiple ways of accomplishing the same technical task in SIP, which can complicate interoperability with "extra" choices. Finally, the interconnection of VoIP networks encompasses issues beyond signaling, such as media and security, which also need to be addressed to define a predictable interface model.

The SIP Forum's **SIPconnect™ Technical Recommendation** has been developed to address these problems. Organized and maintained by the SIP Forum, SIPconnect™ defines a method for enabling direct peering between a SIP-enabled service provider network and a SIP-enabled enterprise network. SIPconnect™ specifies a reference architecture, a minimal set of IETF and ITU-T standards that must be supported, provides precise guidance in the areas where the standards leave multiple implementation options, and specifies a minimal set of capabilities that should be supported by the service provider and enterprise networks.

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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 exceed the $10 billion mark this year. Last year (2005) was a turning point for this communications equipment segment, when revenues surpassed those of traditional, TDM-based PBXs. And by 2008, at least 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 and IP Centrex 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 a way to directly connect, or peer, to IP PBXs on the customer premises.

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

In a typical business communications network, an IP PBX serves as the interface to the enterprise 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).

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

In any case, all packets must be converted to TDM through this gateway, which inevitably introduces delay, or latency, into the transmission. Delay and latency can result in a negative impact on voice quality. 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 a TDM connection between VoIP networks is the introduction of an unnecessary least common denominator that limits the potential of possible features and applications.
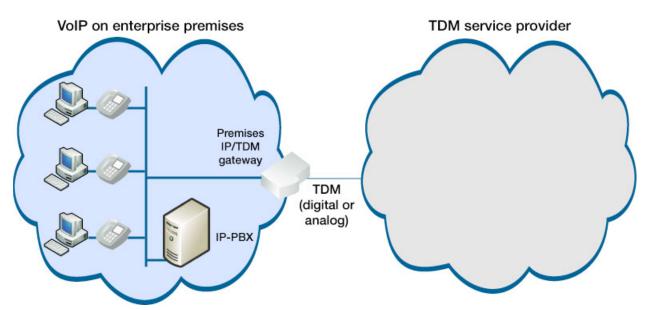

Figure 1: A Typical Customer Premises Setup using an IP PBX and VoIP Gateway

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 is 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. A well-defined methodology for applying SIP to the specific scenario of connecting service provider networks with IP PBXs is necessary, as 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.
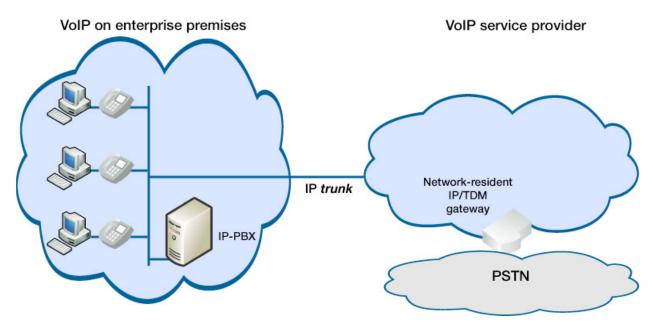
Figure 2: Direct IP Peering between IP PBX and VoIP Service Provider

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 is inherently flexible and already defines a number of methods for accomplishing interconnection between SIP-enabled IP PBXs and service providers' SIP-enabled networks. But herein lay the rub: the choice of multiple approaches for accomplishing such an interconnection in the SIP standard makes interoperability unnecessarily complicated.

In addition, SIP doesn't define the entire scope of the interconnection challenge, such as issues that extend beyond SIP-based signaling to include bearer-path traffic. And while SIP excels at the management of user-to-user identity, it is not inherently good at managing hierarchical logical identities – a requirement for the effective peering of IP-PBXs that represent multiple individual users as well as optional company-level identities that group users into unique service accounts and consolidate authentication credentials.

Therefore, a straightforward, industry-approved approach for accomplishing this interconnection is required – a specification that describes a single option when multiple options exist. Moreover, such an industry-accepted method of interconnection would ideally build on the fundamentals of SIP to allow interoperability among SIP-compliant IP PBXs and service providers, enabling true, direct IP peering.

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

SIP can be used as a foundation to meet the needs of direct IP peering, and by specifying minimum sets of functionality, a baseline list of signaling and media implementation rules to guide interconnection efforts can be established between SIP-enabled service providers and SIP-enabled IP PBX deployments.

The work of SIP Forum has largely been the definition of a IP PBX to service provider peering architecture, the specification of which SIP RFCs must be supported by each element if the architecture, and specific recommendations on how to address the entire scope of interconnection such as required features, security and quality of service issues, and offer a common, predictable solution for retaining intelligent end-user identification throughout a network transmission.

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. 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 Distributors and Channel Partners

One of the biggest challenges for distributors and channel partners -- including value added resellers (VARs), and Interconnects -- 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 SIPconnect™ Interface Specification 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 is important to note that SIPconnect™ is not intended to be a new protocol; rather, it is 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.

SIPconnect™ was originally conceived and developed by Cbeyond in early 2005, with support from Avaya, Broadsoft, Cisco Systems, Mitel Networks, and Talkswitch. Soon afterwards, SIPconnect™ was moved into the SIP Forum -- the leading industry organization dedicated to the advancement and adoption of IP Communications products and services based on SIP -- to facilitate an industry-wide IETF-style review and evolution of the specification. As a result of the SIP Forum's participation and resulting wide-ranging scrutiny of SIPconnect, a new version was subsequently developed that greatly improved on the initial draft.

In June 2006, the SIPconnect™ Technical Recommendation achieved "Proposed Recommendation" status, meaning the specification is complete and awaits validation through actual implementations.   The SIP Forum is now also officially charged with organizing and maintaining the SIPconnect™ specification, and is responsible for the evolution of the initiative.

5.2   SIPconnect™ Reference Architecture

The following reference diagram (Figure 3) outlines the common functional elements required to support the SIPconnect™ Interface Specification. It is important to note that SIPconnect™ treats the network elements in the diagram as separate physical components for the purposes of illustration only. It is perfectly acceptable for an equipment manufacturer to combine one or more of these functions in a single physical device.

For example, a manufacturer many choose to integrate the SIP Proxy Server function with the IP PBX function, while another manufacturer many choose to integrate the SIP Proxy Server, IP PBX and Firewall functions. Both implementations (as well as other combinations) are considered to be

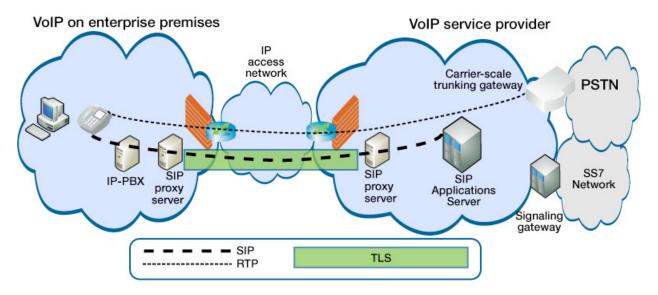conformant as long as they fully adhere to the individual rules governing each of the integrated functions.

Figure 3: SIPconnect™ Reference Architecture

5.3   Key Benefits of Implementing SIPconnect

SIPconnect™ establishes norms for connectivity well beyond the simple population of SIP headers, and offers the following key 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.** Requirement for Transport Layer Security along with defined approaches to identity and authentication provide a secure model of IP PBX to Service Provider peering.

5.4   Implementing SIPconnect™: What Can Existing IP PBX Manufacturers and VoIP Service Providers Do to Support the Specification

As the caretaker of the SIPconnect™ Interface Specification, the SIP Forum is interested in industry-wide adoption of the interface and service, and IP PBX manufacturers and VoIP service providers have an important role to play in ensuring the continued evolution and adoption of the SIPconnect™ specification.

Specifically, it is vital that VoIP industry participants who will utilize direct IP peering support the specification in the following ways:

- Obtain a copy of the current draft of the SIPconnect™ specification, and commit to reading and understanding the specification.

- Embrace SIPconnect™ and deliver services in full accordance with the specification.

- Join the SIP Forum (if not currently a member) and commit to participation in upcoming interoperability events.

- Contribute input and engage in discussion regarding the further development of the specification and expansion of SIP deployment.

To learn how your company can become part of the SIPconnect™ initiative, and to access the complete SIPconnect™ Interface Specification and related documentation, visit the SIP Forum SIPconnect™ web pages at www.sipforum.org/sipconnect.

6.0   About the SIP Forum

The SIP Forum is an industry organization with members from the leading SIP technology companies. Its mission is to advance the adoption of products and services based on SIP.

The forum directs technical activities aimed at achieving high levels of product interoperability, provides information on the benefits and capabilities of SIP, and highlights successful applications and deployments. The Forum promotes SIP as the technology of choice for the control of real-time multimedia communication sessions throughout the Internet, corporate networks, and wireless networks. SIP may also be used for new types of communications, such as instant messaging and application level mobility across various networks, including wireless, and across user devices.

Work on SIP is accomplished primarily in the IETF SIP working group, and at SIP bake-offs. SIP has however manifold interactions with other areas, such as the next generation wireless internetworks, QoS, payments and security. The objective of the SIP Forum is thus to facilitate the integration of SIP with such other areas of work on the Internet.

6.1   The SIP Forum Mission

The SIP Forum's mission is to advance the adoption of products and services based on the Session Initiation Protocol.

To accomplish this mission, the Forum:

- Advances product interoperability by holding live interoperability test events, and by defining and creating operational compliance tests.

- Creates white papers, implementation guides, recommendations, and other technical documents dealing with issues that fall outside the scope of the IETF or other relevant standards bodies.

- Builds awareness about what existing and emerging SIP-compliant technologies can do for users and customers.

6.2   SIP Forum Scope

The SIP Forum is explicitly not a standards-setting body. The Internet Engineering Task Force (IETF) defines the core SIP protocol.

The activities of the Forum are therefore directed at complementing the activities of these standards bodies, though the Forum's activities are often informed by, and guided by the leadership of the standards organizations. These activities, from time to time, do involve working with the industry in creating best practice Recommendations documents specifying how to utilize standard SIP protocol techniques to create interoperable products and services. In addition, the Forum undertakes activities guided by the needs of its membership as determined through interactions with that membership.

Finally, the Forum provides opportunities for its members for inter-personal networking, publicity in various venues, and product testing events. For instance, the Forum creates test cases testing the compliance of modules, products, and systems to SIP in order to promote the interoperability of these items with each other. Standards bodies and members together are likely to guide the definition of these documents or test cases. Test cases developed in this manner may then be used at a SIP Forum sponsored test event to help vendors assure high quality and interoperability.

6.3   SIP Forum Membership

The SIP Forum is the meeting place for developers of commercial SIP based services and Internet technology, such as IP phones, PC clients, SIP servers and IP telephony gateways. The SIP Forum is open to everyone that is willing to contribute to spread information of SIP and accepts the architectural model that SIP relies on. Individual membership is free. Organizations pay a yearly fee to cover the administration costs of the Forum.

Visit www.sipforum.org for more information.

7.0   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 and other channel partners 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 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 about the SIPconnect™ Technical Recommendation, visit www.sipforum.org/sipconnect

About the Author of this White Paper

Marc Robins is an internationally recognized authority in the field of IP telephony and emerging new IP communications technologies. Marc is the Chief Technology Evangelism Officer of RCG (Robins Consulting Group). Founded in 2003, RCG is an IP communications industry consultancy providing an array of marketing, research and advisory services. Marc also serves as the president of the SIP Forum LLC (the operating U.S. subsidiary of the SIP Forum Swedish parent company), and managing director of the SIP Forum parent. Prior to founding 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 RCG, call 718-548-7245 or e-mail info@robinsconsult.com.

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