

SIP Trunking for Small and Medium Enterprises

A joint white paper from Colt Technology Services
and Siemens Enterprise Communications

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Siemens Enterprise Communications

1. Executive summary

Many small and medium enterprises have upgraded to pure, or hybrid IP PBX's for their telecom infrastructure. These companies that have made the move to IP Telephony are looking for new ways to leverage their investment and drive additional cost reductions, and/or employee productivity increases. SIP Trunking has been shown to provide 30 per cent to 50 per cent reduction in voice and data connectivity costs. Additional costs savings are possible from reductions in LD charges and increased use of VoIP. SIP Trunking solutions are available from many service providers all over the world. Siemens Enterprise Communications and Colt Technology Services have worked together to verify interoperability between Siemens' SME IP PBX solutions and Colt's SIP Trunking services.

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2. IPT is the Foundation

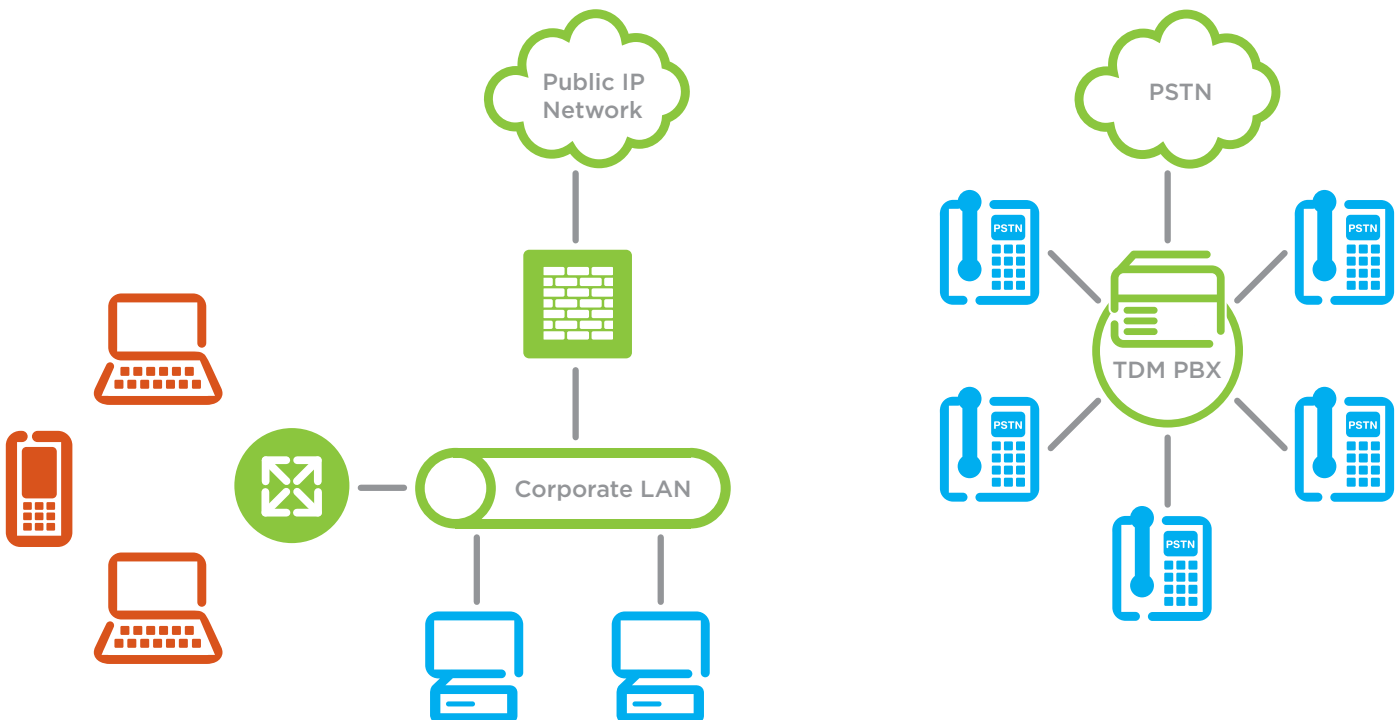
Over the last decade, corporate telephony has changed substantially. Many companies have replaced their legacy Time Division Multiplexing (TDM) telephone systems with Internet Protocol Telephony (IPT). Some companies have made the move to pure IP PBX's, but many others are protecting their telecom investment by installing converged or hybrid PBX's that support both TDM and IP endpoints. IPT allows companies to significantly reduce operating costs and simplify administration by merging their voice and data networks internally. IPT offers companies tremendous deployment flexibility - depending on budget priority (Capex vs. Opex), small and medium sized enterprises (SME's) can choose to install the IP PBX within their own

IT centre or connect to a leased IP PBX hosted by a third party. More than just cost reductions, IPT also enables many innovative, high Return On Investments (ROI) telecommunications technologies such as Voice over Internet Protocol (VoIP), Unified Communications (UC) and Session Initiation Protocol Trunking (SIP Trunking).

VoIP converts your speech into digital packets and sends them out across a data network to their destination. The packets are routed in the most efficient way possible over a data network and then reassembled at the other end to become the sound of your voice. VoIP, first successful in the larger enterprise, has become more appealing for SME's. VoIP technology is field-proven

and mature. It offers many benefits over conventional voice systems, including; lowered Total Cost of Ownership (TCO), reduced local and long-distance telecom costs, and improved system security and reliability. VoIP is so successful, that most of the large telecommunications operators are using it to carry their own long-distance traffic. A 2010 report from In-Stat forecasts that VoIP penetration among US businesses will almost double from 42 per cent in 2009 to 79 per cent in 2013. VoIP solutions are available from most major telecom vendors including Siemens Enterprise Communications. Siemens Enterprise Communications offers a complete portfolio of IPT solutions for companies of every type and size.

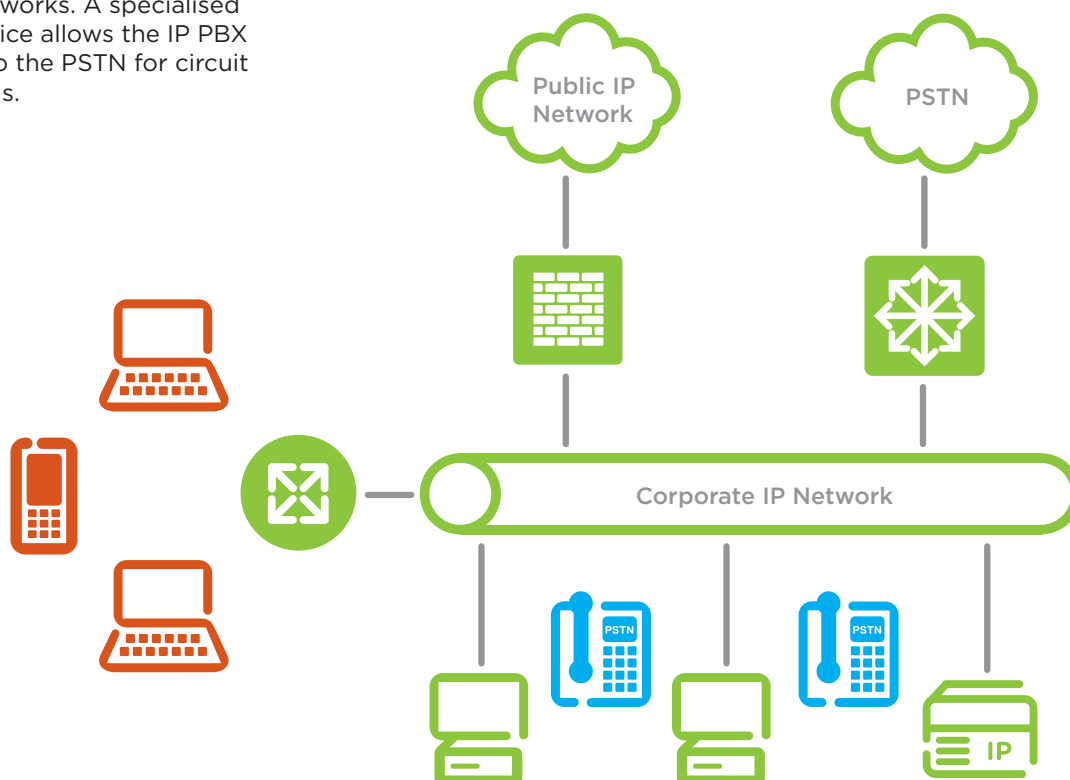
Figure 1: Before IPT, companies were required to manage separate networks for voice and data.



SIP Trunking enables convergence of enterprise voice and data beyond simply using a shared internal network infrastructure. It extends the voice and data convergence using a shared IP-based connection to the outside world (the PSTN and public internet, respectively). Instead of connecting to the Public Switched Telephone Network (PSTN) through their own gateway device, the company's IP-PBX has a broadband connection to an Internet Telephony Service Provider (ITSP).

The ITSP can forward VoIP traffic directly, or as required, translate calls onto the PSTN. According to Frost and Sullivan, the number of SIP Trunking lines is expected to reach 14.5 million in 2012, significantly up from less than 1 million in 2006. This document will introduce the reader to the mechanics and benefits of SIP Trunking and present an integrated solution from infrastructure provider Siemens Enterprise Communications and internet telephony service provider, Colt Technology Services.

Figure 2: Moving to IP allows the company to converge their voice and data networks. A specialised gateway device allows the IP PBX to connect to the PSTN for circuit switched calls.



3. What is SIP Trunking?

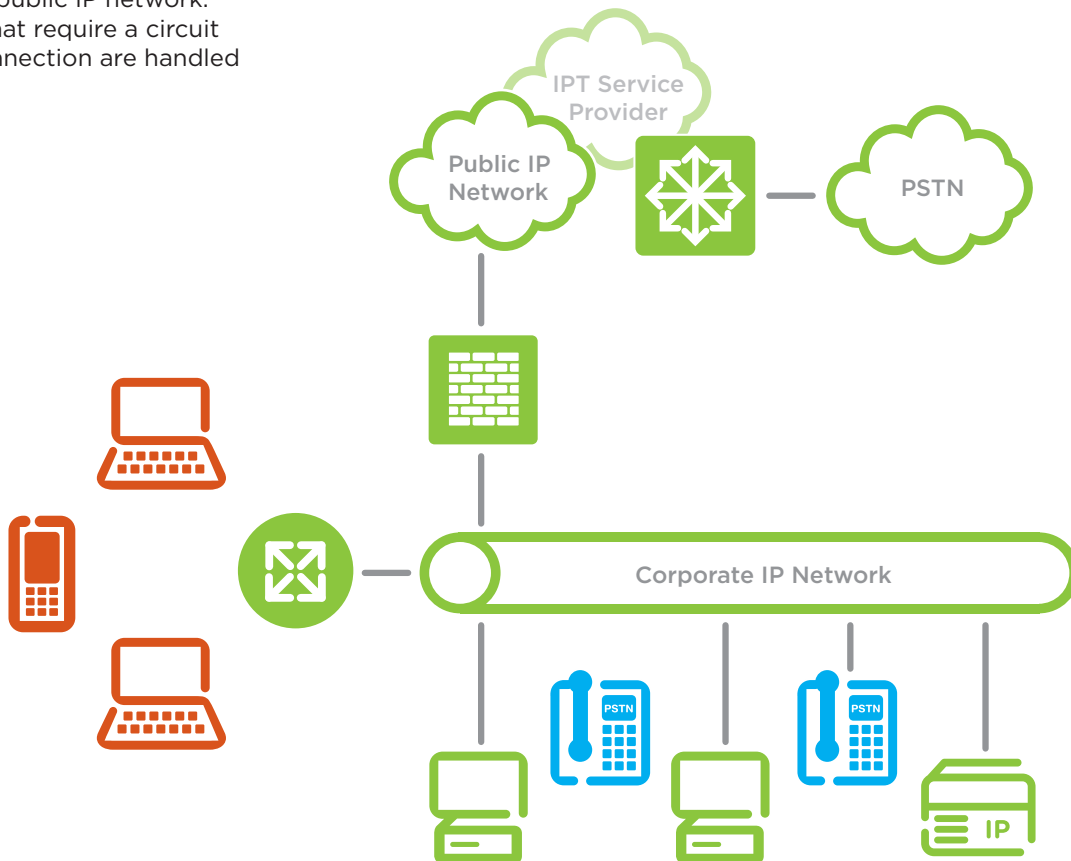
Back in the days of analog enterprise telephony, companies were required to purchase connectivity (lines) to their telecom carrier for voice communications. The number of 'lines' or 'trunks' required, was a function of the number of employees within the company and a complex formula involving call volumes and time of day. Companies had to plan for and purchase excess capacity to satisfy maximum utilisation, often resulting in underutilisation and overspending. Today's cost-conscious companies can replace traditional fixed PSTN lines with a SIP Trunking service - leveraging their existing broadband IP connection and creating a single pipeline for all enterprise voice and data. SIP Trunking allows the enterprise to control

voice call capacity by utilising different advanced encoding and compression techniques (codecs). The number of simultaneous calls is often determined by the desired call quality. For example, a lower quality/high compression codec such as G.729 codec demands less bandwidth per call and thus permits more simultaneous calls on the line. Whereas higher quality codecs such as G.711 or G.722 require more bandwidth, allowing fewer simultaneous calls. SIP Trunking works for both packet switched and circuit switched calls. If a call is placed to an external subscriber who does not use VoIP, the internet telephony service provider (eg Colt) handles the conversion to the PSTN. Calls between a company's global branch offices can be kept

on the IP network saving significant longdistance costs.

Service provider trunking is not new; it has been employed for many years by companies with TDM PBX's to reduce their connectivity cost. However, Trunking with TDM hardware was an expensive and inexact proposition. Each vendor had their own way of doing things and proprietary interfaces for endpoints and gateways. Much of VoIP's newfound popularity among small and medium-sized enterprises is due to the Session Initiation Protocol (SIP). SIP is an open standard that makes implementing VoIP far easier now than at any time in the past.

Figure 3: All voice and data connections are IP-based and traverse the public IP network. Voice calls that require a circuit switched connection are handled by the ITSP.



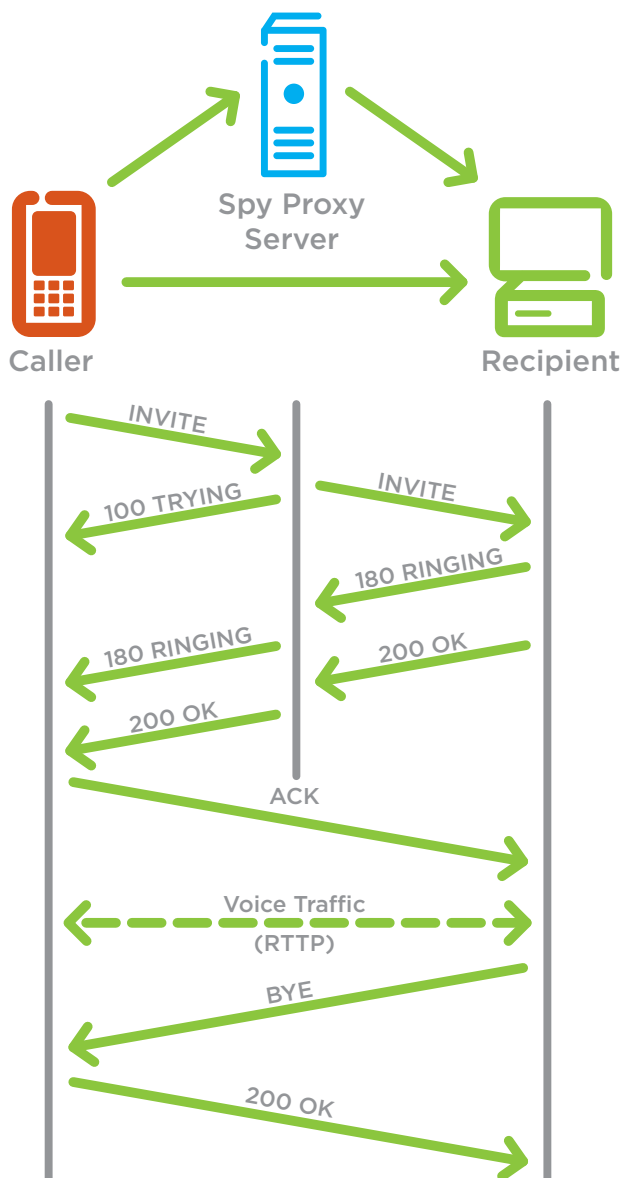
3.1 What is SIP?

SIP is the Internet Engineering Task Force (IETF) signaling protocol (RFC 3261) used for presence, messaging, VoIP, audio/video conferencing and events notification that is becoming for IP-based communications what HTTP is for the Web. In telephony, network engineers have always made a clear distinction between two different phases of a voice call. The first phase is 'call set up', and includes all of the details needed to get two telephones talking. Once a call has been set up, the phones enter a 'data transfer' phase of the call using an entirely different family of protocols to actually move the voice packets between the two endpoints.

In the new world of IP telephony and VoIP, SIP is the 'call set-up' protocol. It manages call set-up, routing, authentication and other feature messages between endpoints within an IP domain. Once the call is established, transport protocols such as Real-time Transport Protocol (RTP) actually send the voice data between phones.

SIP clients can setup and execute a real-time data transfer between each other, but it is more likely that SIP Proxy Servers are mediating the connection between the two client devices. Due to its power and simplicity, SIP has become the primary standard of internet telephony and continues to speed the adoption of VoIP in

the enterprise. New versions of SIP are already making their way onto the scene, promising to add business-class features such as caller ID, missed call support, and other useful features. Unfortunately, as is common with many open standards, basic SIP functionality is somewhat loosely defined within the specification. There is variation from vendor to vendor and even within a vendor from one equipment model and version to the next. This builds a strong case for certification and interoperability testing.



3.2 What kind of connection do businesses need?

Near-toll quality voice can be achieved using compression, using approximately half the bandwidth required compared to a traditional telephone line. This means that theoretically, even the slowest Digital Subscriber Line (DSL) connections are easily capable of supporting quality VoIP calls. However, it is important to note that inexpensive DSL lines are asymmetric (hence the term [Asymmetric DSL]), the maximum upstream bandwidth is often ten times less than the downstream bandwidth (DSL is usually from 512 Kbps to 1.544 Mbps downstream and near 128 Kbps upstream). Another consideration is that typically, the DSL line is not exclusively reserved for telephony and has to be shared with other IP traffic like email and internet downloads. For this reason, ADSL connections would only be recommended for the smallest of enterprises or an option for home

workers (ie so-called 'nomadic' workers) who can connect to their offices IP PBX allowing them to work at home the same as they would work in the office.

By contrast, larger companies with several employees are best served by a symmetric broadband connection to their ITSP to ensure reliable, high-quality voice calls and little or no performance degradation for downloading or other activities. Another important consideration when companies start using VoIP professionally, is that they need to negotiate a Service Level Agreement (SLA) with their service provider that guarantees uptime and a specific minimum bandwidth. In addition, when running a combined voice and data network, Quality of Service (QoS) network equipment should be utilised to give voice traffic priority over data traffic (much like emergency vehicles in urban rush-hour traffic) to ensure that the voice quality is optimised at all times.

3.3 IPT enables new applications

Integrating phones and IT systems on a single IP network offers a wide range of new applications and solutions. With IPT, mobile and office workers have anytime, anywhere access to their company's communication services. They are always reachable on the same number; call forwarding and separate office and home office numbers are no longer required. Even better, deploying centrally installed and maintained applications like unified messaging, every employee, mobile or stationary, can have a single multimedia mailbox for receiving voice messages, emails, faxes, and SMS text messages. Adding presence-based services, employees can automatically notify their contacts where and how they can best be reached. IPT also makes it possible to create telephony Application Programming Interfaces (API's) to allow companies to 'communications-enable' important enterprise applications and business processes. These new Communications Enabled Business Processes (CEBP) provide the worker with new ways to do their work and collaborate with co-workers.

4. Siemens Enterprise Interoperability with Colt Technology Services

Colt Technology Services and Siemens Enterprise Communications have worked together to ensure interoperability of the Siemens platforms with the SIP Trunking solutions from Colt Technology Services. Extensive interoperability tests have taken place with the following SME platforms¹:

- HiPath 3000 V7 and V8
- HiPath OpenOffice ME/
OpenScape Office MX

The Colt solution is based upon redundant Signaling Servers and 20+ gateways that cover all the European countries. This solution provides maximum redundancy, maximum availability and low latency in order to deliver high voice quality. Correct operations of the following features have been verified:

- Caller ID
- Hold, Consultation, Toggle
- Transfer, Call forwarding
- Call Pickup
- Conference
- DTMF transmission
- Fax according to T. 38

Administration of the SIP Trunks has been made particularly easy for customers of Colt. An explicit parameter is provided for Colt that automatically establishes all parameters needed for correct interoperability with Colt SIP Trunks. Further information about the testing methodology adopted and updated test results can be found at http://wiki.siemens-enterprise.com/index.php/Collaboration_with_VoIP_Providers. This site also includes the most up-to-date interoperability test reports for the latest versions of the platforms from Siemens Enterprise Communications.

General requirements for a successful VoIP deployment include:

- LAN with 10/100/1000 Mbps and no more than a 40 per cent network load
- Separate port on the switch or router for every component in the IP network (ie no hubs as concentrators)
- No more than a 50ms delays in one direction (one Way Delay) and no more than 150ms total delay
- No more than 3 per cent packet loss and 20ms of jitter

- Quality of Service (QOS) support – IEEE 802.p, DiffServ
- Sufficient WAN bandwidth (uplink and downlink) for the intended number of simultaneous calls for the chosen CODEC
- External modem, such as a DSL modem
- Intermediate routers must support quality of service and bandwidth control mechanisms if the WAN link is used simultaneously. SIP messages must be routed transparently.

¹Interoperability tests have also taken place with other platforms from Siemens Enterprise Communications including the HiPath 4000 V4 R4 as well as OpenScape Voice V3.1 and V4.1

5. SME Solutions from Siemens Enterprise Communications

5.1 HiPath OpenOffice ME/ OpenScope Office MX

OpenScope Office MX (formerly known as HiPath OpenOffice ME) is a solution for unified communications with integrated applications, including; Presence and Messaging functions and conference management. OpenScope Office MX is a pure IP system that is easy to operate and provides integrated services for mobility, voice and data for small and medium-size companies up to 150 users. It features built-in IP broadband connectivity, and supports SIP Trunking out of the box. Other features include:

- Unified communications applications
- Mobility options for mobile employees
- Services for voice and data
- Open interfaces for the integration of external applications

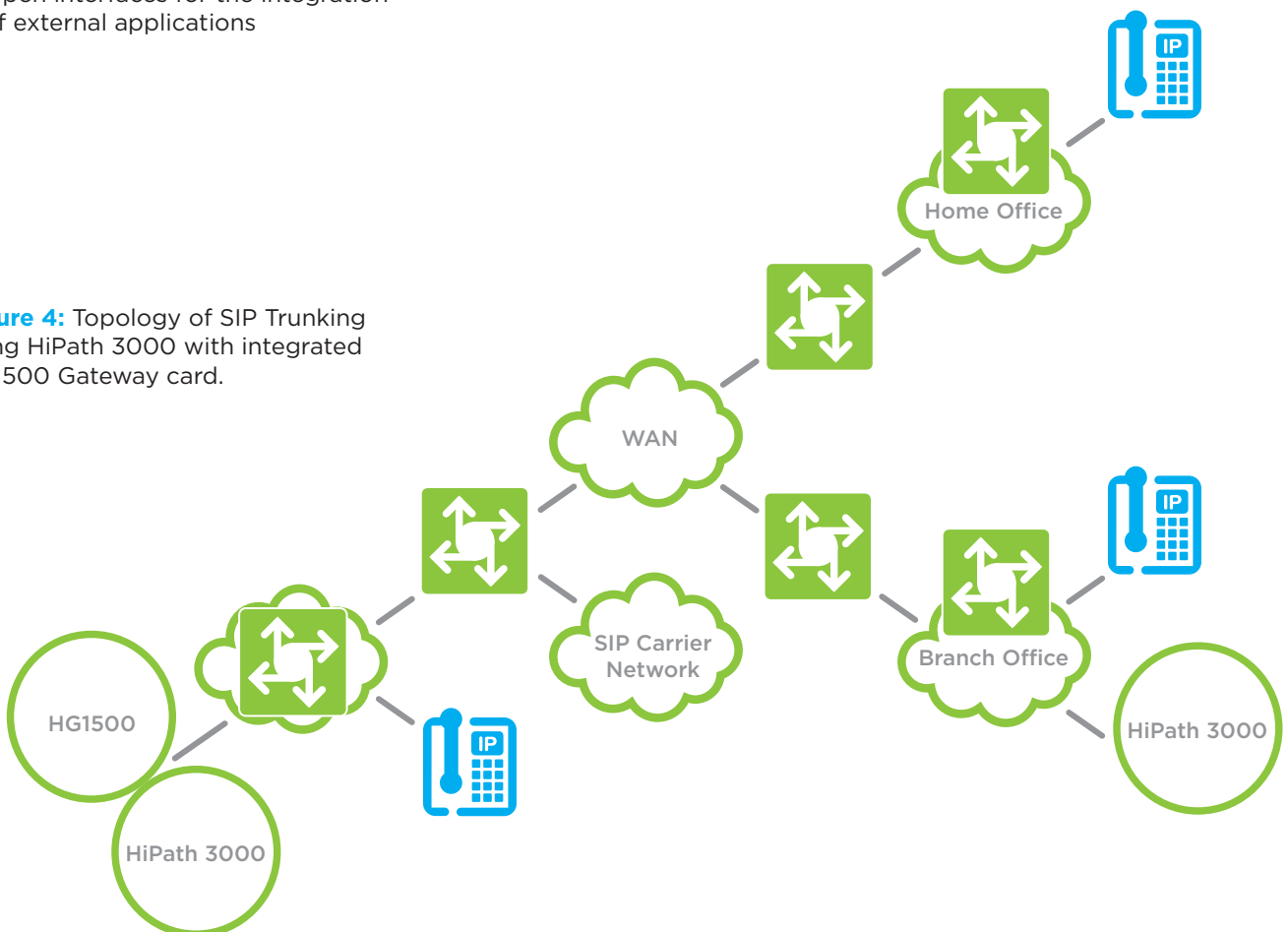
HiPath 3000 V8

The HiPath 3000 converged communications platform offers medium sized enterprises of up to 500 subscribers high reliability voice communications using high quality phones with easy operation. As a converged system, the HiPath 3000 supports both TDM and IP subscribers. Customers that are today using the HiPath 3000 to just connect TDM subscribers will find the path forward to IP can be done as an evolution and not as a revolution. Adding SIP Trunking support to the HiPath 3000 V8 is easy with the addition of the HG1500 Gateway card to the HiPath 3000. Best of all, the HG1500 Gateway card comes with a free unlimited SIP Trunking license as part of the standard package.

All of the HiPath SME platforms support the following SIP Trunking features:

- Direct connection to the DSL modem or operation behind another router using STUN
- Direct Inward Dialing (DID)
- SIP Trunking with single numbers (MSN) and up to 30 SIP user accounts
- Availability of the most important features like Call Forwarding and Call Transfer
- The number of simultaneous VOIP connections is only limited by the DSL bandwidth
- Voice Codes G.711, G. 729 and G. 723
- Automatic fallback to ISDN upon SIP trunk failure
- Fax over IP via T. 38 (or G. 711)
- Up to 4 active SIP providers and one internet connection.

Figure 4: Topology of SIP Trunking using HiPath 3000 with integrated HG1500 Gateway card.



5.2 SIP Trunking Solutions from Colt Technology Services: VoIP Access

Colt's main business focus is the promotion of an information delivery platform for enterprises that can enable easy access to business applications from anywhere, at any time. Their goal is to facilitate the transition of enterprise voice services to VoIP and to verify their interoperability with an increasing number of voice enabled business applications. To pursue these goals they have focused on a high quality of service, with high bandwidth and security that are essential for these services. Colt VoIP Access enables organisations to converge their voice and data traffic onto a single end-to-end IP network which allows them to minimise the infrastructure required to carry voice calls.

Colt VoIP Access is suitable for customers who own and/or operate an IP-PBX on their premises and want to exploit the advantages of IP Telephony via IP connectivity to the PSTN. In fact, Colt VoIP accommodates a wide variety of network architectures: the IP PBX can be located in the customer premises, placed in the partner premises or hosted in one of Colt's Data Centers. This allows partners and customers more flexibility with their existing IP PBX, simplifying integration and management.

VoIP is a sensitive technology and can still seem new for a number of partners and customers. Thus, Colt takes particular care for the delivery and monitoring of this solution.

In fact, Colt will run on-site testing for each VoIP link that is delivered in order to measure the setup of the service, the protocols, and network quality parameters. In addition to this, Colt VoIP offers come with highly rated service level engagements: up to 99.99 per cent target availability and fours on time to repair for fibre or even DSL connections. Also, Colt provides guarantees on the quality of service: maximum packet loss, round trip delays and jitter – ie, those technical parameters that measure the quality of an internet link – which are capped to values that are far below the thresholds that are commonly used for voice services. Colt VoIP Solution is based on two main offers:

Colt Smart Office

Reducing the number of suppliers and service providers is a key goal for the small enterprise. To make this possible, Colt has created a range of bundled products within the 'Colt Smart Office' portfolio, in order to provide an all in one telecom solution. This offer is simple to order, widely available, and modular enough to meet the needs of most any small enterprise. The package includes:

- high-speed internet access (from 1 Mbps to 10 Mbps);
- a voice solution for TDM as well as for IP PBXs that benefits from end to end quality of service, and;
- additional add-on features such as Managed Dedicated Firewall, Managed Email and PC Backup.

Colt VoIP Access

Among medium and large enterprises, VoIP is one key solution that can homogenise telephony services across sites, reduce telecommunication costs, and drastically simplify the integration with new business applications. For those companies, Colt VoIP Access is the solution. VoIP Access enables customers to migrate legacy TDM access services to an IP environment, presenting significant opportunities for consolidation and cost savings. It accommodates any architecture, is available at any bandwidth and can connect to any existing IP VPN or internet-based infrastructure. VoIP Access can also be provided via a dedicated, private IP network including access lines and a managed router. The product includes options for multiple redundancy mechanisms, and handles emergency calls even for companies distributed across multiple countries, and a heterogeneous telephony infrastructure that can include tele-workers or nomadic usage.

6. Benefits of SIP Trunking

Once a company has made the investment in IP Telephony, they are motivated to find new ways to leverage that initial investment for additional cost reductions and service enhancements. SIP Trunking can provide many additional benefits for these organisations, including:

- eliminating the need for Primary Rate Interface (PRI)/Basic Rate Interface (BRI) connections, and lowering telephony costs
- maximum flexibility for new provisioning lines and avoids having to purchase capacity of an entire T1 or E1 at a time. Companies can also avoid paying a premium for the flexibility of fractional T1's
- the reduction of monthly telecom expenses, since only one connection for both data and voice is required
- allowing organisations to outsource their PSTN connectivity to a third party with presence in multiple locations, thus reducing long-distance charges and allowing organisations to establish local numbers
- eliminate the need to invest in costly (and less capable) TDM-gateway infrastructure
- accessing the benefits of a hosted VoIP service, without discarding existing investments in a traditional phone system.

Quantitative calculations of ROI can be more difficult as the return depends on the size and scope of the deployment. According to a May 2009 report from Gartner entitled 'SIP Trunking: Where Are the Savings?', SIP Trunking can deliver a 30 per cent to 50 per cent reduction in voice and data connectivity costs for branch environments. Additional savings are possible from reducing LD charges and increasing the use of VoIP.

7. Conclusion

IP Telephony in the enterprise has reached critical mass – there are more pure IP and Hybrid PBX's in the marketplace than there are legacy TDM systems. According to a report from Infonetics, by end of 2010 almost half of SME's and two-thirds of large enterprises in North America will be using VoIP products and services. The prevalence of IPT justifies the growth of SIP Trunking globally. Companies that have made the initial investment in IPT are looking for additional ways to leverage their investment and further reduce telecom costs. SIP Trunking is inexpensive to deploy and can often pay for itself within a year. The technology can also open up exciting new aspects of telecommunications like company-wide UC and mobile VOIP. Together, Siemens Enterprise Communications and Colt Technology Services have extensively tested SIP Trunking between Siemens' enterprise telephony solutions and the Colt Technology Services SIP Trunking service. There is a fully interoperability tested solution for enterprises of all sizes and complexities.

Siemens Enterprise Communications is a thought leader and innovator in the enterprise communications industry. We are one of the leading players in the market with full coverage of all the relevant markets from a strong European base with global reach. Our people have the passion, commitment, skills and know-how to deliver a broad range of cutting-edge technologies, outstanding products and professional services. All with the support of an enterprise that has the financial strength to outperform the rest in this competitive and consolidating market.

Colt Technology Services is Europe's information delivery platform, enabling its customers to share process and store their vital business information. Colt provides major organisations, midsize businesses and wholesale customers with a powerful resource that combines network and IT infrastructure with expertise in IT managed services, networking and communication solutions. Colt operates a 13-country, 25,000km network that includes metropolitan area networks in 34 major European cities with direct fibre connections into 17,000 buildings and 19 Colt data centres.

About Siemens Enterprise Communications

Siemens Enterprise Communications is a premier provider of end-to-end enterprise communications solutions that use open, standards-based architectures to unify communications and business applications for a seamless collaboration experience.

This award-winning "Open Communications" approach enables organisations to improve productivity and reduce costs through easy-to-deploy solutions that work within existing IT environments, delivering operational efficiencies. It is the foundation for the company's OpenPath commitment that enables customers to mitigate risk and cost-effectively adopt unified communications. This promise is underwritten through our OpenScale service portfolio, which includes international, managed and outsource capability. Siemens Enterprise Communications is owned by a joint venture of The Gores Group and Siemens AG. The joint venture also encompasses Enterasys Networks, which provides network infrastructure and security systems, delivering a perfect basis for joint communications solutions.

For more information about Siemens Enterprise Communications or Enterasys, please visit www.siemens-enterprise.com/open or www.enterasys.com

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About Colt

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For more information about Colt services for midsize business please visit www.colt.net/business or email inbound@colt.net