SIP Trunking:
IP Telephony for the Enterprise

SIP-powered telephony is paving the way to the new world of Unified Communications. For traditional PBX systems as well as state-of-the-art all-IP telephony implementations, SIP trunking connects today's enterprises to the cost-savings, streamlined operations and rich-featured communication that voice-over-IP offers.

PATTON
Electronics Co.

Network Access
&Connectivity

Copyright

Printed in the USA.

Contents

Introduction

SIP—the acronym stands for session initiation protocol—has emerged as the universal standard for Voice-over-Internet Protocol (VoIP) services. As with its predecessors (HTTP for web browsing or SMTP for email), SIP enables users to engage in integrated communications anywhere and anytime—regardless of the underlying network infrastructures employed. The ubiquitous availability of Internet access has ushered in the era of SIP-based trunking services, which enable enterprises to migrate from dual, physically separate voice and data networks to a converged and consolidated all-IP infrastructure.

The migration to SIP-powered IP telephony has taken the world's businesses by storm as it paves the way toward the new world of Unified Communications. Have you joined the crowd? If you have been contemplating moving towards IP- based telephony supported by such services as SIP trunking, perhaps now is the ideal time to make the leap.

What is SIP Trunking?

First off, what do we mean by trunking? With traditional a PBX, the enterprise leases a high-capacity, multi-call phone line—a trunk—that connects the internal phone system to the outside world via the public switched telephone network (PSTN). A traditional trunk is a service offered by the plain old telephone service (POTS) carrier-provider and is based on time division multiplexed (TDM) technology.

A SIP trunk, in contrast, uses Internet protocol (IP) technology to provide the connection to the outside world. So, SIP trunking is a service offered by an Internet telephony service provider (ITSP) that connects a private business-class phone system—such as a hardware-based private branch exchange (PBX), key system, or software-based iPBX—with a

voice-over-IP service network. The SIP trunking service allows the enterprise to realize the cost-saving, technological and operational benefits of VoIP when communicating with branch offices as well the outside world. SIP trunking can be implemented either with or without replacing the organization's existing voice network equipment.

SIP Trunking: Benefits for the Enterprise

For the Enterprise, SIP trunking provides a host of benefits, including the following:

1. **Lowered Total Cost of Ownership**
   a. **Reduced Long distance call charges**
   With globalization, multinational companies have offices all around the world. Consequently, a significant amount of calling traffic exists between these offices as employees collaborate on projects together. This leads to high long distance call charges. Even though International calling rates have significantly come down in the recent years, they are still a significant source of expenses for multinational companies. With SIP trunking and least cost routing, these costs are significantly reduced. In addition, SIP trunking opens up new avenues for conferencing and collaboration

   b. **Reduced Capital Expenditure** (CapEx)
   Moving onto a single platform that handles both voice and data traffic reduces capital expenditure by a huge margin, both initially and for the long term. SIP trunking allows the enterprise to consolidate separate voice and data infrastructures into a single converged network. In addition, by implementing IP-based trunking services, branch offices can preserve their investments in legacy infrastructure and extend its useful life, while leveraging the IP-based services hosted at headquarters.
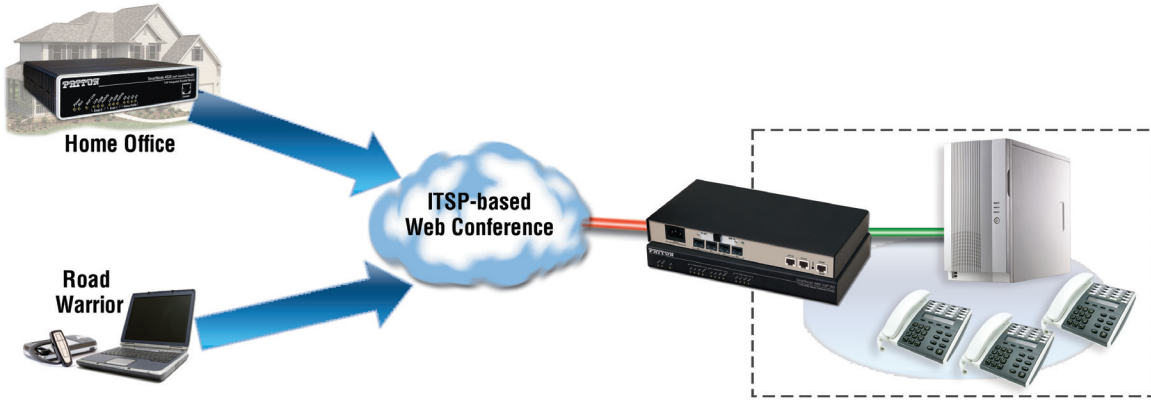
c. Lower Operational Expenditures (OpEx)
SIP based trunking allows one converged network to transport data and voice services simultaneously. This convergence reduces the overhead required to operate and maintain two disparate systems. Moves, adds, and changes (MACs) are much easier and cheaper on an IP-based PBX and telephony platform. With legacy platforms, difficult configuration changes can be cumbersome and costly third-party consulting often comes into picture.

2. Enhanced Employee Productivity
  a. Improved collaborative and conferencing tools
Initially, enterprises began adopting VoIP technology in SIP trunking applications as means of reducing long distance bills. However, these technologies have evolved into a platform that enables such rich featured applications as Unified Communications, conferencing, and worker collaboration. With teams working across the globe, SIP enables multimedia conferencing, which can be easily setup and operated independently from the service provider. In addition, SIP presence features allow users to be notified when the other party is online, ready and available for interaction. SIP also supports such file-sharing and business applications as Microsoft Office Suite on a single platform, enabling still higher levels of collaboration.

b. Reduced mobile phone bills
With to remote voice connectivity via SIP, road warriors no longer need to mount their monthly expenses by using their mobile phones when they are globetrotting. With Internet access universally available, road warriors can use SIP to be connected to all their phone numbers remotely. Furthermore, by using presence, SIP allows a single number to be

Customer Support Center SIP IP CRM Integration One-Touch Web-based Click to Call
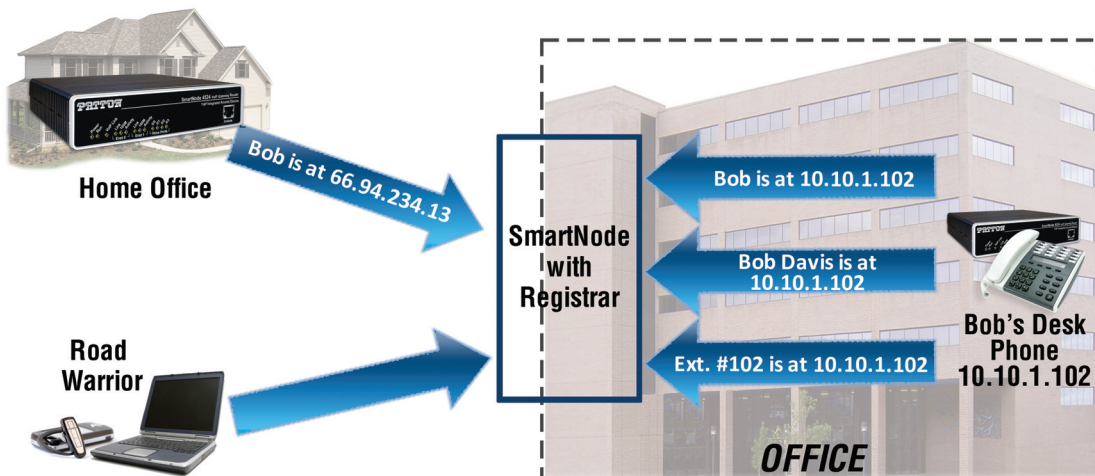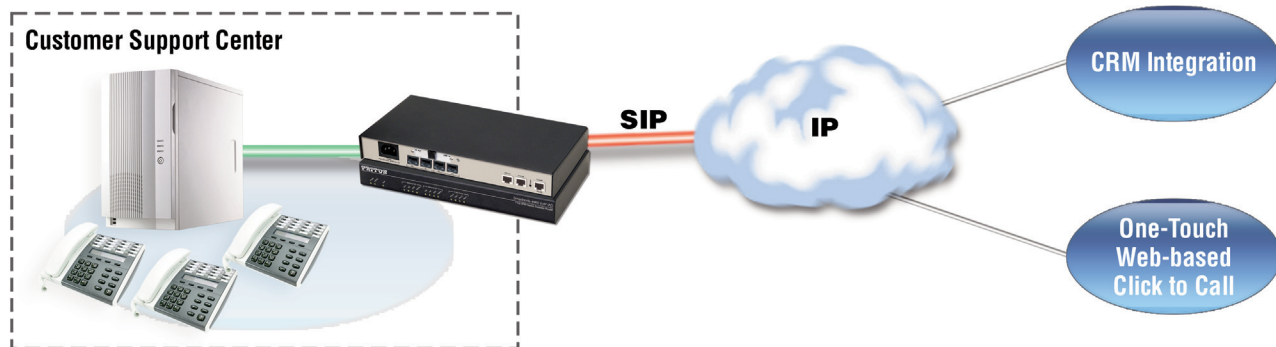
registered to a variety of devices, making staff members available to their customers even when they are travelling.

3. Enable higher customer satisfaction with improved business processes
**a. Integration with CRM and ERP system**
SIP is a text-based protocol, developed by the Internet engineering task force (IETF), the same body that was responsible for laying the foundation for the Internet. As a result, SIP is highly deployable in a variety of scenarios. When integrated with Customer Relationship Management (CRM) software, SIP can immediately offer the option to chat with a customer support rep when a customer logs in to enter a complaint. If the problem is severe, the rep can be reached via one click as the system sets up a call between the customer's registered number and the rep's number. Such value-added service significantly increases customer satisfaction and the perceived value of the offering a customer purchases.

b. Web-based Click to Call
Because SIP is highly deployable, it can be integrated with the corporate website, employee profiles, and other customer-facing platforms. This is especially valuable for sales personnel and other customer-facing functions who cannot afford to miss out on any opportunities to interact with a potential client.

4. Redundancy and Failover
The legacy PSTN was highly resilient and offered excellent up time for the end customer. However, an outage with the service provider would leave end customers without service for the duration of the outage. With IP-based telephony, an enterprise can connect to a very large number of service providers, providing service redundancy. This capability protects against provider network failures to assure a high level of service availability. The enterprise can realize further cost savings when such advanced features as least cost routing (LCR) are implemented. The VoIP system can

Customer Support Center SIP ITSP 1 ITSP 2
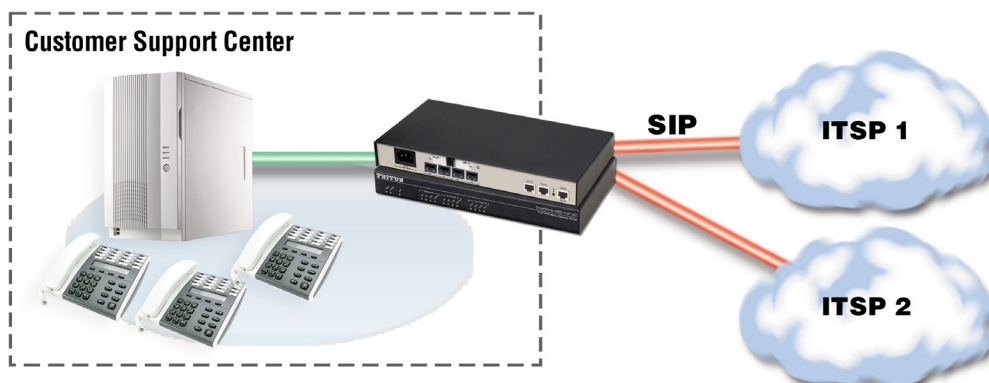
optimize calling rates by automatically selecting the least expensive service provider for each called number.

SIP Trunking: Benefits for the Service Provider

We have discussed some of the ways that SIP trunking is highly beneficial for the enterprises. However, the technology brings a host of benefits for the service provider as well. Providing SIP-based trunks to customers the service provider reaps a host of benefits, including the following:

1. **Single Conduit for both Data and Voice**
   Today, most enterprises already subscribe to IP-based data backbones. By transporting voice traffic across the same conduit, service providers can reduce the total cost of deploying and delivering both services. In addition, the consolidated infrastructure offers the service provider centralized management for all the services they provide.

2. **Reduced Total Cost of Ownership**
   a. Reduced CapEx
   Having two separate networks—one for voice and one for data—significantly increases the total cost of ownership for service providers. In addition, if two networks have to be deployed instead of one, expansion into new markets is not only costlier but operationally much more difficult. By allowing a single network to transport voice and data traffic, SIP trunking reduces capital expenditure for service deployment.

   b. Reduced OpEx
   Transporting both data and voice over a single IP network is not only cheaper, it is also easier to manage and maintain. In addition, both traffic types can be managed from a single system, cutting the necessary network management resources in half.

   c. Moves, Additions and Changes (MAC)
   In today's competitive space, the way MACs are handled is highly critical for controlling the bottom line. Obviously revenues are affected. But the ease and seamlessness of MAC administration--along with the cost of managing the services before and after a change--bear upon monthly operating expenses. With legacy services, MACs often require physical reconfiguration that disrupts services. By controlling such modifications in software, IP telephony greatly simplifies MACs. To modify the number of enterprise users the trunk supports (a complex procedure in a legacy system) becomes a simple software reconfiguration to increase the bandwidth allocated to the trunk.

3. **Offer Differentiated Services**
   Service providers are always looking for services which would generate additional revenue as and differentiate them from their competitors. Offering SIP-based trunking allows Service providers to make expanded inroads into the enterprise market space by offering premium collaborative and conferencing tools in their cloud-based services.

SIP Trunking Implementation

There are two basic approaches to implementing SIP trunking, depending on the type of PBX solution to be used: traditional hardware PBX or IP-based software PBX. Let's summarize the pros and cons of each approach.

SIP Trunking with a Legacy PBX
When an enterprise has invested and deployed a traditional PBX system (which is typically SIP un-aware) with legacy handsets, there is little incentive to rip it all out and replace it with a completely SIP-based solution.
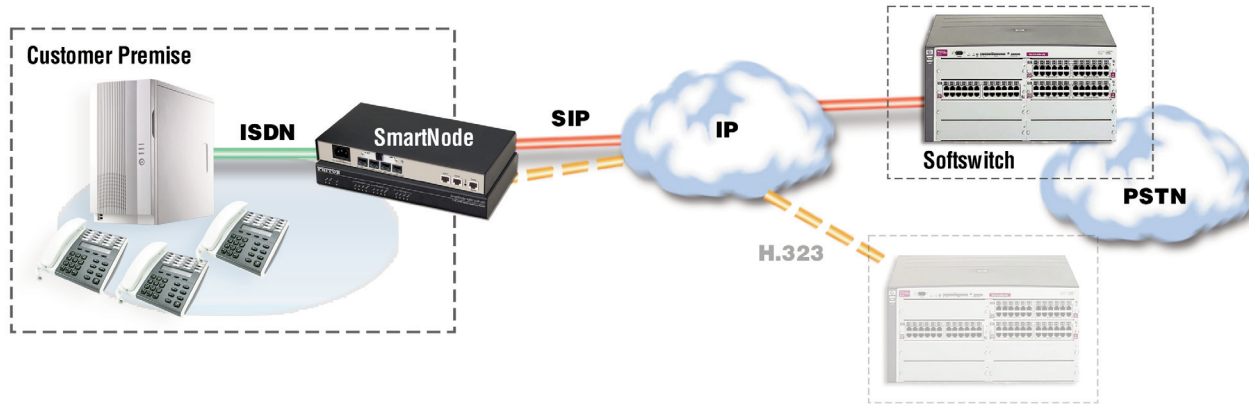
However, to realize the cost savings of IP telephony, a VoIP gateway is required to convert the PBX interfaces to Ethernet while packetizing the voice traffic for transport over the IP-based trunk to the SIP service provider network. Typically, a multiport gateway is deployed between the legacy system and the service provider (see figure above).

Advantages:

    a. Lower cost voice communications with intra-office toll-bypass

    b. Investment in existing telephony equipment is preserved

    c. No new voice hardware expense is required

    d. The end customer experience remains unchanged

    e. No retraining of staff on new processes

Disadvantages

    a. Maintaining the old PBX may be expensive and cumbersome

    b. Rich new features that all-IP telephony enables remain unavailable

SIP Trunking with an IP-PBX

For new offices where infrastructure is just being setup this might be the perfect way to go. The scenario involves providing a SIP trunk for IP devices that are already SIP enabled. These devices could typically interact with the trunk directly. However, to provide certain advanced features—such as call admission control, billing, QoS, security, and others—intermediate devices must be deployed. These may include an IP PBX and a session border controller.

Advantages:

    a. Advanced features like call parking, voice enabled IVR, integration with other business subsystems can be enabled

    b. The cost of maintaining an IP PBX is much less than that of maintaining legacy PBXs

Disadvantages

    a. Must invest in completely new hardware

    b. End users must go through a learning process to master the newer features
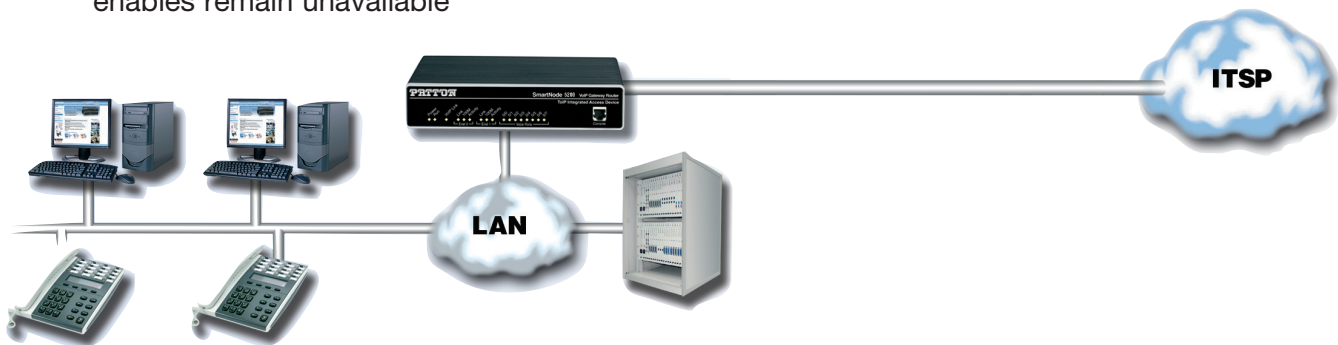
Difficulties in Implementing SIP Trunking

SIP-based telephony supports rich new features that produce substantial cost savings and enable time-saving, productivity-boosting applications for enterprises and service providers. However, the technology is still evolving and is not yet fully standardized. As such, SIP deployments may face certain difficulties faced during implementation.

Interoperability

In order for a SIP trunking installation to work, all the network elements in the enterprise network must interoperate together with all the elements in the service provider network. Handsets, soft-phone clients, hard and/or soft PBXs, VoIP gateways, firewalls, SIP servers, SS7 switches, and more. With so many network elements in play, there's a lot that can go wrong. All it takes to spoil the party is one guest that won't play nicely with the others.

Selecting a VoIP gateway that offers a solid track record of interoperability with major-brand iPBXs and soft switches, goes a long way toward ensuring the deployment actually works. By providing interface and protocol conversion with consistent implementation of open standards, a highly interoperable VoIP gateway can mediate interoperability issues between enterprise and provider elements.
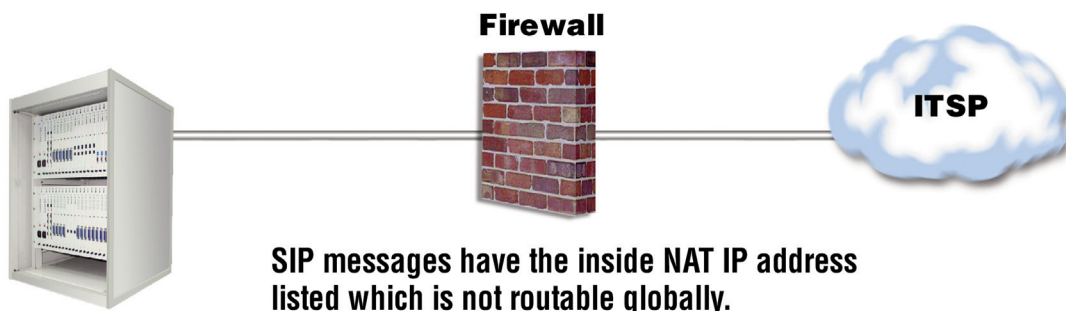
Information Security

Since SIP trunking is an IP-based service, it is subject to the same concerns that plague IP data traffic. These include eavesdropping, denial of service (DOS) attacks etc.
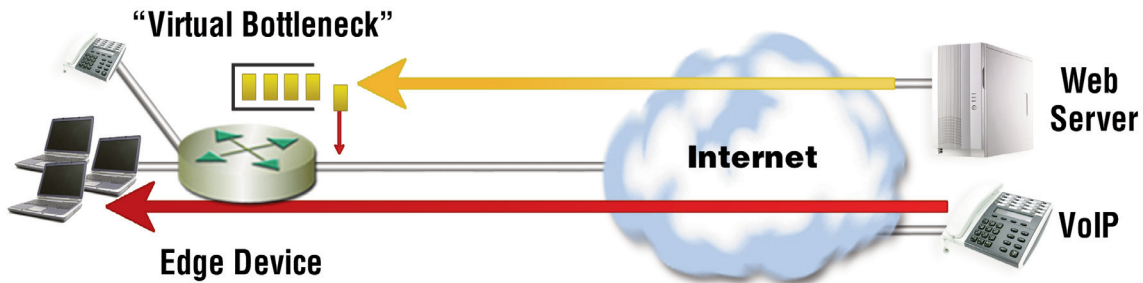
When considering information security, it is important not only to prevent attacks, the security solution must also to provide certain related functions, such as:

a. Hiding the internal network topology from the outside world.

b. Preventing malicious traffic from stealing valuable bandwidth and network resources.

c. Protecting against interception and eavesdropping, without introducing latency, jitter, or delay.

Firewalls typically provide security by employing network address translation (NAT) to hide the internal private network from external users and devices connected to the public network. However there is an inherent problem in the way a typical firewall functions.

The security and NAT functions typically operate at the network and transport layers of the OSI model. Because the SIP protocol works at the application layer, if the firewall is not SIP aware, it will block SIP signaling messages. So SIP won't work with the firewall. This impediment creates a huge conundrum for enterprises: whether to implement a secure firewall that breaks the SIP service or implement SIP trunking with-

Firewall

ITSP

SIP messages have the inside NAT IP address listed which is not routable globally.

"Virtual Bottleneck"

Edge Device

Internet

Web Server

VoIP

Encryption. Most VoIP gateways compromise information security by transporting VoIP and data traffic over public networks without encryption. For organizations that handle sensitive data—notably banks and financial institutions, as well as military and certain government agencies—weak security has made VoIP deployment a non-starter. For many businesses the VoIP system must offer a mechanism for providing voice encryption that prevents snooping, even if the traffic is intercepted as it traverses the Internet. All data, voice and VoIP must be kept from prying eyes and ears while ensuring that all communications came from the trusted source.

Quality of Service

While the Internet is a best-effort service, PSTN services provide guaranteed delivery of traffic. So a major concern for many SIP-trunking adopters is whether SIP trunking can deliver voice quality comparable with that offered by the legacy PSTN. It is important to have a mechanism that marks voice-over-IP traffic so that voice calls can be differentiated from data traffic. Thus marked, the Service Provider network can give VoIP traffic priority treatment that addresses delay, jitter and packet loss considerations in accordance with the requirements of telephony.

In addition, to avoid overbooking that degrades call quality there must be a method which supervises the number of concurrent calls traversing the network at any given time.

Finally, the edge device on the customer premise should provide mechanisms (as shown in the figure below) that throttle downstream surges in data traffic to ensure adequate bandwidth is always available to support clear, high-quality voice.

Enterprise Class Features

As Enterprises embrace SIP trunking as an IP enabled service, they will expect the same enterprise-class features they already receive from other IP-enabled services. Such features include, support for Virtual Private Networks (VPNs), voice encryption, high availability, full-featured IP routing with IP link redundancy, PSTN fallback for voice and data traffic, system survivability for business continuity.

LAN Segment

ITSP
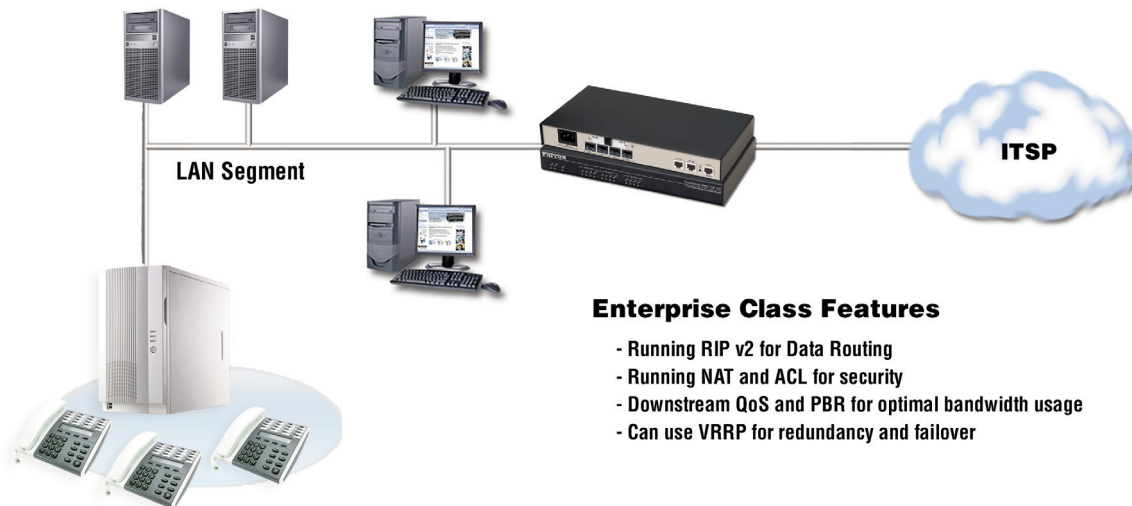
Enterprise Class Features

- Running RIP v2 for Data Routing
- Running NAT and ACL for security
- Downstream QoS and PBR for optimal bandwidth usage
- Can use VRRP for redundancy and failover

VoIP Gateway Router Solutions for SIP Trunking

The ideal solution for a given enterprise will depend on the type of telephony system currently installed. We now present two approaches for SIP trunking.

1. Legacy Network Trunking with the SmartNode 4960 PRI VoIP IAD

In the case where a traditional TDM-based PBX has been deployed, it may be more cost-effective to IP-enable the existing phone system for toll-bypass via the Internet. Patton offers a complete line of SmartNode VoIP gateways and routers (see www.patton.com/voip) that can provide IP connectivity for legacy telephony systems in analog POTS and digital ISDN environments. The specific model selection will depend on the scale of the enterprise, types of interfaces employed, and other application requirements.

For a mid-sized enterprise requiring up to 4 PRIs, the SmartNode™ 4960 T1/E1/PRI VOIP IAD extends the useful life of an installed telephony system, and preserves the enterprise's investment in legacy voice equipment. The SN4960 enables SIP trunking by providing one or four T1 or E1 ports as well as two gigabit Ethernet ports. In addition, the SN4960 enables such enterprise class features as IPsec and Downstream QoS while providing complete and powerful IP routing. Offering industry-leading interoperability and reliability, the SN4960 solution future proofs the company network and sets the stage for migration to all-IP communications further down the road.

2. IP trunking with the SmartNode™ 5200 Enterprise Session Border Controller (SN5200 eSBC)

For some organizations, the legacy PBX cannot be IP-enabled, and must be replaced with an IP-based PBX. In many organizations, especially a new enterprise or new branch office, it makes sense to implement an entirely IP based telephony solution using software-based iPBX.

For such all-IP enterprise communications systems, the SmartNode™ 5200 Enterprise Session Border Router is the ideal VoIP gateway solution. The SmartNode eSBC is used to connect the internal IP-telephony network to the SIP-based Service Provider network while providing additional security and quality of service features at the edge of the enterprise network.

IP Trunking Solution Scenario

Let's take a look at a scenario where SmartNode eSBC is used to solve the problems associated with SIP trunking. (For a legacy telephony deployment, the SmartNode SN4960 would be deployed in a similar manner to solve these problems).

Security

For SIP trunking to work, the SIP traffic must be able to pass the enterprise edge, which provides such functions as firewall, NAT, and access control lists (ACLs) deployed for security purposes. Most firewalls are not SIP aware and so the Enterprise must choose between deploying a firewall that breaks the SIP system and implementing SIP trunking without the security of a firewall. By way of a solution, the SmartNode eSBC offers two alternatives.

1. **Built-in SmartNode Firewall.** On-board security functions built-into the eSBC include ACL support and policy-based routing which together provide a virtual built-in firewall. This alternative not only secures the enterprise against malicious traffic and denial of service attacks, it eliminates the
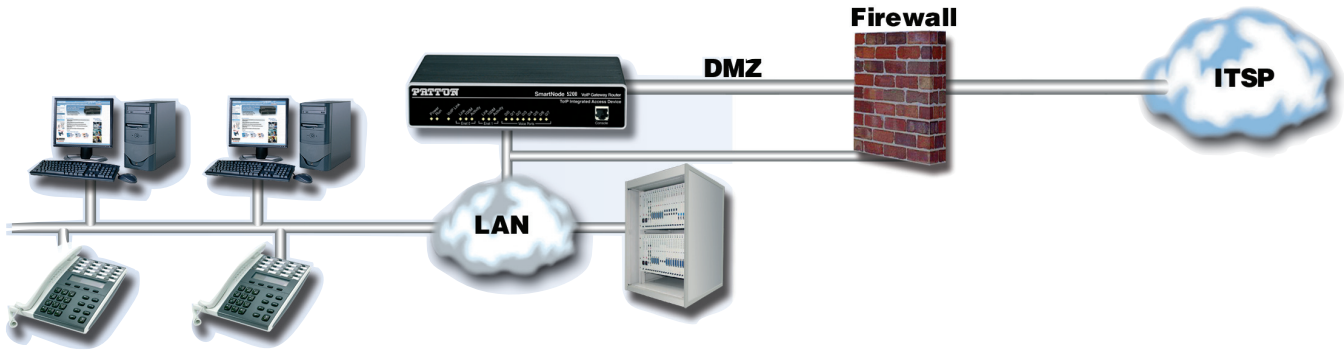
2. **SIP-enabled external firewall.** In this approach, an external firewall (which is not SIP aware) is configured to pass SIP signaling messages to and from the SmartNode™ 5200, which handles all the SIP processing. By SIP enabling the firewall which is installed at the enterprise edge, the SN5200 allows enterprises to benefit from the universal SIP trunking services offered by the ITSP.

Voice Encryption

Both the SN4900 and SN5200 employ Patton's VoIP-over-VPN technology with encrypted voice that delivers secure VoIP with IPsec strong encryption, AES and 3DES keying, and Internet Key Exchange (IKE). IPsec is the standard for securing communications over the Internet, while DES and AES offer strong encryption. The SmartNode IPsec creates private VPN tunnels ensuring secure VoIP, voice and data traffic while protecting all inter-office voice and data communications over the Internet. The tunnels allow users, located in separate offices, to communicate as if they were connected by a single private network. Now all data, voice and VoIP will be kept from prying eyes and ears while ensuring that all communications came from the trusted source.

For more information about voice encryption using VoIP-over-VPN technology, see the Patton white paper: *Securing Internet Telephony: Encrypting Voice with VoIP-over-VPN*.

Interoperability

The SmartNode™ VoIP platform is backed by Patton's interoperability certification program. Patton has certified and extensive and ever-growing list of major, name-brand softswitch vendors and IP-PBX suppliers for interoperability with SmartNode VoIP products. Patton's interoperability program extends beyond the lab, certifying Softswitch vendors when interoperability is both lab tested and demonstrated in live deployments in carrier and service provider networks.

Quality of Service

For SIP trunking to replace traditional telephony, it will require quality of service at the same level that traditional PSTN provides. However with TCP traffic being so prevalent in the public network, this level of quality becomes difficult to achieve. When used as a session border controller, the SN5200 introduces a dynamic virtual bottleneck at the customer premises that begins discarding non-real-time traffic (which can be retransmitted and recovered by end-user applications) before it starts to block the voice traffic in the edge router.

Though not all vendors offer it, SmartNode has always included upstream QoS as a standard feature. Since SmartWare™ 3.10 (released 2004), SmartNode has provided Patton's unique DownStreamQoS™. The feature gives incoming voice traffic priority over data, so you always hear the other speaker clearly. When data floods the downstream line, DownStreamQoS™ can throttle incoming data to ensure time-sensitive voice traffic gets through promptly. Patton SmartWare for SmartNode includes

DownStreamQoS and upstream QoS to deliver consistently clear voice in both directions.

For technical details on all the QoS mechanisms built into Patton SmartNode™, see the white paper *VoIP: Addressing QoS Beyond the Provider Network.*

As a further safeguard for high voice quality, the traffic profiles and calling profiles built into SmartWare provide call admission control. By ensuring only valid calls are permitted to use system resources, the voice quality of legitimate calls is protected.

In summary, the SmartNode 5200 enables the following benefits:

1. Hide the internal network topology from the outside world. With NAT and ACL configured and operating, the SmartNode controls all voice traffic while the firewall enables and controls the data traffic. This configuration allows the external firewall to hide the internal topology of the network from the outside world.

2. The firewall provides security functions while Patton's DownstreamQoS™ prevents the introduction of delay, jitter, and latency.

3. Using a combination of policy based routing and access control lists, The SN5200 prevents malicious traffic from using up valuable bandwidth and resources.

4. The SN5200 employs Patton's voice-encrypted VoIP VPN technology to deliver secure VoIP with IPsec strong encryption, AES and 3DES keying, and Internet Key Exchange (IKE).

Conclusion

For both traditional PBX systems and state-of-the-art all-IP telephony systems, SIP trunking connects today's enterprises to the cost-savings, streamlined operations and rich-featured communication that voice-over-IP offers. The highly interoperable SmartNode 4960 and SmartNode 5200 series of VoIP Routers, provide critical interface connectivity, interface and protocol conversion, and powerful, feature-rich software functions to enable secure, reliable SIP trunking services—not only for today, but well into the future.

Enterprise Class Features built into the SmartNode 5200

The SN5200 goes well beyond enabling SIP trunking for the enterprise. The eSBC provides the following high-end enterprise-class features:

- VPN—Standard IPsec VPN's provide voice encryption with IPsec AES/DES strong encryption to enable secure communications

- VRRP for high availability, redundancy and failoverW

PATTON
VoIP Product Selection Guide

	Family	Gateways	Gateway/Routers	IADs
		with 1 Ethernet Port	with 2 or More Ethernet Ports	with Integrated WAN Dual Ethernet + Modem
Analog	SIP ATAs (1–2 channels)	M-ATA	SL4020	
	Multiport FXS/FXO (4–8 channels)	SN4110	SN4520 2290	SN4830/**AA** SN4830/**C** SN4830/**D** SN4830/**G** SN4830/**T**
	IP ChannelBank (12–32 channels)		SN4900 SN4400	SN4900/**C** SN4900/**D** SN4900/**G** SN4900/**K** SN4900/**T**
Digital	SOHO ISDN (2 channels)	S-DTA	SN4522 (5 Eth Ports) SN4562 (5 Eth Ports)	
	Multiport ISDN (4–8 channels)	SN4554	SN4630 High Precision	SN4650/**AA** SN4650/**AB** SN4650/**C** SN4650/**D** SN4650/**G** SN4650/**K** SN4650/**T**
	T1/E1/PRI Trunking (15–120 channels)		SN4960/61 High Precision	SN4960/**G**
eSBC	32 SIP Calls		SN5200	
	64 Fax & Codec Transcoding Calls			SN5400/**G**

Model Code Key
AA: ADSL–Annex A **D**: X.21 **T**: T1 Data
AB: ADSL–Annex B **K**: E1 Data **G**: G.SHDSL
C: V.35