Ingate SIP Trunking

For enterprises wanting to make full use of their installed IP-PBXs and not only communicate over IP within the enterprise, but also outside the LAN, the Ingate SIP Trunking software module offers an easy and smooth transition to a modern and future-proofed SIP trunk solution. Using Ingate's advanced routing technology, the SIP Trunking module provide seamless connectivity between an enterprise IP-PBX and a SIP trunk.

SIP trunking is a service provided by Internet Telephony Service Providers (ITSPs) to connect to the traditional PSTN network. It permits businesses to adopt Voice-over-IP (VoIP) using the same connection as the Internet connection and remain in touch with others who rely on the PSTN. This can be the case since the enterprise IP-PBX is connected to the service provider's PSTN gateways over the Internet.

There are a number of benefits to utilizing a SIP trunk. For instance, SIP trunking offers significant cost-savings for enterprises, eliminating the need for local PSTN gateways, costly ISDN BRIs (Basic Rate Interfaces) or PRIs (Primary Rate Interfaces). SIP trunking also provides a more flexible solution for growing enterprises.

Ingate SIP Trunking has proven interoperability with several leading IP-PBX vendors. It is designed to be the demarcation point of the enterprise network and to work seamlessly in conjunction with IP-PBXs installed on the enterprise LAN to allow for the IP-PBX to be connected to a SIP trunk.

Ingate SIP Trunking can be used with Ingate Firewall® and Ingate SIParator® products to provide the advanced routing capabilities necessary for enterprises to connect to SIP trunks, while employing Ingate's proxy-based traversal and security solutions to allow the SIP-based communication to reach intended users.

Advanced Routing Capabilities

Ingate SIP Trunking provides advanced routing capabilities that enable enterprises to connect to SIP trunks. Frequently, the SIP traffic coming from the IP-PBX to be routed to the ITSP (and vice versa) is neither written in the format that the other expects nor contains the correct routing information to get it to its intended destination. Ingate's SIP Trunking software module can overcome these issues, and provide a seamless connection to and from the provider.

Ingate SIP Trunking can also handle authentication at the service provider to validate the enterprise as the correct user of the SIP trunk. It also provides the flexibility to interoperate with carrier-specific requirements like addressing formats.

NAT/Firewall Traversal

The SIP Trunking software module, working in conjunction with an Ingate Firewall or SIParator, solves the Network Address Translation (NAT) traversal issues that are faced by businesses using a SIP trunk. Together, they control both incoming and outgoing communications and route the communication to the intended users. All voice traffic (as well as data traffic) must traverse the enterprise firewall/NAT. However, SIP traffic cannot traverse traditional enterprise firewalls and NAT devices. As a result, the firewall/NAT device blocks all SIP traffic, which includes VoIP. Ingate resolves this issue, enabling enterprises to utilize SIP trunks.

Security Over The Public Internet

Ingate SIP Trunking works with Ingate Firewalls and SIParators to serve as the demarcation point of the enterprise network. Ingate's enterprise-class Firewalls secure data and SIP traffic, while SIParators secure SIP media while leaving the traditional firewall in place (working in parallel to the SIParator) to secure data traffic.

Both products feature Ingate's full SIP proxy technology, which allows for advanced filtering, verification, authentication and routing, as well as dynamic control of the opening and closing of media ports. They also encrypt the SIP signaling using Transport Layer Security (TLS) and media (voice, video, etc.) using Secure RTP (SRTP). With encryption, sessions are kept private with no chance of eavesdropping.

Support for Multiple SIP Trunks

Eliminating international calling costs, Ingate SIP Trunking supports multiple SIP trunks to direct international calls to national, or local, PSTN lines within the country being called. Businesses can use multiple service providers by establishing least cost routing rules, and switch between them depending on which offers the best possible rates. Long distance calls cost the same as a local call, reducing expenses for businesses as well as their customers, partners, etc. trying to reach, for example, the corporate sales force.

Ingate's flexible routing functions allow for calls from groups within the enterprise to be routed to specified service providers while others are routed to a different service provider, in order to leverage the best rates.

Support for multiple SIP trunks also offers redundancy. If a connection to one ITSP goes down, Ingate can immediately transfer the traffic to another ITSP.

ENUM support

Ingate SIP Trunking supports ENUM. This feature can automatically look up phone numbers to determine whether they match a known SIP address, allowing the call to be completed over the Internet (instead of transferring it out to the PSTN). Since no traffic is placed on the PSTN, ENUM provides an additional means of cost-savings for businesses that communicate with other firms using SIP.

For more information, visit us at www. ingate.com or write to info@ingate.com.

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Technical specifications Ingate SIP Trunking Module

Features
Demarcation Point
Dynamic opening/closing of ports
Strict SIP Parser
Filtering
IP Address
TO/FROM Header (SIP)
User name / Domain Name
Content/MIME Type
SIP Method
DIGEST Authentication
Definable using Regular Expressions
Routing
NAT/Firewall traversal of SIP signalling and media
Routing of incoming and outgoing calls
Handling multiple service providers
Least cost routing
Failover to secondary service provider if the first fails
Route to different service providers for different users
Dial Plan
Matching using predefined fields and regular expressions
ENUM Support
Support for outbound proxy
Handling for domains that cannot be resolved *via* DNS
SIP Proxy and B2BUA functionality
Rewriting to resolve interop issues
Domain modification of incoming calls
Domain modification of outgoing calls
Prefix Addition
Authentication
Authentication of enterprise at service provider
Encryption
SRTP encryption (media)
TLS encryption (SIP)
SIP& Media Transcoding
Compatibility
Interoperable with leading IP PBX vendors (contact Ingate for updated list)

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