Ingate Knowledge Base - a vast resource for information about all things SIP – including security, VoIP, SIP trunking etc. - just for the reseller community. *Drill down for more info!*

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The introduction of SIP to a network brings the challenge of protecting the network from an untrusted network, and the opportunity to manage the routing of calls to a degree not possible with traditional telephony. This instalment of our continuing Knowledge Base will review some of the things that can be configured with an Ingate Enterprise Session Border Controller to address both the challenges and opportunities.

Prepare for SIP Trunking

After a brief summer hiatus we are ready to resume the weekly Ingate Knowledge Base articles to help you better understand and prepare for SIP Trunking.

SIP Trunking is being brought to enterprises today by more and more service providers, both traditional telephony providers as well as new entrants. We have heard from many of them that they often encounter problems in delivering SIP Trunks effectively to their customers. Among the most prevalent are traversing the firewall and NAT of the enterprise, easily connecting to one of many IP-PBXs that may be at the enterprise site, providing a demarcation point between the delivery network and the customer, and having adequate diagnostic tools to isolate and resolve any issues that may develop. We will address each of these items in turn over the next few weeks.

Today I'd like to focus on another important item that is often overlooked, namely converging voice and data on the same enterprise LAN.

At first glance, some may think that having separate networks for voice and data is a natural extension of the way we have received telephony services for 100 years and data services for the past 25. But in reality, the promise of SIP, and of SIP Trunking, is the ability to use a variety of devices to communicate. Of course, a desktop SIP phone can be deployed on a separate LAN and voice calls will work. But what about the laptop and desktop PCs that have built in cameras and often pre-installed SIP clients? If those devices are not on the same LAN as the SIP Trunks, they cannot be used to place calls, which may eventually turn into video sessions. And as time goes on, more clients will have SIP capability, whether in a mobile device, or in devices yet to be invented.

When implementing SIP the service provider and the enterprise should look ahead to how all of the tools on the desktop can be used together to result in truly Converged Communications. With the introduction of an Enterprise Session Border Controller, like the Ingate SIParator, this is easily accomplished. Now both data and voice access can be shared on a single LAN, resulting in efficiencies and preparing the enterprise for the next step in the future of communications, while allowing the enterprise to maintain control and security over all services.

Unified Communications begins with a converged network, and a converged network is possible with the Ingate SIParator.

**We would like to hear from you.
Let us know of any topics you'd like to see addressed in future issues of the Knowledge Base series by writing to sofia@ingate.com or steve@ingate.com.**

WANT MORE INFORMATION

Follow the link to find out more
http://www.ingate.com/files/white_paper_What_is_SIP_Trunking_A.pdf

Please visit the Ingate SIP Trunking at Community http://sip-trunking.tmcnet.com/