What is sip trunk

Sociology, Communication

What a SIP trunk is? A SIP trunk is an IP connection that establishes a SIP communications link between your organization and an Internet telephony service provider (ITSP) beyond your firewall. Typically, a SIP trunk is used to connect your organization's central site to an ITSP. In some cases, you may also opt to use SIP trunking to connect your branch site to an ITSP. Unlike in traditional telephony, where bundles of physical wires were once delivered from the service provider to a business, a SIP trunk allows a company to replace these traditional fixed PSTN lines with PSTN connectivity via a SIP trunking service provider on the Internet.

SIP trunks can offer significant cost-savings for enterprises, eliminating the need for local PSTN gateways, costly ISDN BRIs (Basic Rate Interfaces) or PRIs (Primary Rate Interfaces). Why you would use a SIP trunk? Session Initiation Protocol (SIP) is used to initiate and manage Voice over IP (VoIP) communications sessions for basic telephone service and for additional real-timecommunicationservices, such as instant messaging, conferencing, presence detection, and multimedia. This section provides planning information for implementing SIP trunks, a type of SIP connection that extends beyond the boundary of your local network.

Deploying SIP trunking can be a big step toward simplifying your organization's telecommunications and preparing for up-to-date enhancements to real-time communications. One of the primary advantages of SIP trunking is that you can consolidate your organization's connections to the public switched telephone network (PSTN) at a central site, as opposed to its predecessor, time division multiplexing (TDM) trunking, which typically requires a separate trunk from each branch site. RFCs that discuss SIP trunking Best Practices for SIP Trunks:

Since SIP trunks are meant for interconnection between servers, they SHOULD run over TCP. Authentication SHOULD be done using mutual TLS authentication, with both sides of the trunk providing a TLS Certificate. TODO: might be interesting to recommend some practices for usage of phone numbers, but this might be out of scope here. Security Considerations: Servers providing SIP trunks will need to authenticate and authorize access to those trunk services. This specification recommends usage of the practices defined and required in RFC 3261 - mutual TLS uthentication - for this purpose. In some cases, the requests sent on SIP trunks can require confidentiality and message integrity. In such cases, usage of mutual authenticated TLS is RECOMMENDED. [RFC3261] Rosenberg, J. , Schulzrinne, H. , Camarillo, G. , Johnston, A. , Peterson, J. , Sparks, R. , Handley, M. , and E Schooler, " SIP: Session Initiation Protocol", RFC 3261, June 2002. [RFC3263] Rosenberg, J. and H. Schulzrinne, " Session Initiation Protocol (SIP): Locating SIP Servers", RFC 3263, June 2002.

Informative References: [RFC4458] Jennings, C. , Audet, F. , and J. Elwell, " Session Initiation Protocol (SIP) URIs for Applications such as Voicemail and Interactive Voice Response (IVR)", RFC 4458, April 2006. [RFC4480] Schulzrinne, H. , Gurbani, V. , Kyzivat, P. , and J. Rosenberg, " RPID: Rich Presence Extensions to the Presence Information Data Format (PIDF)", RFC 4480, July 2006. [RFC3903] Niemi, A. , " Session Initiation Protocol (SIP) Extension for Event State Publication", RFC 3903, October 2004.