

SIP Trunking

Breakout into the All-IP Provider Network

Company connections to the public voice network are referred to as SIP trunks. Depending on a customer's requirements in terms of throughput, security and availability, the design and implementation of a SIP trunk can differ from provider to provider.

The result is frequently recurring error patterns. A session border controller (SBC) plays an important role at the interface between customer and provider. The special features as well as the numerous variants of a SIP trunk are analyzed in detail in this SIP trunking course and examined using practical exercises.

Course Contents

- Session Initiation Protocol
- SIPS (SIP over TLS)
- High Availability
- SIP Connect 1.1
- E-SBC
- Interoperability
- Quality of Service
- Fax over IP
- T.38
- (S)RTP
- Troubleshooting
- STUN, TURN, ICE
- Emergency Calls
- Features
- Call Number Administration
- Privacy Extensions
- IP Centrex
- VoIP Gateways
- Mobility

E-Book The detailed digital documentation package, consisting of an e-book and PDF, is included in the price of the course.

Target Group

The course is aimed at anyone who is already involved with VoIP and SIP and needs a better understanding of a SIP trunk.

Prerequisites

Basic knowledge of VoIP and SIP is useful. This can be acquired, for example, in the courses VoIP Fundamentals - SIP, RTP & Co. in use or SIP - The universal signaling protocol.

This Course in the Web



You can find the up-to-date information and options for ordering under the following link:

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Table of Contents

SIP Trunking – Breakout into the All-IP Provider Network

1 State of affairs	3.2.1 Registration of a client	4.5.1 SIPS
1.1 TC Systems	3.2.2 Proxy Authentication	4.6 Securing the media stream
1.1.1 Control protocols	3.3 A session in detail	4.6.1 SRTP and SRTPC packet formats
1.2 System connection	3.3.1 INVITE	4.6.2 Key management
1.2.1 Features for DSS1	3.3.2 100 Trying	4.6.3 Key management for signaling
1.2.2 Performance features	3.3.3 180 Ringing and 183 Session Progress	4.6.4 Key management in Session Description Protocol
1.3 Voice as an application	3.3.4 Disconnection and BYE - What to consider?	4.6.5 DTLS-based key exchange
1.3.1 Voice coding and compression	3.4 SUBSCRIBE and NOTIFY	4.7 Availability
1.3.2 RTP transport	3.5 OPTIONS	4.7.1 Load balancing methods (1)
1.3.3 RTCP - information about RTP connections	3.6 PRACK - Reliable confirmation	4.7.2 SIP in the domain environment
1.4 IP Centrex	3.7 UPDATE - I still have one!	4.8 Fax on the SIP trunk
1.5 The provider network	3.8 REFER	4.8.1 Special features of fax transmission
1.5.1 User data	3.9 Session Description Protocol	4.8.2 Fax over IP - the possibilities
1.5.2 Provider Access	3.9.1 Structure of the Message Body with SDP	4.8.3 The fax as a normal VoIP call
1.5.3 QoS in Provider Access	3.9.2 SDP for advanced users	4.8.4 T.37 - Fax as e-mail attachment
1.5.4 Provider coupling	3.9.3 RTP profiles	4.8.5 T.38 - Fax in real time
2 SIP Trunking at a glance	3.10 Key tones	4.9 Features
2.1 The basic principle of SIP trunking	4 SIP Trunking in detail	4.9.1 MMTEL features
2.2 Connection variants	4.1 Identity and Authentication	4.9.2 Communication Diversion
2.2.1 Connection concepts without E-SBC	4.1.1 Registration Mode and Static Mode	4.10 Emergency call
2.2.2 Connection concepts with E-SBC	4.1.2 Registration of groups	5 SIPConnect
2.2.3 Firewalls	4.1.3 P-Header Extensions on the SIP Trunk	5.1 SIP Trunk according to SIP Forum
2.2.4 Encryption variants	4.1.4 P-Preferred/ Asserted ID	5.2 SIP Trunking Architecture
2.2.5 SIP trunking and NAT	4.1.5 Authentication using P-Asserted-Identity	5.3 Security
2.2.6 Redundancy concepts	4.1.6 P-Early-Media	5.4 Registration mode
2.2.7 SIP trunk with classic PBX system	4.2 Session Border Controller	5.5 Static mode
2.2.8 Number blocks	4.2.1 SBC in detail	5.6 Enterprise public identities
2.2.9 Distributed sites	4.2.2 Access control via session border controller	5.7 Handling of calls 1/2
2.3 Interconnection	4.3 Integration into the DMZ	5.7.1 P-Asserted-Identity
2.3.1 Packet Switched to Packet Switched	4.3.1 A network between networks	5.7.2 Call transfer
2.3.2 Packet Switched to Circuit Switched	4.3.2 Integration of the E-SBC into the DMZ	5.7.3 Media endpoints
2.4 Provider without IMS	4.4 NAT - Network Address Translation	5.8 NAT & Firewalls
3 Session Initiation Protocol	4.4.1 STUN tool for handling NAT	5.9 Emergency call
3.1 The basic idea of SIP trunking	4.4.2 Interactive Connectivity Establishment (ICE)	
3.1.1 The SIP message	4.4.3 SBC and NAT	
3.2 Registration and control	4.5 Securing signaling	

