

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

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.

Course Target

In this course, you will learn about the special features of a SIP trunk. However, the knowledge you gain is not limited to this one interface; instead, the entire process of corresponding call flows from a company connection across one or more provider networks is analyzed and potential sources of error are described. Practical examples allow you to expand your SIP knowledge in a targeted manner.

This Course in the Web



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