ExperTeach NETWORKING

SIP Trunking

Breakout into the All-IP Provider Network

Classic PBX access (BRI and PRI) for the connection of a company to the public voice network will be a thing of the past in a few years, In the course of the migration to Voice over IP, all PBX accesses therefore have to be replaced by an SIP trunk. In addition to the pure scaling of such an access, numerous technical details have to be clarified in the single case and challenges be mastered. The SIP Trunking course introduces the different connection scenarios with and without Enterprise Session Border Controller (E-SBC) and discusses the corresponding pros and cons. Redundancy concepts are just as important as ensuring voice quality and the transfer of fax messages. The frame stipulations of SIP Connect 1.1 are explained, which are meant to provide a homogenization. Hands-on exercises round off the course and illustrate the contents.

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 You will receive the comprehensive documentation package of the ExperTeach Networking series – printed documentation, e-book, and personalized PDF! As online participant, you will receive the e-book and the personalized PDF.

Target Group

The course addresses persons already concerned with VoIP and SIP, who wish to acquire a better understanding of SIP trunks.

Prerequisites

Basic knowledge of VoIP and SIP will be helpful. This know-how can e.g. be acquired in the courses VoIP Fundamentals—SIP, H.323 & Co.in Application 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: www.experteach-training.com/go/**SITR**

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SIP Trunking

Provider Signaling

ExperTeach

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