

# SIP

## The Universal Signaling Protocol

The Session Initiation Protocol (SIP) is now firmly established as the most important signaling protocol both in the enterprise environment and in provider networks. The main advantage of SIP is that it is easy to expand: New formats are no problem, synchronous as well as asynchronous data streams can be initiated, and the communication partners can be in a peer-to-peer or client-server relationship. After attending the course, participants will be familiar with the advantages, special features and possible applications of SIP architecture in general and SIP trunking in particular.

### Course Contents

- The Components SIP Proxy, Location Server, and User Agent
- Back-to-Back User Agent (B2BUA) and Session Border Controller (SBC)
- SIP Protocol: Message Types and their Setup
- Typical SIP Processes during Connection Setup and a SIP Call
- SIP URIs and Tel URIs: Address Formats, Identities, and their Application
- SDP: Setup, Options, and Profiles
- VoIP and Video over IP (RTP and Signaling) Data Streams
- Features—Instant Messaging—Presence
- Interaction of SIP with NAT and Firewalls
- Fax with T.38 and Interaction with SIP
- SIP as a Protocol in the IP Multimedia Subsystem (IMS)
- Application of SIP in Provider Networks
- SIP Trunking

Practice-related presentations and the analysis of traces will make the course contents easier to understand.

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

### Target Group

Persons working in planning and conceptual work will find themselves in this course just as much as employees who need to understand SIP at protocol level.

### Prerequisites

In-depth knowledge of voice and IP is a prerequisite. Basic knowledge of VoIP is very helpful for attending the course.

### This Course in the Web



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

[www.expertteach-training.com/go/KSIP](http://www.expertteach-training.com/go/KSIP)

### Reservation

On our Website, you can reserve a course seat for 7 days free of charge and in a non-committal manner. This can also be done by phone under +49 6074/4868-0.

### Guaranteed Course Dates

To ensure reliable planning, we are continuously offering a wide range of guaranteed course dates.

### Your Tailor-Made Course!

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