

VoIP Fundamentals

SIP, RTP & Co.in Application

Voice communication over IP is going to be the technology of the future. No matter whether it is in the corporate environment or for providers—soon all services relating to voice and video will be placed on an IP platform. If voice and IP merge, specialists from telecommunications and network sectors have to work our solutions together. This course offers a detailed introduction into Voice over IP (VoIP) for both sides. It includes the concepts of voice communication over IP and deals with RTP as the most important protocol for voice transmission as well as with the two signaling protocols SIP and H.323. Further focal points are Quality of Service and solutions for fax transmission. The course introduces into the planning and design of VoIP solutions for companies of differing sizes and shows migration strategies for future PSTN connections over SIP trunks. One day with hands-on exercises serves to explain the functioning of VoIP solutions.

Course Contents

- Voice over IP—Basics, Concepts, and Protocols
- Basics of Voice Communication
- Codecs and Bandwidths for VoIP/IP Telephony
- Media Streams over IP-RTP
- Basics of SIP—Terms, Concepts, and Processes
- Signaling over SIP—Registration and Call Setup
- Negotiation of Media Streams over SDP
- VoIP in Practical Application—Quality of Service and Fax Transmission over IP
- Basics of VoIP Security—Encryption, Firewalls, and NAT
- VoIP Design—Concepts for Small, Medium-Sized and Large Companies
- Cloud, Hosting, or IP-Centrex—The PBX at the Provider
- SIP Trunking—The VoIP Connection to the Provider
- Future Trends in Telephony

One day of hands-on exercises 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

The course addresses designers, consultants, decision-takers, and technicians from the areas of telecommunications and network technology who are looking for a basic introduction into the world of Voice over IP. It offers solid information required to plan and implement the migration to VoIP.

Prerequisites

Basic knowledge of the telecommunications and IP world is mandatory for attendance at this course.

Course Target

After the course, the students will be able to assess VoIP concepts, plan and implement migration concepts, and deepen their product-specific knowledge.

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/VOIP

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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