

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!

We can precisely customize this course to your project and the corresponding requirements.

Training		Prices, excl. of V.A.T.
Classes in Germany	4 Days	€ 2,395
Classes in Switzerland	4 Days	€ 3,190
Online Training	4 Days	€ 2,395
Date/course venue	Course language German	
02/06-05/06/25 München	25/08-28/08/25 Online	
02/06-05/06/25 Online	29/09-02/10/25 Hamburg	
02/06-05/06/25 Zürich	29/09-02/10/25 Online	
08/07-11/07/25 Düsseldorf	11/11-14/11/25 München	
08/07-11/07/25 Online	11/11-14/11/25 Online	
25/08-28/08/25 Frankfurt	11/11-14/11/25 Zürich	

Status 05/07/2025



EXPERTeach



Table of Contents

VoIP Fundamentals – SIP, RTP & Co.in Application

1 Introduction and Motivation

- 1.1 Voice Networks of Today and Tomorrow
 - 1.1.1 User Trends
 - 1.1.2 Trends in the Enterprise Market
 - 1.1.3 NGN—The Network of the Providers
 - 1.1.4 Trends in the Data Centers
 - 1.1.5 All IP—Internet for Everything
- 1.2 Voice over IP—Basics, Concepts, and Protocols
 - 1.2.1 VoIP Protocols
 - 1.2.2 VoIP in the ISO/OSI Model
 - 1.2.3 VoIP Signaling
 - 1.2.4 Media Streams
- 1.3 VoIP Infrastructure and Application Areas
 - 1.3.1 VoIP in the Enterprise Environment
 - 1.3.2 VoIP for Private Customers
 - 1.3.3 VoIP in the Provider Environment
 - 1.3.4 VoIP over the Internet
- 1.4 Conferencing and WebRTC
 - 1.4.1 WebRTC—The Open Conferencing Solution
 - 1.4.2 Browser or Apps
 - 1.4.3 Audio and Video for WebRTC
 - 1.4.4 Security Model
 - 1.4.5 WebRTC Application Examples

2 Media Streams with RTP

- 2.1 Voice Transfer
 - 2.1.1 Digitizing Voice
 - 2.1.2 Codecs—PCM and More
 - 2.1.3 Hybrid Encoding via CELP and MP-MLQ
- 2.2 Transporting Voice over IP
 - 2.2.1 The Anatomy of RTP Packets
 - 2.2.2 IP Addressing and Routing
 - 2.2.3 The Transport Protocols
- 2.3 Real-time Transport Protocol (RTP)
 - 2.3.1 Demands Made on RTP
 - 2.3.2 The Frame Format of RTP
 - 2.3.3 RTP Profiles
- 2.4 Real-Time Transport Control Protocol (RTCP)
 - 2.4.1 Classic RTCP
 - 2.4.2 RTCP Extended Reports (RTCP XR)
- 2.5 RTP Applications
 - 2.5.1 Dial Tones via DTMF
 - 2.5.2 Speech Pauses and VAD
 - 2.5.3 Bandwidths for VoIP
- 2.6 Parameters Influencing Voice Quality
 - 2.6.1 Delay—End-to-End
 - 2.6.2 Jitter and Jitter Buffer
 - 2.6.3 Packet Loss and Packet Loss Concealment
- 2.7 Voice Quality—Models and Calculation
 - 2.7.1 Mean Opinion Score (MOS)
 - 2.7.2 Subjective Perception: E-Model with R Factor
 - 2.7.3 POLQA and TOSQA

3 SIP—The Session Initiation Protocol

- 3.1 SIP—An Overview
 - 3.1.1 Standardization
 - 3.1.2 SIP in the ISO/OSI Model
 - 3.1.3 SIP Addressing: SIP URI and TEL URI
- 3.2 The Components of the SIP Architecture

- 3.2.1 The End Devices: User Agents
- 3.2.2 The SIP Proxy
- 3.2.3 SIP Gateways
- 3.3 Protocol Setup
 - 3.3.1 Setup of SIP Messages
 - 3.3.2 SIP Requests—SIP Methods
 - 3.3.3 SIP Responses
- 3.4 The Message Body
- 3.5 SDP—The Session Description Protocol
- 3.5 Registration and Authentication
 - 3.5.1 SIP Registration—Processes
 - 3.5.2 SIP Register without Authentication
 - 3.5.3 Register with Authentication
- 3.6 SIP Call Setup with Proxy
 - 3.6.1 SIP Invite over the Classic Proxy
 - 3.6.2 SIP Server Terminates the Dialog
 - 3.6.3 Domain Environments and DNS
- 3.7 Deployment of SIP Today and Tomorrow

4 Gateway Concepts for VoIP

- 4.1 Gateway Control
- 4.2 H.323 Used in Companies
 - 4.2.1 H.323 Implementation
- 4.3 MGCP
 - 4.3.1 Application Scenario Enterprise
 - 4.3.2 Application Scenario Provider
- 4.4 H.248/Megaco
 - 4.4.1 Termination and Context

5 VoIP—Practical Application

- 5.1 Encryption for VoIP
 - 5.1.1 Encryption of the Signaling over SIPs
 - 5.1.2 Encryption of the Media Stream via SRTP
 - 5.1.3 Key Management in the Session Description Protocol
 - 5.1.4 Encryption between Sites
- 5.2 VoIP with NAT and Firewalls
 - 5.2.1 VoIP and Stateful Firewalls
 - 5.2.2 VoIP and NAT
- 5.3 Solution #1: Application Layer Gateway (ALG)
- 5.4 Solution #2: STUN, TURN, and ICE
- 5.5 Solution #3: Hosted NAT Traversal (HNT)
- 5.6 Solution #4: Enterprise SBC
- 5.3 Fax Transmission over IP
 - 5.3.1 Special Features during Fax Transmission
 - 5.3.2 Fax Transmission Process
 - 5.3.3 Fax as a Default VoIP Call
 - 5.3.4 T.37—Fax as E-Mail Attachment
 - 5.3.5 T.38—Fax in Real-Time
 - 5.3.6 Error Patterns with Fax over IP
- 5.4 Quality of Service
 - 5.4.1 What is Quality of Service?
 - 5.4.2 Classification and Tagging
 - 5.4.3 Queuing
 - 5.4.4 Policing
 - 5.4.5 Traffic Shaping
 - 5.4.6 Admission Control

6 Concepts and Application Scenarios on the Enterprise Sector

- 6.1 Questions and Concepts

- 6.1.1 Access to Telephony Service Providers
- 6.1.2 Emergency Calls
- 6.1.3 Features
- 6.1.4 Features for VoIP vs. PSTN
- 6.2 Enterprise Solutions for a Site
 - 6.2.1 Voice VLANs and PoE
- 6.3 Enterprise Solutions for Several Sites
 - 6.3.1 WAN Interconnection—Private or Public
 - 6.3.2 Central PBX
 - 6.3.3 Decentralized PBXs
 - 6.3.4 Connection of Mobile Workplaces
- 6.4 Cloud Telephony
- 6.5 Access to Telephony Service Providers over SIP Trunks
 - 6.5.1 SIP Trunking Concept
 - 6.5.2 Integration of the SBC—Stand-alone Devices
 - 6.5.3 Registration Mode and Static Mode
 - 6.5.4 Registration at the SIP Trunk
 - 6.5.5 Identities: P-Asserted-Identity and From:
 - 6.5.6 Signaling at the SIP Trunk
- 6.6 Conferencing Solutions
 - 6.6.1 Conferencing Solution “on Premise”
 - 6.6.2 Cloud Conferencing plus Local SIP Trunk
 - 6.6.3 Cloud Conferencing plus Trunk Access to the Cloud
- 6.7 Quo Vadis—VoIP?

A VoIP Fundamentals—Lab Exercises

- A.1 Lab Setup for Practical Exercises
 - A.1.1 Virtual Lab
 - A.1.2 Lab Setup with Softphones
 - A.1.3 Lab Setup for Demonstration Purposes
 - A.1.4 Hardware and Software
- A.2 Hands-on Exercises
 - A.2.1 Lab Exercise—Registration
 - A.2.2 Lab Exercise—Basic Call with SIP
 - A.2.3 Lab Exercise—Call Hold
 - A.2.4 Lab Exercise—Call Transfer
 - A.2.5 Lab Exercise—Simple Three-Party Conference
 - A.2.6 Lab Exercise—Videotelephony
 - A.2.7 Lab Exercise—Selection of the Codec
 - A.2.8 Lab Exercise—Incompatible Codecs
- A.3 Wireshark in a Brief Overview
 - A.3.1 Capturing with Wireshark
 - A.3.2 Capture Filter
 - A.3.3 Preferences
 - A.3.4 Preferences and Profiles
 - A.3.5 Display Filters
- A.4 Evaluation of RTP with Wireshark
 - A.4.1 RTP Statistics
 - A.4.2 RTP Stream Analysis
 - A.4.3 Lab Exercise: RTP Basic Functions
 - A.4.4 Lab Exercise: RTP Operation
- A.5 SIP Analysis with Wireshark
 - A.5.1 VoIP Calls—Statistics
 - A.5.2 SIP Statistics
 - A.5.3 Lab Exercise: SIP Registration
 - A.5.4 Lab Exercise: SIP—Basic Call with Wireshark

B List of Abbreviations

