SIP ESSENTIALS

Course Code: 3251

Learn the SIP protocol and important protocols related to SIP implementations.

In this course, you will learn about Session Initiation Protocols (SIPs) and important protocols related to SIP implementations through a process of lecture and hands-on training. Gain insights into what SIP is, how it works, and get a practical guide on how to use it. The lessons in this course are clear, very technical, and always practical. Since more than half are hands-on, you can investigate and reinforce each lesson. In this course, you’ll examine how SIP interoperates into the current telecommunications network by going beyond the basics of the protocol and getting a big picture understanding of how it all fits together.

What You’ll Learn

- Why SIP is a valuable protocol
- SIP architecture
- SIP Uniform Resource Indicators (URIs)
- SIP headers
- SIP-related IP Services
- SIP for Instant Messaging and Presence Leveraging Extension (SIMPLE)
- How SIP intelligently routes calls over any network
- SIP security

Who Needs to Attend

- Individuals who want to learn more about SIP
- Individuals responsible for installing SIP trunking or internetwork telephone systems using SIP
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| CLASSROOM LIVE | $1,795 USD | 5 days |

## Classroom Live Outline

1. **VoIP Introduction**
   - Circuit Switching
   - VoIP Protocols
   - VoIP Deployments: First Installations to Now
   - SIP and the Softswitch

2. **SIP Architecture**
   - The SIP Architecture
   - UA, Proxy, Redirect, Forking, and B2BUA
   - Multimedia Architecture
   - RTP/RTCP
   - SDP
   - Methods
   - REGISTER
   - INVITE and ACK
   - UPDATE
   - OPTIONS
   - REFER
   - CANCEL
   - SUBSCRIBE and NOTIFY
   - MESSAGE
   - BYE
   - SIP Responses
   - Via Path
   - Record-Route

3. **REGEX**
   - Regular Expression

4. Routing the SIP INVITE
• The Via: path
• Creation of Response-Path
• Response Merging
• Record-Route: and Route:
• Forking
• Loops and Spirals

5. The SIP Dialog
• The Purpose of the SIP Dialog
• How to Begin and End a Dialog
• The Dialog ID

6. SIP Entities
• B2BUA
• Proxy
• SBC
• Outbound Proxy
• UA

7. SIP Call Flow Examples
• The Following Call Flows Set Up and Examined Using Wireshark
• REGISTER
• Normal Call
• Busy
• Redirect
• Transfer (REFER)

8. SIP Call Routing
• How SIP Routing Is Used to Route Calls
• Use of Record-Route in Stateless Routing Proxies
• How SIP Is Used in the PSTN Migration to an All IP Network

9. SIP Uniform Resource Indicators (URIs)
• Generic URI information (RFC 2396)
• Direct or Proxy
• PSTN Number (RFC 2808)
• Instant Messaging
• Presence
• In Registrations

10. SIP Message Headers
• Via
• Branch
• Max-Forwards
• Dialog (To, From, and tag= fields, Call-ID)
• CSeq
• Proxy Authenticate
• Proxy-Authorize
- Contact
- Expires
- User-Agent
- Content-Length
- Allow
- Supported
- P-Access
- Network-Info
- P-Charging-Vector, P-Preferred-Identity, P-Asserted-Identity
- Authorization
- Security-Client
- Security-Server
- Content-Type

11. Session Description Protocol (SDP)
   - Session Parameters
   - SDP Format
   - Extending SDP
   - SDPng
   - Media Negotiation
   - Changing Session Parameters
   - Controlling the Media

12. SIP and the DNS
   - Basic Resource Records (RR)
   - A-Record, SOA, NS Record, MX Record
   - The SRV Record (RFC 2782)
   - How SIP Uses the SRV Record (RFC 3263 Locating SIP Servers)
   - How to Configure a SRV Record
   - The NAPTR Record (RFC 2915)

13. ENUM
   - ENUM Protocol (RFC 3761)
   - Dynamic Delegation Discovery System (RFC 3401, 3402, 3403, 3761, 3764)
   - How SIP Uses ENUM

14. SIP and DHCP
   - DHCP Protocol
   - SIP DHCP Options

15. Interoperating SIP with Legacy PSTN Signaling
   - Call Transfer (REFER)
   - 183 Early Media
   - Interworking SIP with Local Call Control (E&M or DID)
   - SIP and the PSTN
   - SIP-T

16. RTP and Real-Time Control Protocol (RTCP)
• Dealing Packet Loss, Latency, Jitter
• How RTP Defines the Session
• Session Description Protocol
• The RTP Profile
• The RTP Payload Type Field
• RTP Telephony Events (RFC 2833)
• How RTP Removes Jitter
• How RTP Handles Packet Loss
• How RTP Identifies the Talking Party
• How RTP Handles Silence Suppression
• How RTP Handles Fixed Length Packets (Padding)
• How RTP is Used to Mix Voice (Conference Calls)
• The RTP Header
• RFC 2833 Protocol
• RTP Control Protocol
• SDES
• Sender/Receiver Reports
• Bye Reports

17. DTMF Handling
• Inband
• RFC 2833
• SIP INFO

18. Fax Handling
• Inband
• Fax Relay
• T.38

19. Presence
• SIMPLE – SIP for Instant Messaging and Presence Leveraging Extensions
• Terminology
• Framework
• Resource List Manipulation Requirements
• Authorization Policy Manipulation
• Acceptance Policy Requirements
• Notification Requirements
• Content Requirements
• General Requirements

20. SIP Timers
• T1, T2, T4
• Timer A – K

21. SIP Security
• Security for Call Setup
• Authentication
• S/MIME
• TLS

22. NAT Traversal
• How NAT Operates on SIP and SDP
• NAT Types
• STUN
• TURN
• ICE

23. SIPp: A SIP Testing Tool
• SIPp
• SIPp XML Examples

Classroom Live Labs
Lab 1: Construct and Enable a VoIP Network
Lab 2: SIP User Agent Configuration
Lab 3: Direct UA to UA Routing with No Proxy
Lab 4: Proxy Based SIP Routing
Lab 5: Adding Authorized UAs to a Domain
Lab 6: Intra Domain Routing (SIP in the Same Domain)
Lab 7: SIP REGISTER – Registering a SIP UA
Lab 8: Registering a SIP UA Soft Client
Lab 9: Registering a SIP UA Client to a Mobile Device
Lab 10: Inter Domain Routing (SIP in Different Domains)
Lab 11: Strip off the Leading 9
Lab 12: PDT Management
Lab 13: Using Wireshark
Lab 14: Capture a SIP Registration via Wireshark
Lab 15: Capture a Normal SIP Call via Wireshark
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Lab 18: Capture a Call Forward via Wireshark
Lab 19: Via, Record Route, and Route Headers
Lab 20: Examining Max Forwards
Lab 21: INVITE with SDP – sendonly vs. sendrecv
Lab 22: Silence Suppression
Lab 23: DTMF RFC 2833 and SIP INFO
Lab 24: SIP B2BUA Configuration Example
Lab 25: Register Linksys SIP Phone with Asterisk PBX
Lab 26: SIP Presence (NOTIFY)
Lab 27: RTP Relay
Lab 28: Direct RTP Flow between Two UAs – 3PCC
Lab 29: ENUM Call Routing
Lab 30: Testing SIP Connectivity Using SIP OPTIONS
Lab 31: Advanced: SIP Testing with SIP-p
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Jun 8 - 12, 2020 | 10:00 AM - 6:00 PM EST
Sep 14 - 18, 2020 | 10:00 AM - 6:00 PM EST
Dec 7 - 11, 2020 | 10:00 AM - 6:00 PM EST
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