SIP Essentials

- 5 Day Course
- Lecture & Labs (with Instructor led training)

Session Initiation Protocol (SIP) is the protocol uniting every communication management suite, be it Cisco Call Manager, Avaya Session and Communication Manager, Avaya IP Office, Oracle Session Border Controllers, Ericsson IMS cores, Asterisk, ShoreTel and Mitel products. You'll make live call analyses with Wireshark and TCPDump. Via the PCAPs you create, as well as those accessed from an extensive library of premade captures, you'll have no problems understanding why SIP makes the phone ring, how RTP carries real time voice and video, or troubleshooting and identifying errors.

The lessons in this course are clear and very technical. Attending students will receive access to the Alta3 Research SIP certification exam. Upon successful completion of the exam, students will be awarded a SIP certificate.

- SIP Requests and Responses
- Live call capture
- Wireshark Analysis (pcaps & ng-pcap)
- RTP Voice and Video
- Session Description Protocol (SDP) negotiation
- DTMF transmission
- SIP Routing and Dialplan construction (regular expression)
- Call flow analysis
- Testing with SIP-p
- Troubleshooting (failed calls, 1-way or no way voice)
- STUN / TURN / ICE

1. SIP Introduction

- SIP Message Format
- Legacy Call Control
- Compare SIP
- Packetizing Voice
- SIP Call Flow
- How SIP Routes Media
- SIP Call Control
- SIP in 4G

2. SIP Architecture

- SIP UA
- SIP Requests
- SIP Response
- SIP URI
- SIP Architecture
- SIP Domain
- SIP Registration
- SIP Call Routing
- Loose Routing

3. Regular Expression

- Metacharacters
- Substitution
- REGEX Modifications

4. Routing the SIP INVITE

- Proxy Routing
- Via and Record-Route

5. The SIP Dialog

- SIP Dialog
- The reINVITE

6. SIP Entities

- SIP Topology
- SIP Proxy
- B2BUA
- Outbound Proxy

7. SIP Call Flow Examples

- Wireshark Colors
- Wireshark Preferences
- SIP Stack
- REGISTER with Authentication
- Wireshark Analysis of SIP Dialog
- SIP Redirect
- CFNA
- REFER and Call Transfer

8. SIP Call Routing

- PRACK 100-rel
- Call Forking
- Loop and Spiral
- Third Party Call Control
- Path Minimization
- SIP in the PLMN
- OPTIONS Method

9. SIP Uniform Resource Indicators (URIs)

- URI vs. URL vs. URN
- SIP URI Examples
- URI Delimiters
- SIP and SIPs
- tel URI
- URI Escape Codes

10. SIP and the DNS

- Zone File
- SOA and NS Records
- A-Record
- SRV Record
- NAPTR Record
- Locating SIP Servers

11. ENUM
    - ENUM Database Example
    - ENUM Query and Response
    - ENUM REGEX
    - Post ENUM Routing
12. SIP and the PSTN
    - Early Media
    - SIP-T and SIP-I
13. SIPp
    - SIP QA testing
    - SIP DOS Testing
14. SIP Message Headers
    - SIP Header Overview
    - Dialog ID Headers
    - User-Agent
    - SIPp Header Modification
    - Proxy-Authenticate
    - Allow and Supported
    - History Info
    - Join
    - Session Expires
    - PPI and PIA
    - Establish Service Path
    - IMS Hosted
    - Content-Type
15. Session Description Protocol (SDP)
    - SDP Background
    - SDP Format
    - SIP = one way?
    - SDP Lines
    - SDP Offer/Answer
    - Call Hold
16. RTP and Real-Time Control Protocol (RTCP)
    - RTP Headers
    - RTP Dejitter
    - Conferencing
    - RTCP
17. DTMF Handling
    - DTMF
    - SIP INFO
    - RFC 2833
18. Fax Handling
    - T.30
    - T.38
    - SDP RFC 3407
19. Presence

- Presence Overview
- PIDF XML Example
- Rich Presence
- Presence Message Flow
- Instant Messaging

20. SIP Timers

- Standard Timer Values
- Session-Expires

21. SIP Security

- Security for Call Setup
- Authentication
- S/MIME
- TLS

22. SIP NAT Traversal

- NAT
- NAT Types
- STUN & TURN

Course Overview

What You'll Learn

- DEEP DIVE into the SIP protocol and related protocols including RTP, RTCP, DNS and more
- Differences in RFC 2543 and RFC 3261
- Overview of relevant RFCs
- Using Wireshark
- Deploying a SIP proxy and SIP UA
- SIP Troubleshooting
- Audio Troubleshooting
- AI LLM prompt engineering for relevant configuration snippets and solutions

Outline

AI LLM Toolkit

- 💻 Lecture + Lab: Large Language Model toolkit for AI Solution Assistance

VOIP Fundamentals

- 💬 Lecture: Introduction to VoIP

Packet Captures

- 💻 Lecture + Lab: Upload PCAPs for analysis (OPTIONAL)
- 💻 Lecture + Lab: Introduction to Wireshark
- 💻 Lecture + Lab: Termshark

SIP Registrars

- 💬 Lecture: SIP Architecture
- 💻 Lecture + Lab: Successful REGISTER by a User Agent
- 💻 Lecture + Lab: REGISTER Fails Auth
- 💻 Lecture + Lab: deREGISTER Log Out

SIP INVITE

- 💬 Lecture: Regular Expression
- 💻 Lecture + Lab: Building a Dial Plan with RegEx
- 💬 Lecture: Routing the INVITE
- 💻 Lecture + Lab: The SIP INVITE
- 💻 Lecture + Lab: SIP INVITE Packet Analysis with Wireshark

Establishing Calls

- 💬 Lecture: SIP Dialog
- 💻 Lecture + Lab: Troubleshooting Common SIP Failures with Wireshark
- 💬 Lecture: SIP Entities

Call Flows

- 💬 Lecture: Basic SIP Call Flows
- 💬 Lecture: SIP 3xx Redirection
- 💻 Lecture + Lab: Call Forwarding or 3xx responses
- 💬 Lecture: SIP REFER
- 💻 Lecture + Lab: SIP REFER for Call Transfer
- 💬 Lecture: SIP PRACK 100rel
- 💻 Lecture + Lab: SIP PRACK 100rel
- 💬 Lecture: Call Forking
- 💻 Lecture + Lab: Call Forking

SIP Proxies

- 💬 Lecture: Call Routing
- 💻 Lecture + Lab: INVITE Relay by SIP Proxies
- 💻 Lecture + Lab: No Record Routes
- 💬 Lecture: SIP URIs
- 💻 Lecture + Lab: CANCELed SIP call
- 💻 Lecture + Lab: Global Failures or 6xx responses

Supporting Systems

- 💬 Lecture: SIP and the DNS
- 💬 Lecture: ENUM
- 💬 Lecture: Interop with the PSTN

SIP Tools

- 💻 Lecture + Lab: Install Asterisk
- 💻 Lecture + Lab: Linux Fundamentals
- 💻 Lecture + Lab: Using vim
- 💻 Lecture + Lab: Making pcaps with tcpdump
- 💻 Lecture + Lab: Making pcaps with tshark
- 💬 Lecture: SIPp
- 💻 Lecture + Lab: SIPp SIP Tester
- 💻 Lecture + Lab: SIP Swiss Army Knife

SIP Headers

- 💬 Lecture: Common SIP Headers

Session Description Protocol

- 💬 Lecture: Session Description Protocol
- 💻 Lecture + Lab: Session Description Protocol
- 💻 Lecture + Lab: SDP Video Call Setup
- 💻 Lecture + Lab: SDP Video Call Setup Fails

Real-Time Transport Protocol

- 💬 Lecture: Real-time Transport Protocol
- 💻 Lecture + Lab: One-Way Media

Dual Tone Multi Frequency

- 💬 Lecture: Transmitting DTMF
- 💻 Lecture + Lab: Methods for Transport of DTMF

Fax

- 💬 Lecture: Fax Handling

Presence

- 💬 Lecture: Presence
- 💻 Lecture + Lab: SIP PUBLISH
- 💻 Lecture + Lab: Presence and IM Exchange

SIP Timers

- 💬 Lecture: SIP Timers

SIP Security

- 💬 Lecture: SIP Security

NAT Issues

- 💬 Lecture: NAT