Session Initiation Protocol (SIP) Essentials

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

Review this course online at https://www.alta3.com/courses/sip

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

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