SIP Essentials Training

Lecture, Labs, and Certification Exam

5 Day Course

COURSE DESCRIPTION

In this course, students learn Session Initiation Protocol (SIP), as well as other protocols related to SIP implementations. Lecture is highly technical and reinforced with hands-on labs. Students manage SIP communications within a domain, and make packet captures with Wireshark.

In addition to what SIP is and how SIP works, class provides a practical guide on how to implement SIP within your environment. Students will learn how to interoperate in the current telecommunications network, and get a big picture understanding of how it all fits together.

By the conclusion of this course, students will receive access to Alta3 Research's SIP certification exam. Upon successful completion of the exam, students will be awarded a SIP certificate.

DAY 1 - SIP Architecture

On day one, we explain what VoIP is, where SIP fits into the VoIP model, how Packet Switching differs from Circuit Switching, and the network entities that commonly 'speak' SIP. It is an introduction to RFC 3261, SIP request and response codes, and a deep-dive into the SIP REGISTER.

Ch 1. VoIP Introduction

- Circuit Switching
- VoIP Protocols Overview
- VoIP Deployments from the First Installations to Now
- SIP and the Softswitch

Ch 2. SIP Architecture

- The SIP Architecture
- UA, Proxy, Redirect, Forking, B2BUA
- Multimedia Architecture
- RTP/RTCP
- SDP

- Methods: REGISTER, INVITE and ACK, UPDATE OPTIONS, CANCEL, REFER, SUBSCRIBE and NOTIFY, MESSAGE, BYE
- SIP Responses
- Via Path
- Record-route

DAY 1 - Lab Topics

Lab 0. Understanding the Lab Environment
Lab 1. Using Wireshark
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. Registering a SIP UA (Capturing a SIP REGISTER with Wireshark)

DAY 2 - Understanding the SIP Dialog

Day two is all about brining SIP protocol into focus. We start to refine the students understanding of how SIP headers 'steer' messages through the network, and examine how two SIP entities are able to build trust with the creation of a SIP dialog.

Ch 3. REGEX

- Regular Expression
- Building SIP Dialplans with REGEX

Ch 4. Routing the SIP INVITE

- The Via: path
- Creation of Response-Path
- Response Merging
- Record-route and Route:
- Forking
- Loops and Spirals

Ch 5. The SIP Dialog

- The Purpose of the SIP Dialog
- How to Begin and End a Dialog
- The Dialog ID

Ch 6. SIP Entities

- User Agents
- Back-to-Back UAs
- Proxy
- Session Border Controller
- Outbound Proxies

DAY 2 - Lab Topics

Lab 7. Intra Domain Routing (SIP routing within the same domain)
Lab 8. Inter Domain Routing (SIP routing to different domains)
Lab 9. Digit translation
Lab 10. Prefix domain transfer (PDT) management
Lab 11. Capturing a “normal’ SIP call via Wireshark

DAY 3 - Advanced SIP Messaging

Day three begins a deep-dive into SIP messaging, including examining REFER and 3xx type messages. All common, and some uncommon, headers are examined using Wireshark packet-capture techniques.

Ch 7. SIP Call Flows Examples

- REGISTER
- Normal call
- Busy
- Redirect
- Transfer (REFER)

Ch 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

Ch 9. SIP Uniform Resource Indicators (URIs)

- Generic URI Information (RFC 3986)
- Direct or Proxy
- PSTN Number (RFC 2808)
- Instant Messaging
- Presence
- In Registrations

Ch 10. SIP Message Headers

- SIP Dialog (To:, From:, tag= fields, Call-ID:)
- Via: & Branch
- Max-Forwards:
- 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:

DAY 3 - Lab Topics

Lab 12. Capturing a call to a vacant seat via Wireshark
Lab 13. Capturing a call to a busy seat via Wireshark
Lab 14. Capturing a call-forward (3xx response) via Wireshark
Lab 15. Via, Route, and Record-Route headers
Lab 16. Examining and manipulating Max-Forwards header

DAY 4 - Session Description Protocol, Real-time Transport Protocol, and Legacy Interop

On day four, students learn about SDP's role in the setup of media (RTP). Both RTP audio and video streams are examined. The role DNS plays on SIP routing (RFC 3263) is also made clear. By the end of this day, students should be comfortable capturing SIP, SDP, RTP, RTCP, and DNS messages in Wireshark, and understand how these protocols are working together to provide VoIP services.

Ch 11. Session Description Protocol (SDP)

- Session Parameters
- SDP Format
- Extending SDP
- SDPng
- Media Negotiation
- Changing Session Parameters
- Controlling the Media

Ch 12. SIP and the DNS

- Basic Resource Records (RR)
- A-record, SOA, NS Record, MX Record (Important for Comparison to the SRV 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)

Ch 13. ENUM

- ENUM Protocol RFC 3761
- Dynamic Delegation Discovery System (RFC 3401, 3402, 3403, 3761, 3764)

- How SIP Uses ENUM

Ch 14. SIP and DHCP

- DHCP Protocol
- SIP DHCP Options

Ch 15. Interoperating SIP with Legacy STN Signaling

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

Ch 16. Real-time Transport Protocol (RTP) and Real-time Control Protocol (RTCP)

- Dealing with 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

DAY 4 - Lab Topics

Lab 17. Capturing SDP offer and answer
Lab 18. Silence suppression
Lab 19. DTMF RFC 2833 and SIP INFO
Lab 20. SIP Back-to-Back UA configuration example (Asterisk)
Lab 21. REGISTER SIP device to Back-to-Back UA
Lab 22. Capture SIP call through a Back-to-Back UA and compare to a Proxy
Lab 23. RTP Relay

DAY 5 - Applications of SIP and Troubleshooting

On day five, students wrap up an understanding of some legacy interop concepts from the previous day (DTMF and Fax), however most of the day will be spent understanding how SIP is applied in real environments (delivering rich presence features), how to keep your SIP environment secure (security), and finally how to troubleshoot SIP (common issues caused by NAT and troubleshooting with SIP-p).

Ch 17. DTMF Handling

- Inband
- RFC 2833
- SIP INFO

Ch 18. Fax Handling

- Inband
- Fax Relay
- T.38

Ch 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

Ch 20. SIP Timers

- T1, T2, T4
- Timer A-K

Ch 21. SIP Security

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

Ch 22. SIP NAT Traversal

- How NAT operates on SIP and SDP
- NAT Types
- STUN
- TURN
- ICE

Ch 23. SIPp: A SIP Testing Tool

- SIPp

- SIPp XML Examples

DAY 5 - Lab Topics

Lab 24. Real-Time Control Protocol (RTCP)
Lab 25. Routing with DNS / ENUM
Lab 26. Testing Connectivity using SIP OPTIONS
Lab 27. SIP testing with SIP-p