SIP ESSENTIALS

Course Code: 3251

Learn about Session Initiation Protocol (SIP) and other protocols related to SIP implementations.

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.

What You'll Learn

- 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

Who Needs to Attend

Any company or individual who wants to advance their comprehension of VoIP and SIP

Prerequisites

To gain the most from this class, you should have networking experience.

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CLASSROOM LIVE	$3,650 CAD	5 Day

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
Lab 16: Capture a Call to a Vacant Number via Wireshark
Lab 17: Capture a SIP Call to Busy Number via Wireshark
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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VIRTUAL CLASSROOM LIVE	$3,650 CAD	5 Day

Virtual Classroom Live Outline

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
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18. **Fax Handling**
    - T.30
    - T.38
    - SDP RFC 3407
19. **Presence**
    - Presence Overview
    - PIDF XML Example
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20. **SIP Timers**
    - Standard Timer Values
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22. **SIP NAT Traversal**
    - NAT
    - NAT Types
    - STUN & TURN

Virtual Classroom Live Labs

- **Summary - A short overview on how to navigate the lab environment**
    1. Welcome to Alta3 Labs
    2. Linux Fundamentals
    3. Using vim
- **Packet Captures - Learn how to use wireshark to test and debug SIP calls**
    1. Wireshark
    2. Making pcaps with tcpdump
    3. Making pcaps with tshark
- **SIP Registrars - Analyze the SIP REGISTRATION**
    1. REGISTER a SIP UA to SipGate
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- **SIP Proxies - Analyze call flow through a proxy**
    1. INVITE Relay by SIP Proxies
    2. Canceled SIP call
    3. No Record Routes
- **SIP Tools - Use various SIP testing tools to view special call flows.**
    1. SIPp SIP Tester
    2. SIP Swiss Army Knife
- **Exploring Media**
    1. Methods for Transport of DTMF

Apr 20 - 24, 2026 | 10:00 AM - 6:00 PM EDT

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Sep 14 - 18, 2026 | 10:00 AM - 6:00 PM EDT

Nov 30 - Dec 4, 2026 | 10:00 AM - 6:00 PM EST

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ON-DEMAND	$345 CAD

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