

SIP Essentials

Duration: 5 Days **Course Code: GK3251** **Delivery Method: Company Event**

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.

Target Audience:

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

Objectives:

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

Attendees should meet the following prerequisites:

- Networking Experience

Testing and Certification

Recommended as preparation for the following exams:

- There is no exam aligned to this course
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Content:

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

SIP Architecture

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

Regular Expression

- Metacharacters
- Substitution
- REGEX Modifications

Routing the SIP INVITE

- Proxy Routing
- Via and Record-Route

The SIP Dialog

- SIP Dialog
- The reINVITE

SIP Entities

- SIP Topology
- SIP Proxy
- B2BUA
- Outbound Proxy

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

SIP Call Routing

- PRACK 100-rel
- Call Forking
- Loop and Spiral
- Third Party Call Control

SIP and the PSTN

- Early Media
- SIP-T and SIP-I

SIP

- SIP QA testing
- SIP DOS Testing

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

Session Description Protocol (SDP)

- SDP Background
- SDP Format
- SIP = one way?
- SDP Lines
- SDP Offer/Answer
- Call Hold

RTP and Real-Time Control Protocol (RTCP)

- RTP Headers
- RTP Dejitter
- Conferencing
- RTCP

DTMF Handling

- DTMF
- SIP INFO
- RFC 2833

Fax Handling

- T.30
- T.38
- SDP RFC 3407

Presence

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

SIP Timers

Labs

Summary - A short overview on how to navigate the lab environment

- Welcome to Alta3 Labs
- Linux Fundamentals
- Using vim

Packet Captures - Learn how to use wireshark to test and debug SIP calls

- Wireshark
- Making pcaps with tcpdump
- Making pcaps with tshark

SIP Registrars - Analyze the SIP REGISTRATION

- REGISTER a SIP UA to SipGate
- Successful REGISTER by a User Agent
- REGISTER Fails Auth
- Live capture of SIP REGISTER with tcpdump

SIP Calls - Analyze SIP call flows via B2BUA

- The SIP INVITE
- SIP INVITE Packet Analysis with Wireshark
- Troubleshooting Common SIP Failures with Wireshark
- Live capture of SIP INVITE with tcpdump

SIP Proxies - Analyze call flow through a proxy

- INVITE Relay by SIP Proxies
- Canceled SIP call
- No Record Routes

SIP Tools - Use various SIP testing tools to view special call flows.

- SIPp SIP Tester
- SIP Swiss Army Knife

Exploring Media

- Methods for Transport of DTMF

- Path Minimization
- SIP in the PLMN
- OPTIONS Method

SIP Uniform Resource Indicators (URIs)

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

SIP and the DNS

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

ENUM

- ENUM Database Example
- ENUM Query and Response
- ENUM REGEX
- Post ENUM Routing

- Standard Timer Values
- Session-Expires

SIP Security

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

SIP NAT Traversal

- NAT
- NAT Types
- STUN ; TURN

Further Information:

For More information, or to book your course, please call us on Head Office 01189 123456 / Northern Office 0113 242 5931

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