SIP Essentials Training

Duration: **5 Day**

BILGINC IT Academy
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Overview

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.

Is This The Right Course?

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

Who Should Attend?

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

Prerequisites

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What You Will 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

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

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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

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- 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