SIP Essentials Training

Course Summary

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. Upon successful completion of the exam, students will be awarded a SIP certificate.

Topics

- VoIP Introduction
- SIP Architecture
- REGEX
- Routing the SIP INVITE
- The SIP Dialog
- SIP Entities
- SIP Call Flows Examples
- SIP Call Routing
- SIP Uniform Resource Indicators (URIs)
- VoIP Introduction
- SIP Architecture
- REGEX
- Routing the SIP INVITE
- The SIP Dialog
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- SIP Call Flows Examples
- SIP Call Routing
- SIP Uniform Resource Indicators (URIs)
- Real-time Transport Protocol (RTP) and Real-time Control Protocol (RTCP)
- DTMF Handling
- Fax Handling
- Presence
- SIP Timers
- SIP Security
- SIP NAT Traversal
- SIPp: A SIP Testing Tool

Duration

Five Days

SIP Essentials Training

Course Outline

I. ***VoIP Introduction***
   A. Circuit Switching
   B. VoIP Protocols Overview
   C. VoIP Deployments from the First Installations to Now
   D. SIP and the Softswitch

II. ***SIP Architecture***
   A. The SIP Architecture
   B. UA, Proxy, Redirect, Forking, B2BUA
   C. Multimedia Architecture
   D. RTP/RTCP
   E. SDP
   F. Methods: REGISTER, INVITE and ACK, UPDATE OPTIONS, CANCEL, REFER,
   G. SUBSCRIBE and NOTIFY, MESSAGE, BYE
   H. SIP Responses
   I. Via Path
   J. Record-route
   - Labs:
     - Understanding the Lab Environment
     - Using Wireshark
     - SIP User Agent Configuration
     - Direct UA to UA Routing with No Proxy
     - Proxy Based SIP Routing
     - Adding Authorized UAs to a Domain
     - Registering a SIP UA (Capturing a SIP REGISTER with Wireshark)

III. ***REGEX***
   A. Regular Expression
   B. Building SIP Dialplans with REGEX

IV. ***Routing the SIP INVITE***
   A. The Via: path
   B. Creation of Response-Path
   C. Response Merging
   D. Record-route and Route:
   E. Forking
   F. Loops and Spirals

V. ***The SIP Dialog***
   A. The Purpose of the SIP Dialog
   B. How to Begin and End a Dialog
   C. The Dialog ID

VI. ***SIP Entities***
   A. User Agents
   B. Back-to-Back UAs
   C. Proxy
   D. Session Border Controller
   E. Outbound Proxies
   - Labs
     - Intra Domain Routing (SIP routing within the same domain)
     - Inter Domain Routing (SIP routing to different domains)
     - Digit translation
     - Prefix domain transfer (PDT) management
     - Capturing a “normal’ SIP call via Wireshark

VII. ***SIP Call Flows Examples***
   A. REGISTER
   B. Normal call
   C. Busy
   D. Redirect
   E. Transfer (REFER)

VIII. ***SIP Call Routing***
   A. How SIP Routing is Used to Route CALLS
   B. Use of Record-Route in Stateless Routing Proxies
   C. How SIP is Used in the PSTN Migration to An All IP Network

IX. ***SIP Uniform Resource Indicators (URIs)***
   A. Generic URI Information (RFC 3986)
   B. Direct or Proxy
   C. PSTN Number (RFC 2808)
   D. Instant Messaging
   E. Presence

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Course Outline (cont.)

F. In Registrations

X. ***SIP Message Headers***

A. SIP Dialog (To:, From:, tag= fields, Call-ID:)
B. Via: & Branch
C. Max-Forwards:
D. CSeq:
E. Proxy-Authenticate:
F. Proxy-Authorize:
G. Contact:
H. Expires:
I. User-Agent:
J. Content-Length:
K. Allow:, Supported:
L. P-Access-Network-Info
M. P-Charging-Vector:
N. P-Preferred-Identity:
O. P-Asserted-Identity:
P. Authorization:
Q. Security-Client:
R. Security-Server:

S. Content-Type

- Labs
  - Capturing a call to a vacant seat via Wireshark
  - Capturing a call to a busy seat via Wireshark
  - Capturing a call-forward (3xx response) via Wireshark
  - Via, Route, and Record-Route headers
  - Examining and manipulating Max-Forwards header

XI. ***Session Description Protocol (SDP)***

A. Session Parameters
B. SDP Format
C. Extending SDP
D. SDPng
E. Media Negotiation
F. Changing Session Parameters
G. Controlling the Media

XII. ***SIP and the DNS***

A. Basic Resource Records (RR)
B. A-record, SOA, NS Record, MX Record (Important for Comparison to the SRV
C. Record)
D. The SRV Record (RFC 2782)
E. How SIP Uses the SRV Record (RFC 3263 Locating SIP servers)
F. How to Configure a SRV Record
G. The NAPTR Record (RFC 2915)

XIII. ***ENUM***

A. ENUM Protocol RFC 3761
B. Dynamic Delegation Discovery System (RFC 3401, 3402, 3403, 3761, 3764)
C. How SIP Uses ENUM

XIV. ***SIP and DHCP***

A. DHCP Protocol
B. SIP DHCP Options

XV. ***Interoperating SIP with Legacy STN Signaling***

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

XVI. ***Real-time Transport Protocol (RTP) and Real-time Control Protocol (RTCP)***

A. Dealing with Packet Loss, Latency & Jitter
B. How RTP Defines the Session
C. Session Description Protocol
D. The RTP Profile
E. The RTP Payload Type Field
F. RTP Telephony Events (RFC 2833)
G. How RTP Removes Jitter
H. How RTP Handles Packet Loss
I. How RTP Identifies the Talking Party

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Course Outline (cont.)

J. How RTP Handles Silence Suppression
K. How RTP Handles Fixed Length Packets (Padding)
L. How RTP is Used to Mix Voice (Conference Calls)
M. The RTP Header
N. RFC 2833 Protocol
O. RTP Control Protocol
P. SDES
Q. Sender/Receiver Reports
R. Bye Reports
- Labs
  - Capturing SDP offer and answer
  - Silence suppression
  - DTMF RFC 2833 and SIP INFO
  - SIP Back-to-Back UA configuration example (Asterisk)
  - REGISTER SIP device to Back-to-Back UA
  - Capture SIP call through a Back-to-Back UA and compare to a Proxy
  - RTP Relay

XVII. ***DTMF Handling***

A. Inband
B. RFC 2833
C. SIP INFO

XVIII. ***Fax Handling***

A. Inband
B. Fax Relay
C. T.38

XIX. ***Presence***

A. SIMPLE: SIP for Instant Messaging and Presence Leveraging Extensions
B. Terminology
C. Framework
D. Resource List Manipulation Requirements
E. Authorization Policy Manipulation
F. Acceptance Policy Requirements
G. Notification Requirements
H. Content Requirements
I. General Requirements

XX. ***SIP Timers***

A. T1, T2, T4
B. Timer A-K

XXI. ***SIP Security***

A. Security for Call Setup
B. Authentication
C. S/MIME
D. TLS

XXII. ***SIP NAT Traversal***

A. How NAT operates on SIP and SDP
B. NAT Types
C. STUN
D. TURN
E. ICE

XXIII. ***SIPp: A SIP Testing Tool***

A. SIPp
B. SIPp XML Examples
- Labs
  - Real-Time Control Protocol (RTCP)
  - Routing with DNS / ENUM
  - Testing Connectivity using SIP OPTIONS
  - SIP testing with SIP-p