SIP Essentials

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.

VoIP Introduction

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

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

Understanding the SIP Dialog

Day two is all about brining SIP protocol into focus. We start to refine your 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.

REGEX

- Regular Expression
- Building SIP Dialplans with REGEX

Routing the SIP INVITE

- The Via: path
- Creation of Response-Path
- Response Merging

- Record-route and Route:
- Forking
- Loops and Spirals

The SIP Dialog

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

SIP Entities

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

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 packetcapture techniques.

SIP Call Flows Examples

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

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

SIP Uniform Resource Indicators (URIs)

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

- Presence
- In Registrations

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

Session Description Protocol, Real-time Transport Protocol, and Legacy Interop
On day four, you'll 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 taught. By the end of this day, you 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.

Session Description Protocol (SDP)

- Session Parameters
- SDP Format
- Extending SDP
- SDPng

- Media Negotiation
- Changing Session Parameters
- Controlling the Media

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)

ENUM

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

SIP and DHCP

- DHCP Protocol
- SIP DHCP Options

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

Real-time Transport Protocol (RTP) and Real-time Control Protocol (RTCP)

- Dealing with Packet Loss, Latency and 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

Applications of SIP and Troubleshooting
On day five, we'll 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 how to troubleshoot SIP (common issues caused by NAT and troubleshooting with SIP-p).

DTMF Handling

- Inband
- RFC 2833
- SIP INFO

Fax Handling

- Inband
- Fax Relay
- T.38

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

SIP Timers

- T1, T2, and T4
- Timer A-K

SIP Security

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

SIP NAT Traversal

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

SIPp: A SIP Testing Tool

- SIPp
- SIPp XML Examples