SIP Advanced

CONTENTS

This course offers an in-depth exploration of the Session Initiation Protocol (SIP)—the core technology behind VoIP and modern IP-based communications. We'll go far beyond the basics, examining SIP's architecture, message flows, registration, addressing, and interactions with protocols like SDP and RTP. You'll learn how SIP enables advanced services such as call routing, mobility, presence, messaging, and security, as well as how it interoperates with legacy telephony and the IMS architecture. Along the way, we'll cover critical topics like NAT traversal, DNS-based routing, dialog management, extensibility, and troubleshooting with real-world tools. By mastering SIP's technical details and best practices, you'll be equipped to design, deploy, and troubleshoot robust, scalable, and secure communication solutions for today's networks.

TARGET AUDIENCE

This course is designed for technical professionals, engineers, and architects who work with VoIP, unified communications, or telecom networks and want to deepen their expertise in SIP. It is ideal for those who already have a basic understanding of IP networking and telephony, and who are responsible for designing, deploying, integrating, or troubleshooting SIP-based systems. The course is also valuable for developers, solution architects, and support staff working with PBXs, SBCs, IMS, or related infrastructure who require a thorough, practical understanding of advanced SIP concepts and real-world scenarios.

PREREQUISITES

No strict prerequisites are required. However, a solid technical background in IP networking or telecommunications is recommended, and prior exposure to basic SIP or VoIP concepts will help you get the most out of the course.

SIP and VoIP Basics

- Why VoIP is needed; basic QoS requirements (delay, jitter, loss)
- RTP and RTCP in real-time media transport
- Standards landscape for VoIP and SIP
- SIP origins, goals, and main components (UA, proxy, registrar, B2BUA)
- SIP message syntax, basic call flows, and the need for registration

Addressing and Registration

- SIP URIs and numbering schemes (user@domain, phone numbers, etc.)
- REGISTER and binding users to one or more contacts
- Registration lifetimes, refresh, and updates
- PBX registration models and internal addressing
- PBX example using a B2BUA

Message Forwarding and Routing

- How proxies route SIP requests and responses
- Retransmissions, timers, and loop detection
- Choosing transports (UDP/TCP/TLS) and using DNS for SIP
- Response routing rules and policy routing
- Path header and its impact on edge routing

Sessions, Calls and Dialogs

- Establishing SIP dialogs and relationship to calls/sessions
- Using SDP in the offer/answer model
- Dialog state management; early vs confirmed dialogs
- Early media and its implications
- Session modification (re-INVITE/UPDATE) and Record-Route usage

Transaction Statefulness

- Stateless vs transaction-stateful behavior in SIP servers
- Client and server transaction models
- Forking proxies and parallel branches
- Proxy response selection rules
- Transaction timers and detailed state machines

Protocol Extensions

- How SIP is extended; Require/Supported mechanisms
- Caller preferences and callee capabilities
- Session timers and session refresh
- QoS mechanisms and reliable provisional responses (PRACK)
- Using SIP over IPv6

Security

- Firewall and NAT problems for SIP signaling and media
- NAT types, traversal techniques, and signaling workarounds
- Outbound and edge proxy patterns
- Securing SIP with TLS, S/MIME, and identity (incl. STIR/SHAKEN)
- SRTP and secure media transport

Services and Applications

- Building SIP services and improving failure handling
- Call control services: REFER, 3PCC, call park/pickup, transfer, GRUU
- Presence, events, and SIP-based messaging
- Roles of B2BUA and Session Border Controller
- Conferencing, SIP applications, and SIP–WebRTC integration

Classic Telephony and SIP

- Legacy telephony services and migration to SIP
- PSTN interworking: in- and outbound connectivity
- Signalling interworking with legacy protocols
- IMS architecture at a glance
- IMS call example

Summary and Good "BYE"

- Recap of major SIP and VoIP concepts
- Typical pitfalls and best-practice design patterns
- How all components fit together in real deployments

SIP Wireshark Exercises

Exercise 1–2 – Registration

- Analyze REGISTER flows from different clients and how the registrar is discovered
- Compare registration lifetimes, multiple simultaneous contacts, and limit handling
- Observe deregistration methods and effects of very short or long expiration times

Exercise 3 – Simple Call

- Trace a basic intra-domain call: ports, codec negotiation, and media direction
- Compare normal calls with calls to unregistered or unknown users/domains
- Inspect SDP offer/answer and how unwanted media streams are rejected

Exercise 4 – Looking into the Details

- Follow an inter-domain call and how the Request-URI and ACK evolve on each hop
- Study re-INVITEs for hold/resume and changes in SDP and routing

- Analyze abnormal terminations (crashed endpoints) and resulting call flows

Exercise 5 – Forking

- See when and why proxies fork INVITEs to multiple clients
- Examine timers, branch cancellations, and handling of multiple 2xx responses
- Identify stateful proxies and differences between branches in headers and behavior

Exercise 6 – Reliable Provisional Responses

- Work with 1xx responses that require PRACK and added reliability headers
- Follow PRACK routing, early SDP, and their effect on the final 2xx and media setup
- Compare client behaviors for hold/resume and how PRACK and CANCEL interact in forked calls

Exercise 7 – Presence

- Analyze SUBSCRIBE/NOTIFY flows and event packages for presence
- Explore presence documents and how clients signal “available”, “busy”, and “DND”
- Look at subscription lifetimes and in- vs out-of-dialog SUBSCRIBE/PUBLISH behavior

Exercise 8 – Instant Messaging

- Examine MESSAGE requests and payload formats used for IM
- Compare stand-alone MESSAGEs with in-dialog messaging during an active call
- Discuss when SIP MESSAGE is appropriate vs richer messaging/file-transfer solutions

Exercise 9 – Security

- Analyze authentication for REGISTER and INVITE (401 vs 407 usage)
- Inspect how NAT traversal and SIPS/TLS/SRTP are signaled, including mandatory-SRTP cases
- Consider size/fragmentation issues from security headers and proper proxy handling

Exercise 10 – Analyzing Call Flows and Routing Logic

- Follow calls using number translation and see how many dialed numbers map to one user
- Locate where translations appear in signaling and how they change the call flow graph
- Explore transfer scenarios and multiple NOTIFYs to understand service logic and status reporting