SIP deep dive

Course code: VOICE3

In today's Unified Communications networks, SIP protocol is used whether for communication between endpoints, but also between different server systems. In this course, the participants will 'dive' into theoretical knowledge of the signaling protocol SIP. Theoretical knowledge will be tested in a number of practical exercises. This knowledge and experience is essential for all systems engineers who are dedicated for configuration of the devices that uses SIP signaling protocol. Understanding the details and functions of this signaling protocol is essential for successful implementation, configuration or troubleshooting of VoIP/SIP network. In the practical part, the participants will trace signaling in different scenarios and situations. The course is suitable for participants who do not work with Cisco Technology.

Training format

As a standard, we implement a full-time course (onsite or ILT *) in the ALEF Training Center. Upon agreement, it is possible to implement the course at the client's premises.

Due to the impossibility of preparing a remote lab and at the same time the complexity of the course, the course cannot be implemented in online / vILT form).

Explanations:

ILT - Instructor Led-Training * - instructor-led training in the classroom. ** vILT (Virtual Instructor-Led Training) - this is a form of distance learning, where the instructor conducts training from the classroom through an online platform to which students connect from their office or the comfort of their home.

Teaching materials

Participants will receive access to an electronic version of the study materials.

Course outline

- Signaling protocols SIP, H.323, MGCP, SCCP - basic overview
- Components of the network with SIP signaling
- User Agent Client
- User Agent Server
- Registrar, Location, Proxy, Redirect Server
- SIP addresses formats
- SIP messages (request / response) - content of headers and message body
- SIP dialog
- Session Description Protocol
- Registration and Authentication
- Early Offer, Offer Delayed
- Early Media
- SIP Back-to-Back User Agent - B2BUA
- Troubleshooting SIP in an environment with NAT and firewalls
- SIP Presence SIP SIMPLE
- SIP Security SIPS, TLS, SRTP
- SIP to PSTN Interworking

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