Cisco Collaboration

DTSIP: SIP Trunk Operations

$4,295.00

- 5 Days

Upcoming Dates

Feb 03 - Feb 07

Course Description

SIP Trunk Operations (DTSIP) is a 5-day instructor-led course that is intended for Cisco collaboration administrators who need to understand the features and functionality of the SIP protocol, as implemented in Cisco's Collaboration deployments.

The course begins with an examination of SIP Request and Response messages, their purpose, their meanings. We examine the Session Description Protocol (SDP) offers and answers. We explain SIP early offer and SIP early media. We also cover the purpose and configuration of Media Termination Points (MTP) and transcoders in our SIP deployments. We examine the headers that makeup all SIP messages.

Next, we examine the features and capabilities of CUCM SIP trunks. We cover the purpose of options available on the CUCM SIP Profiles that are used for trunks and line-side endpoint configurations. We will configure SIP URI dialing on CUCM. We will use ILS, GDPR, and an SME server to dynamically distribute the dial plan among multiple CUCM clusters. We configure the Cisco SIP Proxy to route enterprise calls.

We will configure Session Border Controllers (CUBES) for a variety of connective purposes. We will demonstrate how the use of E.164 Pattern Maps and Server-Groups will significantly improve and simplify the CUBE configuration. We examine the call routing logic of both inbound and outbound dial peers. We configure Voice Translation Profiles and Dial Peer Groups. We configure URI Call Routing on the CUBE and demonstrate how Provisioning Policies allow administrators to select outbound dial peers based on inbound dial-peer matching. We show you how to configure SIP Normalization on both the CUBE and CUCM, as well as how to configure the SIP OPTIONS ping keepalive feature.

In this course, we will spend an extensive amount of time Troubleshooting SIP calls. We will demonstrate many ways to collect SIP Traces and Debugs and show you how to use diagnostic programs that are available to examine and understand the various SIP debug and trace output.

Finally, you will configure a summary lab that will challenge you to use the knowledge and skills you will have learned throughout the course. You will configure an end-to-end SIP solution using multiple CUCM clusters and CUBEs. You will fulfill a list of SIP configuration requirements like what you will encounter in your real-world collaboration deployment.

Course Outline

Module 1 Examining Collaboration Solutions

Section 1: Exploring the Path to Collaboration - CLFNDU

- Describe On-Premise deployment
- Examine cloud deployments
- Examine collaboration endpoints

Module 2: Examining SIP Call Signaling and Codecs

Section 4: Exploring Codecs and Call Signaling- CLFNDU, and

Section 2: Exploring Call Signaling over IP Networks -CLCOR

- Describe SIP call signaling, voice and video codecs, RTP and RTCP
- Describe the Call Setup and Teardown Process
- Describe SIP Call Signaling for Call Setup and Teardown
- Explore Media Streams at the Application Layer
- Compare Audio Codecs
- Compare Video Codecs

Module 3: Analyzing and Troubleshooting SIP Signaling

Section 1: Analyzing and Troubleshooting Signaling and Media Protocols- CLACCM

- Analyze and troubleshoot SIP and media protocols
- Examine the characteristics and features of SIP
- SIP Trunking Considerations
- SIP Troubleshooting Tools
- Configuring SIP Traces using RTMT
- Using Wireshark and TranslatorX to read SIP debugs and traces
- Using Cisco Support Tools like CUBE DNA and Collaboration Analyzer to troubleshoot SIP calls

Module 4: Configuring Cisco SIP Trunks and Proxy

- Examine and configure SIP Proxy to route calls and CUCM SIP trunk features and capabilities
- Configuring SIP trunks to provide call routing
- Examining CUCM SIP trunk settings and understanding their purpose
- Examining CUCM SIP Profile settings and understanding their purpose
- Examining SIP Proxy Call Processing
- Configuring SIP Proxy to manage enterprise calls

Module 5 Implementing SIP URI Calling on CUCM

Section 11: Implementing URI Calling in Cisco Unified Communications Manager- CLACCM

- Implementing URI calling in CUCM for calls within a cluster and between clusters
- Provide an overview of URI call routing in CUCM
- Describe Directory URIs in CUCM
- Describe the URI call routing process in CUCM
- Describe how CUCM routes SIP URI calls to other call control systems using SIP route patterns and SIP trunks
- Describe what needs to be considered when implementing URI call routing in CUCM

Module 6: Deploying ILS and GDPR

Section 13: Examining Global Dial Plan Replication- CLACCM

- Describe how to implement ILS between CUCM clusters and enable GDPR This lesson
- Describe global dial plan issues
- Describe the characteristics of ILS and its services
- Describe the components of GDPR and their interaction
- Describe how calls are routed using GDPR
- Describe how to implement PSTN backup for intercluster calls when using GDPR

Module 7: Deploying Cisco SIP Voice Gateways

Section 9: Describing the Cisco ISR as a Voice Gateway - CLFNDU

- Describe the function, purpose, and configuration of the Cisco SIP ISR gateway
- Describe Cisco Voice Gateways
- Describe SIP gateways
- Describe Call Legs and Dial Peers
- Describe Digital Signaling Processors
- Explore the DSP Calculator

Module 8: Configuring Session Border Controllers (CUBEs)

Section 14: Configuring and Troubleshooting Cisco Unified Border Element- CLACCM

- Configure and troubleshoot Cisco Unified Border Element (CUBE)
- Describe the Cisco Unified Border Element
- Describe the call-routing logic in CUBE for numeric and URI calls
- Understand the advanced options for CUBE
- Describe how to manipulate SIP header and SDP elements in CUBE using SIP profiles
- Understand CUBE signaling and media bindings

Module 9: Configuring Additional SIP CUBE Settings

Section 8: Implementing Voice Gateways - CLCOR

- Describe how to implement digit manipulation, Early Offer, and OPTIONS on a Cisco SIP CUBE
- Configuring Voice translation profiles on CUBE
- Configuring SIP Early offer on the CUBE
- Configuring MTP on SIP Trunk to support early offer
- Configuring SIP OPTIONS keepalives on CUBE

Module 10: Configuring CUBE based URI Call Routing

- Configuring inbound URL dial-peer matching
- Configuring outbound URL dial-peer matching
- Configuring SIP Calling and Connected Party Info
- Configuring Provisioning Policies
- Normalizing SIP Messages

Module 11: Configuring the Summary Lab

- Configuring SIP trunks, CUBE, dial plan, and a variety of other settings students learned during the class
- There is a list of requirements that students will fulfill and SIP related problems that students will solve
- This lab helps students solidify concepts and demonstrates their proficiency in building SIP deployments

Audience

This course is intended for students who have general knowledge about:

- Cisco Unified Communications Manager
- Professionals with CCNA Voice and/or CCNP Voice Certification
- Customers that need to better understand the SIP protocol

Prerequisites

To fully benefit from this course, students should have the following prerequisite skills and knowledge:

- CCNA Voice or equivalent knowledge or,

- Knowledge gained from attending prerequisite courses: VFCC or ACUCM w/AUC

What You Will Learn

After this course, students will be able to:

- Examine and understand the purpose of SIP requests, responses, and SDP
- Configure SIP trunks and SIP Profiles on Cisco Unified Communication Manager (CUCM)
- Configure SIP call routing on Cisco SIP Proxy (CUSP)
- Configure URI Call routing on both CUCM and Session Border Controllers (CUBEs)
- Configure SIP CUBEs using a variety of features, including translation-profiles, patterns-maps, server groups, provision policies
- Gather SIP traces from servers, CUBEs, routers, phones, endpoints
- Diagnose and resolve SIP call routing issues, including one way audio, misconfiguration, and many other commonly encountered “real world” issues
- Configure and troubleshoot SIP throughout their collaboration enterprise