SURESKILLS Enabling The Performance Of People, Process & Technology

Implementing Cisco Unified Communications IP Telephony Part 2 (CIPT2)

Certifications

This course is part of the following Certifications:

- CCIE Voice (CCIEV Voice)
- Cisco Certified Voice Professional (CCVP)

Duration: 5 Days

Prerequisites

- Working knowledge of converged voice and data networks.
- Working knowledge of the MGCP, SIP, and H.323 and their implementation on Cisco IOS gateways.
- Ability to configure and operate Cisco routers and switches.
- Ability to configure and operate Cisco Unified Communications Manager in a single-site environment.

Course Objectives

Upon completing this course, the learner will be able to meet these overall objectives:

- Describe multisite deployment issues and solutions, and describe and configure required dial plan elements .
- Implement call processing resiliency in remote sites using SRST, MGCP fallback, and Cisco Unified * Communications Manager Express in SRST mode.
- Implement call admission control to prevent oversubscription of the IP WAN.
- Implement features and applications that are pertinent for multisite deployments.
- Secure a Cisco Unified Communications IP Telephony deployment.

Course Content

Implementing Cisco Unified Communications IP Telephony Part 2 (CIPT2) v6.0 prepares you for installing and configuring, a Cisco Unified Communications Manager solution in a multisite environment. This course focuses on Cisco Unified CallManager Release 6.0, the call routing and signaling component for the Cisco Unified Communications solution. It also includes H.323 and Media Gateway Control Protocol (MGCP) gateway implementation, the use of a Cisco Unified Border Element, and configuration of Survivable Remote Site Telephony (SRST), different mobility features, and voice security. This course includes lab activities in which you will apply a dialplan for a multisite environment, configure survivability for remote sites during WAN failure and implement solutions to reduce bandwidth requirements in the IP WAN. You will also enable Call Admission Control (CAC) and automated alternate routing (AAR), a feature that allows rerouting of calls over the public switched telephone network (PSTN) in case of no available bandwidth. There are labs for implementing device mobility, extension mobility, Cisco Unified Mobility, and voice security.