Implementing Cisco IP Telephony & Video, Part 1 (CIPTV1)

Course Summary

Description

Implementing Cisco IP Telephony & Video, Part 1 (CIPTV1) is a five-day course that prepares the learner for implementing a Cisco Collaboration solution at a single-site environment. This course focuses primarily on Cisco Unified Communications Manager Version 10.x, which is the call-routing and signaling component for the Cisco Collaboration solution.

Lab exercises included in the course help learners to perform postinstallation tasks, configure Cisco Unified Communications Manager, implement MGCP and H.323 and, SIP trunks, and build dial plans to place single site on-cluster and off-cluster calling for voice and video. Learners will also implement media resources, audio and video conferencing, and describe QoS.

Objectives

By the end of this course, students will be able to:

- Describe the role of Cisco Unified Communication Manager in a Cisco Collaboration Solution, including its functions, architecture, deployment, and redundancy options, and how to deploy endpoints, users, and Cisco IP Phone Services
- Describe the functions and purpose of a dial plan and explain how to implement on-cluster calling.
- Describe how to configure MGCP, H.323 and SIP gateways. The module also describes how to create a dial plan that supports inbound and outbound off-cluster calling for numbers and URIs.
- Describe the types of media resources that Cisco Unfied Communications Manager supports, how to configure Cisco Unified Communications Manager server software-based media resources, and how to implement Cisco hardware-based media resources
- Describe how to implement audio and video conferencing devices that can be used with Cisco Unified Communications Manager, built-in Cisco Unified Communications Manager software audio bridge, Cisco IOS-based audio and video conferencing bridges, and Cisco TelePresence conferencing products including Cisco TelePresence MSE 8000, Cisco TelePresence Server, Cisco Telepresence MCU, and Cisco TelePresence MCU, and Cisco TelePresence Conductor.
- Provide an introduction to QoS with emphasis on the QoS components, often referred to as the QoS toolkit, that are used to provide services for various business applications

Topics

- Cisco Unified Communications Manager Introduction
- Dial Plan Introduction and Implementation of Single-Site On-Cluster Calling
- Implementation of Single-Site Off-Cluster Calling
- Media Resources
- Audio and Video Conferencing
- Quality of Service

Audience

The primary target audiences for the course are:

- Network administrators and network engineers
- CCNP Collaboration candidates

The secondary audiences are:

- Systems engineers

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Course Summary (cont'd)

Prerequisites

The knowledge and skills that a learner must have before attending this course are as follows:

- Working knowledge of fundamental terms and concepts of computer networking, including LANs, WANs, switching and routing
- Ability to configure and operate Cisco routers and switches and to enable VLANs and DHCP
- Basics of digital interfaces, PSTN, and VoIP
- Fundamental knowledge of converged voice and data networks

Duration

Five days

"Charting the Course ...

... to Your Success!"

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Course Outline

I. **Cisco Unified Communications Manager Introduction**
   A. Describing the Role of Cisco Unified Communications Manager, Its Architecture, and Its Deployment and Redundancy Options
   B. Performing Initial Cisco Unified Communications Manager Configuration
   C. Deploying Endpoints and Users
   D. Deploying IP Phone Services

II. **Dial Plan Introduction and Implementation of Single-Site On-Cluster Calling**
   A. Describing Dial Plan Components
   B. Implementing Endpoint Addressing and Call Routing
   C. Implementing Calling Privileges
   D. Implementing Call Coverage in Cisco Unified Communications Manager

III. **Implementation of Single-Site Off-Cluster Calling**
   A. Analyzing Single-Site Off-Cluster Calling Requirements
   B. Implementing PSTN Access Using MGCP Gateways
   C. Describing Cisco IOS H.323 and SIP Gateways
   D. Implementing PSTN Access Using H.323 Gateways
   E. Describing the Cisco Unified Border Element
   F. Using the Cisco Unified Border Element to Access the PSTN via a SIP Trunk
   G. Using Cisco Unified Border Element for URI Dialing
   H. Describing Dial Plan Interworking

IV. **Media Resources**
   A. Describing Media Resources in Cisco Unified Communications Manager
   B. Implementing Annunciators and MOH
   C. Implementing MTPs

V. **Audio and Video Conferencing**
   A. Describing Conferencing Devices and Their Functions
   B. Implementing Conference Bridges
   C. Describing Cisco TelePresence MSE 8000
   D. Implementing Cisco TelePresence Server
   E. Implementing Cisco TelePresence Conductor

VI. **Quality of Service**
   A. Analyzing Quality of Service Requirements
   B. Describing QoS Components and their Functions

Labs:

- Hardware Lab 1: Configuring Cisco Unified Communications Manager Initial Settings
- Hardware Lab 2: Deploying Endpoints and Users
- Hardware Lab 3: Implementing Endpoint Addressing and Call Routing
- Hardware Lab 4: Implementing Calling Privileges
- Hardware Lab 5: Implementing Call Coverage
- Hardware Lab 6: Implementing PSTN Calling Using MGCP Gateways
- Hardware Lab 7: Implementing PSTN Calling Using H.323 Gateways
- Hardware Lab 8: Implementing PSTN Calling Using SIP Trunks Through Cisco Unified Border Element
- Hardware Lab 9: Using Cisco Unified Border Element for URI Dialing
- Hardware Lab 10: Implementing Annunciators and MOH
- Hardware Lab 11: Implementing Conference Bridges
- Lab 12: Implementing Cisco TelePresence Conductor