REAL WORLD
TECHNOLOGY TRAINING & SOLUTIONS
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Implementing Cisco® IP Telephony & Video Part 1 (CIPTV1) v1.0

Duration: 5 Days
Method: Instructor-Led Training (ILT) | Live Online Training

Course Description

This course prepares participants for implementing a Cisco Collaboration solution at a single-site environment. It focuses primarily on Cisco Unified Communications Manager Version 10.x, which is the call-routing and signalling component for the Cisco Collaboration solution. Lab exercises included in the course will help participants to perform post-installation 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. They will also implement media resources, audio and video conferencing, and describe Quality of Service (QoS). At the end of this course, participants will have increased and refined their Cisco video and voice knowledge and learned how to implement a Cisco Collaboration.

Target Audience

This course is intended for:

- Network Administrators and Network Engineers
- CCNP Collaboration Candidates
- Systems Engineers

Prerequisites

To attend this course, candidates must have:

- 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.

Course Objectives

Upon successful completion of this course, attendees will be able to:

- Describe the role of Cisco Unified Communications 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 the 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 Unified 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 conference bridges, and Cisco TelePresence conferencing products including Cisco TelePresence MSE 8000, Cisco TelePresence Server, 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.

Course Topics

Module 1: Cisco Unified Communications Manager Introduction

- Describing the Role of Cisco Unified Communications Manager, Its Architecture, and Its Deployment and Redundancy Options
- Performing Initial Cisco Unified Communications Manager Configuration
- Deploying Endpoints and Users
- Deploying IP Phone Services

Module 2: Dial Plan Introduction and Implementation of Single-Site On-Cluster Calling

- Describing Dial Plan Components
- Implementing Endpoint Addressing and Call Routing
- Implementing Calling Privileges
- Implementing Call Coverage in Cisco Unified Communications Manager

Course Topics *Continued*

Module 3: Implementation of Single-Site off Cluster Calling

- Analysing Single-Site Off-Cluster Calling Requirements
- Implementing PSTN Access Using MGCP Gateways
- Describing Cisco IOS H.323 and SIP Gateways
- Implementing PSTN Access Using H.323 Gateways
- Describing the Cisco Unified Border Element
- Using the Cisco Unified Border Element to Access the PSTN via a SIP Trunk
- Using Cisco Unified Border Element for URI Dialling
- Describing Dial Plan Interworking

Module 4: Media Resources

- Describing Media Resources in Cisco Unified Communications Manager
- Implementing Annunciators and MOH
- Implementing MTPs

Module 5: Audio and Video Conferencing

- Describing Conferencing Devices and Their Functions
- Implementing Conference Bridges
- Describing Cisco TelePresence MSE 8000
- Implementing Cisco TelePresence Server
- Implementing Cisco TelePresence Conductor

Module 6: Quality of Service (QoS)

- Analysing Quality of Service Requirements
- Describing QoS Components and their Functions
- Implementing Marking
- Implementing Policing and Shaping

LABS INCLUDED