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Installing and Configuring Cisco Cube/Gateways (ICCC-CT)

In this 3-day Cisco Course, students will learn how to deploy Voice Gateways/CUBE and setup Cisco Unified Communication Manager (CUCM) to deploy SIP Trunking.

The course starts out with an overview of Cisco gateways and their uses. Next, students learn about MGCP and SIP and how to implement each protocol. Students will then learn how to use Cisco CUBE to connect CUCM, Gateways and Service Providers together. This course details how to connect a Cisco environment to a Service Provider using a Cisco CUBE.

Why Attend with Current Technologies CLC

- Our Instructors are the top 10% rated by Cisco
- Our Lab has a dedicated 1 Gig Fiber Connection for our Labs
- Our Labs run up to Date Code for all our courses

Objectives

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

- Configuring Gateway Voice Ports
- SIP Protocols
- Configuring VoIP Call Legs
- Implementing a Dial Plan
- Configuring Cisco Unified Communication Manager 11.x
- Deploying Cisco VCUBE
- Implementing Cisco Unified Border Element
- High Availability on ISR G2, ISR 4k, and ASR
- Security on Cisco Unified Border Element
- Monitoring and Troubleshooting on Cisco Unified Border Element

Course Duration

3 days

Course Price

$3,895.00 or 40 CLCs

Methods of Delivery

- Instructor Led
- Virtual ILT
- On-Site

Certifications

NA

Course Exam

NA

Who Should Attend

The primary audience for this course is as follows:

- Network Video Engineer
- Voice / UC / Collaboration / Communications Engineer

- Collaboration Tools Engineer
- Collaboration Sales/Systems Engineer

Course Outline

Module 0: Introduction

- Module Introduction
  - Topic List
- Lesson 1: Introductions
  - Topic List
  - Learner Skills and Knowledge
  - Couse Goals
  - WebEx Basics
  - General Administration
  - Introductions
- Lesson 2: SIP Trunking
  - SIP Trunking Overcomes TDM Barriers
  - Why does an enterprise need an SBC?
  - Primary CUBE Differentiators (1)
- Module Summary

Module 1: Introduction to Voice Gateways

- Module Introduction
  - Topic List
- Lesson 1: Gateways
  - PSTN Access Methods
  - TDM Gateway vs. Cisco UBE
  - Gateway Functionality (1)
  - VoIP Signaling Protocols (1)
  - Gateway Deployment Example
  - Cisco Unified Communications Deployment Models
    - Single-Site Deployment (1)
    - Multisite WAN with Centralized Call Processing (1)
    - Multisite WAN with Distributed Call Processing (1)
  - Gateway Hardware Platforms
    - Cisco 4000 Series ISR Portfolio
    - Gateway Hardware - Voice Gateways
    - Voice Gateway Overview
    - Voice Gateway Call Legs
  - Cisco Unified Border Element
  - Summary
- Lesson 2: Gateway Call Routing Components
  - Inbound and Outbound Dial Peers
    - Most Prevalent Dial-Peer Types
    - POTS and VoIP Dial Peers
    - VoIP Dial Peers

        - VoIP Dial Peer Examples
    - IP and Call Routing Comparison
    - Call Routing
    - Call Legs (1)
- Lesson 3: Dial Peer Matching
    - Use of String Matching
    - String-Matching Characters
    - Number-Matching Examples
    - Matching Inbound Dial Peers (1)
    - Matching Outbound Dial Peers (1)
    - Summary
- Lesson 4: Introduction to Voice Gateways
    - Verifying Voice Ports (1)
    - Voice Codecs (1)
    - Voice Codec Packet Rates and Payload Sizes
    - Voice Quality Evaluation
    - Codec Quality
    - Evaluating Overhead (1)
    - Per-Call Bandwidth Using Common Codecs
- Lesson 5: Digital Signal Processors
    - Digital Signal Processors (1)
    - DSP Modules
    - DSP Module Comparison
- Lesson 6: Codec Complexity
    - Codec Complexity (1)
    - Packet Voice DSP Module Conferencing
- Lesson 7: Configuring DSPs
    - Configuring DSPs for Voice Termination (1)
    - Codec Complexity Configuration
    - Configuring DSP Resources for Transcoding, Conferencing, and MTP (1)
    - Transcoding and Conferencing Example
    - Verifying DSPs (1)
    - Summary
- Module Summary

Module 2: Gateway Dial Plans

- Module Introduction
    - Topic List
- Lesson 1: VoIP Overview
    - VoIP and Traditional Telephony Comparison
        - VoIP Components
        - VoIP Media Transmission Overview
    - Real-Time Transport Protocol
        - Real-Time Transport Control Protocol
        - Secure RTP
    - VoIP Media Considerations
    - Voice Activity Detection Overview

- Lesson 2: SIP Signaling Protocol
  - SIP Architecture Overview
    - SIP Signaling
    - SIP Architecture Components
    - SIP Servers
    - SIP Architecture Examples
    - SIP Direct Call Flows
- Lesson 3: SIP Addressing
  - SIP Address Types
  - Address Registration
  - Address Resolution
- Lesson 4: Codecs in SIP
  - Session Description Protocol
    - SDP Examples
    - Delayed Offer
    - Early Offer
    - Early Media (1)
- Lesson 5: Configuring Basic SIP
  - Basic SIP Configuration Overview
    - User Agent Configuration
    - Dial Peer Configuration
    - Basic SIP Configuration Example
- Lesson 6: Configuring SIP ISDN Support
  - SIP SRTP Support Overview
    - SIPS Global and Dial Peer Commands
    - SRTP Global and Dial Peer Commands
    - SIPS and SRTP Configuration Example
- Lesson 7: Customizing SIP Gateways
  - SIP Gateways Tuning Overview
    - SIP Transport
    - SIP Source IP Address and UA Timers
    - SIP UA Timers
    - SIP Early Media
  - Gateway-to-Gateway Configuration Example
  - UA Example
- Lesson 8: Verifying SIP Gateways
  - show sip-ua Command Overview
    - SIP-UA General Verification
    - SIP-UA Registration Status and Timers
    - SIP-UA Call Information
  - SIP Debugging Overview
  - Examining the INVITE Message
  - Examining the 200 OK Message
  - Examining the BYE Message
- Lesson 9: Audio Clarity Requirements
  - Audio Clarity Factors
  - Delay Sources

    - Delay Types
    - Acceptable Delay (G.114)
    - Jitter
    - Packet Loss
    - Summary of QoS Objectives
- Module Summary

Module 3: Cisco Unified Communication Manager (CUCM)

- Module Introduction
    - Topic List
- Lesson 1: Configuring CUCM for Gateways and Trunks
    - Check CUCM Version
    - Cisco UCM Audio Codec Preference List (1)
        - Audio Codec Preference List - Copy Existing
        - Audio Codec Preference List - New List
        - Audio Codec Preference List - Name and Order
    - Cisco UCM Region Configuration
    - Region Configuration - Region Relationships
    - Device Pool Configuration - Add New Device Pool (1)
    - Annunciator Configuration
    - Conference Bridge Configuration
    - Media Termination Point Configuration
    - Music on Hold Server Configuration
        - Music on Hold Service (IP Voice Media Streaming App) Parameter Settings (1)
        - Music on Hold Service (Duplex Streaming) Parameter Settings
    - Media Resource Group Configuration (1)
        - Media Resource Group List Configuration
- Lesson 2: SIP Trunk Configuration
    - SIP Trunk Security Profile Configuration (1)
    - SIP Profile Configuration (1)
    - SIP Trunk to Cisco CUBE Configuration (1)
    - Reset SIP Trunks
    - Path Selection Configuration Elements in Cisco Unified Communications Manager
    - Create Route Pattern to get to the CUBE
- Module Summary

Module 4: Configuring Cisco Unified Border Element (CUBE)

- Module Introduction
    - Topic List
- Lesson 1: What Does a Session Border Controller (SBC) Do?
    - What Is Driving SBC Adoption?
    - CUBE’s Primary Strategic Differentiators
    - Cisco Unified Border Element – Router Integration
    - CUBE Offers Architectural Choice and Flexibility

- CUBE Is Scalable
- CUBE Is Robust and Reliable
- CUBE Session Capacity Summary
- CUBE Interoperability
- CUBE Is Easy to Monitor and Manage
- SIP Trunking to Cisco WebEx
- Using CUBE in Your Contact Center Environment
- CUBE Licensing (1)
- Cisco Session Management & CUBE
- CUBE/vCUBE Deployment Scenarios
- The Centralized Model
- The Distributed Model
- The Hybrid Model

- Lesson 2: CUBE Call Flow
  - CUBE Call Processing
  - Cisco Unified Border Element Basic Call Flow
  - Transitioning to Centralized SIP Trunking...
  - Steps to transitioning...
    - Step 1: Configure CUCM to route calls to the edge SBC
    - Step 2: Get details from SIP Trunk provider
    - Step 3: Enable CUBE Application on Cisco routers
    - Step 4: Configure Call routing on CUBE
  - WAN Dial-Peer Configuration
  - LAN Dial-Peer Configuration
- Lesson 3: CUBE Dial-Peers Call Routing
  - Understanding Dial-Peer Matching Techniques
  - Understanding Inbound Dial-Peer Matching Techniques (1)
  - Understanding Outbound Dial-Peer Matching Techniques (1)
- Lesson 4: CUBE Advanced Call Routing
  - Additional Headers for Outbound Dial-Peer Matching
  - Introducing Outbound Dial-peer Provision Policy
  - Dial-peer Provision Policy Configuration (1)
  - Dial-peer Provision Policy Example – Match on FROM (1)
  - Destination Dial-peer Group
  - Destination Dial-peer Group Configuration
  - Destination Server Group
  - Multiple Destination-Patterns Under Same
  - Multiple Incoming Patterns Under Same
- Lesson 5: Media Manipulation
  - Audio Transcoding and Transrating
  - Configuration for SCCP based Transcoding
  - Configuration for LTI based Transcoding
- Lesson 6: External/PSTN Call Recording
  - External/PSTN Call Recording Options

    - CUBE Controlled Recording Option – Media Forking
    - Audio only Media Forking for an Audio/Video Call
    - CUBE Controlled Recording Option – SIPREC
- Lesson 7: Call Admission Control
    - Call Admission Control Based on Total Calls, CPU and Memory usage (1)
    - Call Admission Control at the edge
    - Call Admission Control based on Call spikes (1)
    - Call Admission Control based on Bandwidth
- Lesson 8: Multiple Non-Authenticated SIP Trunks on a CUBE
    - Non-Authenticated SIP Trunking to more than one Service Provider
- Lesson 9: Multiple Authenticated/Registered SIP Trunks on a CUBE
    - Multiple Instances of SIP-UA on a CUBE
    - Introducing Tenants on CUBE
    - “Voice class Tenant” Overview
    - Authenticating Multiple trunks with same Realm
    - Configuring Voice Class Tenant
- Module Summary

Module 5: Configuring High Availability

- Module Introduction
    - Topic List
- Lesson 1: High Availability
    - CUBE High Availability Options
        - CUBE HA Design Considerations on ISR-G2 for Box-to-Box Redundancy (1)
        - CUBE Configuration on ISR-G2 Box-to-Box Redundancy (1)
        - CUBE HA Design Considerations on ASR1K/ISR-4K/vCUBE for Box-to-Box Redundancy (1)
        - CUBE Configuration on ASR1K-ISR-4K/vCUBE Box-to-Box Redundancy (1)
    - Additional Supported Options for CUBE HA
    - ASR B2B Redundancy: PROTECTED MODE
    - CUBE SIP Trunk Monitoring with OOD Options Message (1)
    - OOD Options Ping Keepalive Enhancement (1)
    - SIP Trunk to TDM PSTN Failover
- Lesson 2: MMoH
    - Multicast MoH to Unicast MoH Conversion- CUBE
- Module Summary

Module 6: Security

- Module Introduction
    - Topic List
- Lesson 1: CUBE Security
    - Five Layers of Security in CUBE
    - Cube Security Best Practices Summary
    - Topology/Addressing Hiding

    - SIP Trunk to ITSP
    - IP Trust List for Signaling
    - Toll Fraud Mitigation
    - Configure Call Routing on CUBE
        - Understanding Dial-Peer Matching Techniques: LAN & WAN Dial-Peers
        - WAN Dial-Peer Configuration
        - LAN Dial-Peer Configuration
        - ACLS Applied on WAN Interfaces
    - SIP Listening Port Protection
    - Close Unused Session Transport Mechanisms
    - NBAR to Protect Against SIP Flooding and UDP Attacks at Opened RTP Ports
    - Firewall: General Guidelines
        - CUBE Firewall Deployment Scenarios
- Lesson 2: SIP TLS Support with SRTP
    - Secure SIP
    - SRTP Passthrough Configuration (Unsupported Crypto Suites)
    - SIP TLS/SRTP Support for Microsoft Skype for Business (Lync) Interop
- Module Summary

Module 7: Monitoring and Troubleshooting

- Module Introduction
    - Topic List
- Lesson 1: Monitoring and Troubleshooting Cisco Cube
    - Dialed Number Analyzer (DNA) for CUBE
        - Input Option - Calling/Called Number (1)
        - Input Option- SIP Message (1)
- Lesson 2: SIP Profile Test Tool
    - SIP Profile Test Tool (1)
    - CUBE Monitoring
    - Prime Collaboration (1)
        - Prime Collaboration- Assurance (1)
        - Prime Collaboration- Analytics (1)
- Lesson 3: Troubleshooting
    - Troubleshooting of Calls
        - CUBE Debugging
        - SIP Debug (1)
        - CUBE Per-Call Debugging (PCD) (1)
    - IOS Embedded Packet Capture on ISR-G2 (1)
        - Pcap
        - Export pcap
    - Call Setup Failure
        - IOS Command Output after Outbound Call
        - IOS Command Output after Inbound Call Attempt
    - Debug ccsip message
        - debug voip ccapi inout

        - debug ccsip transport   debug ip tcp transaction   debug crypto pki
        - Troubleshooting (1)
        - Verifying Certificates
- Lesson 4: Serviceability
    - New CUBE Serviceability Features
    - Call History Stats- Graphical or Tabular Form
    - Total Number of Active Concurrent Calls
    - Debugging Made Easier (1)
- Module Summary

LAB OUTLINE

Lab 0: Connect to Lab

Lab 1: Verify Cisco Unified Communications Manager Initial Settings

- Explore the Lab Environment
- Initial Default Configuration and Perform Initial Configuration

Lab 2: Deploying Endpoints and Users

Pod Mac Addresses for HQ Phone 1

- Configure CUCM
- Configure IP Phones

Lab 3: Connecting Enterprise Network to SIP Trunk Provider

- Configure SIP Trunk Security Profile
- Configure SIP Profile
- Configure SIP Trunk CUBE1
- Configure SIP Trunk CUBE2
- Configure Route Groups for Outbound Calling
- Configure a Route list for Outbound Calling
- Set up Long Distance Route Pattern
- Set up Local Route Pattern

Lab 4: Configure SIP Gateway for CUBE1

- Global CUBE Configuration
- Translation Rules and Profiles for CUBE
- CUBE Firewall Configuration
- CUBE Inbound LAN Dial-peer Configuration
- CUBE Outbound LAN Dial-peer Configuration for Publisher
- CUBE Outbound LAN Dial-peer Configuration for Subscriber
- CUBE Inbound WAN Dial-peer Configuration
- Configure SIP User Agent

- CUBE Outbound WAN Dial-peer for Long Distance Calls
- CUBE Outbound WAN Dial-peer for Local Calls
- CUBE Outbound WAN Dial-peer for International Calls

Lab 5: Configure SIP Gateway for CUBE2

- Global CUBE Configuration
- Translation Rules and Profiles for CUBE
- CUBE Firewall Configuration
- CUBE Inbound LAN Dial-peer Configuration
- CUBE Outbound LAN Dial-peer Configuration for Publisher
- CUBE Outbound LAN Dial-peer Configuration for Subscriber
- CUBE Inbound WAN Dial-peer Configuration
- Configure SIP User Agent
- CUBE Outbound WAN Dial-peer for Long Distance Calls
- CUBE Outbound WAN Dial-peer for Local Calls
- CUBE Outbound WAN Dial-peer for International Calls
- Verify and Test Cube Status

Lab 6: Configure SIP Normalization on both CUBE1 and CUBE 2

- SIP Normalization
- SIP Custom Non-Standard Header
- Troubleshooting SIP Profiles
- Remove SIP Profile

Lab 7: Configure Call Admission Control

- Configure CAC based on call spike to manage call arrival rate
- Configure CAC based on Max Connections
- Configure CAC based on different thresholds

Lab 8: Configurable SIP Error codes

- Configure Error code at the dial-peer level when total-calls are exceeded

Lab 9: Destination Dial-peer Group and Inbound SIP Profiles

- Create a new Route Pattern within your CUCM for the TAC Toll Free number

Lab 10: Multiple E164 Pattern matching under the same dial-peer

Lab 11: Destination Server Group

- Configure Destination Server Groups to deliver call legs to your respective CUCM

Lab 12: CUBE Redundancy

- Perform the Steps on Both Routers
- Configure interface tracking
- Configure Redundancy Group (RG)
- Configure the interfaces
- Challenge (optional)