Voice Foundations for Cisco Collaboration v10.5 (VFCC)

Course Summary

Description

VFCC combines elements from standard Cisco courses including CVOICE, CIPT, ICOMM and an introduction to Contact Center courses. The focus of this particular course is implementation of the 'Dial Plan' across the Voice Gateway and CUCM environment. Heavy emphasis will be placed on Call Routing within the UC environment. Little time will be spent on 'Features' of the telephony environment (for these topics consider attending CIPT1, CIPT2 or ACUCM). As an example, discussion and implementation of Gatekeepers and SIP Proxy servers are a part of VFCC, changing speed dials for your users is not (we'll assume you can figure that out on your own). This course is a lab intensive course with lots of hands-on time spent working on the equipment; our goal is to make you a better administrator by making you a better engineer. Other courses are available from SLI that focus on more basic Administrative features and functions.

VFCC is also an excellent pre-requisite to attending an advanced Contact Center Course (UCCX, DUCCE, and AUCCE 1 & 2, etc.). Students will be better prepared for advanced courses by taking VFCC and will learn more during their time in the advanced Contact Center courses if they first attend a VFCC course or have equivalent experience. This course will address the fundamentals of the Cisco Voice infrastructure including; Voice Gateways, Cisco Unified Communications Manager (CUCM) as well as the Call Signaling protocols and components used between these components (TDM, SIP, H.323, MGCP). The components and protocols discussed in this course are common to both the Contact Center Express and Enterprise environments. The goal of this course is to focus on the myriad of Trunk and Line side connections which will be used in the Contact Center environment - Express or Enterprise.

Objectives

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

- Effectively configure and utilize Unified CM Device Pools and all the accouterments which accompany them.
- Configure inbound/outbound Trunk functionality on Voice Gateway.
- Configure VoIP functionality on Ingress/Egress Gateways including SIP, H.323 and MGCP.
- Describe which Call Control Protocols are suitable for a given deployment and benefits of each.
- Configure basic SIP Proxy and/or Gatekeeper functionality.
- Configure corresponding Trunk type(s) in Unified CM including SIP, H.323 and MGCP.
- Configure digit manipulation on Voice Gateways and Unified CM.
- Implement telephony class-of-service using Calling Search Spaces and Partitions.
- Implement Media Resources (MOH, XCODE) and deploy using MRG's and MRGL's.
- Integrate Unified CM with LDAP.

Topics

- Basic Telephony Overview
- Unified Communications Manager
- On-Net Calling
- Off-Network (PSTN) Calling
- Advanced Dial Plan Considerations
- CTI Deployment Considerations

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Course Summary (con't)

Audience

This course is intended for anyone supporting the dial-plan across any of the basic telephony components of the Cisco Collaboration environments including Voice Gateways, SIP Proxy, Gatekeeper and Unified Communications Manager. It is intended for engineers who are:

- New to Voice, but not new to Data
- Not new to Voice, but new to Cisco Voice
- Need a fundamental knowledge of basic Cisco Voice architecture solutions used in a Unified Communications environment, including anyone who will be working with Contact Center Express or Enterprise.

Prerequisites

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

- Working data and/or voice background. In this course, the assumption is that you have experience in either the 'data world' or the 'telephony world', and are now being asked to gain knowledge on 'Cisco Unified Communications', which combines both worlds.
- ICND is a highly recommended prerequisite for this course if you are new to the data world

Duration

Five days

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Course Outline

I. Basic Telephony Overview
- A. Cisco Contact Center Overview
- B. PBX functionality
- C. ACD functionality
- D. IVR/VRU functionality

II. Unified Communications Manager
- A. What is a Cluster?
- B. Server Roles in the Cluster
- C. Enabling Server Services
- D. Deployment Models
- E. Redundancy Deployment
- F. CCE Resource sizing limitations
- G. A/D Integration
- H. Unified CM Navigation, tips and tricks
- I. Device Pools and all that surround them!
- J. Phones/DN's
- K. Using Auto-Registration to your advantage
- L. Got CCX? Use TAPS
- M. RTMT Overview
- N. Diagnostic Framework/RTMT-Analysis Manager for the Contact Center

III. On-Net Calling
- A. Special Purpose VXML Gateways
- B. IP vs. TDM Call Control Protocols overview
- C. SIP vs. H.323 vs. MGCP
- D. The Role of the IOS Voice Gateway
- E. CUBE Gateways
- F. DSP's for the Gateway
- G. Voice Cards for the Gateway
- H. Traditional PSTN Gateways
- I. Inter-Cluster Trunks
- J. SCCP
- K. SAF/CCD

IV. Off-Network (PSTN) Calling
- A. TDM Trunk Considerations
- B. The Importance of Binding
- C. Depth with Dial-Peers
- D. H.323
- E. Can a Gateway be a Gatekeeper?
- F. SIP
- G. MGCP
- H. Configure SIP Trunk to GW
- I. Configure H.323 Trunk to GW
- J. Consolidation of Trunks using Gatekeeper
- K. Consolidation of Trunks using SIP Proxy
- L. Configure MGCP

V. Advanced Dial Plan Considerations
- A. Numbering Plan Type Considerations
- B. Call Control Protocol Considerations
- C. Modifying the numbering plan in Unified CM
- D. Modifying the numbering plan on the Gateway
- E. Voice Translation Rules on Gateway
- F. Translating Unused DID block in Unified CM
- G. Route Groups/Route Lists
- H. What is Telephony Class of Service/Restriction
- I. Line based CSS Considerations
- J. TOD routing
- K. Gateway/Trunk Considerations
- L. Restrict inbound/outbound access using CSS/Partitions
- M. What are Media Resources?
- N. Software/Hardware Media Resources
- O. MTP's
- P. Transcoders
- Q. Conference Bridges
- R. MOH and MOH server considerations
- S. Configuring Media Resources
- T. MRG's and MRGL's

VI. CTI Deployment Considerations
- A. Call Control Considerations for CCX/CCE
- B. Dial-Plan Scale/Summarization with Gatekeeper/SIP Proxy
- C. QOS/CAC Considerations
- D. CTI Manager Service
- E. CTI Route Points
- F. AXL accounts
- G. CTI Ports
- H. Monitoring Contact Center Resources
- I. What is the Diagnostic Framework (DF)??
- J. Diagnostic Framework Portico
- K. Unified System CLI
- L. Analysis Manager via RTMT

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Course Outline (con't)

Labs:

- Lab 1-1: Explore the Lab Environment
- Lab 2-1: Initial Unified CM Configuration
- Lab 2-2: Device Pools and Initial Phone Registration
- Lab 2-3: Enable DNA, Install RTMT
- Lab 3-1: Unified CM Trunking with Intercluster Trunks (ICT's) and SIP Trunks
- Lab 3-2: Unified CM Trunking with SIP Proxy
- Lab 3-3: Service Advertisement Framework - Call Control Discovery (SAF/CCD)
- Lab 4-1: Implement MGCP Gateways
- Lab 4-2: Configure Digital Voice Interfaces (ISDN PRI T-1) for H.323 and SIP
- Lab 4-3: Implement H.323
- Lab 4-4: Implement SIP
- Lab 5-1: Digit Manipulation and Least-Cost-Routing
- Lab 5-2: Media Resources
- Lab 5-3: Calling Privileges
- Lab 6-1: Active Directory integration via LDAP and Extension Mobility