

Study Guide for the Cisco CCIE Collaboration Written Exam

Author: Jeany Lindberg

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Published by:

Network Learning Inc. (Cisco Premier Partner)

375 N. Stephanie Street, Building 21, Suite 2111

Henderson, NV 89014 USA

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Printed in the United States of America

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Author – Jeany Lindberg

Jeany Lindberg is the author, and Cisco Certified System Instructor (CCSI) 99822.

Contributing Author – Brad Ellis

Brad Ellis (CCIE #5796, CCSI #30482, CSS1, CCDP, CCNP, MCNE, and MCSE) works as a network engineer and is the CEO of Network Learning Inc. He has been dedicated to the networking industry for over 12 years. Brad has worked on large scale security assessments and infrastructure projects. He is currently focusing his efforts in the security and voice fields. Brad is a dual CCIE (R&S / Security) #5796.

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Introduction

This book is targeted toward the potential Cisco CCIE Collaboration candidates preparing for the new CCIE Collaboration Written Exam based upon Cisco Unified Communications Manager 9.0, Cisco Unified Communications Manager Express 9.0, Unity Connection 9.0, Cisco Unified Presence Server 9.0 and Cisco Unified Contact Center Express 9.0. The written guide is more than just a guide to assist you in passing the written exam (400-051), but to assist you with your career as well. This guide can also be used as a reference guide for it contains a combination of notes, white papers and Cisco technical tutorial as well classroom material from CCBOOTCAMP.

This guide also provides some sample questions that are not directly related to actual questions you will see on the exam but questions that will help you understand the topics and concepts within each chapter. These questions serve as a guide and will help you build confidence as you prepare for the CCIE Collaboration Written Exam. Some of the concepts are complex and this guide will help you understand these concepts as you prepare for the written exam as well as the CCIE Collaboration Lab practical exam.

I also recommend you read the CUCM SRND guide found on http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab09/clb09.html.

Also, visit http://www.cisco.com/web/learning/certifications/expert/ccie_collaboration/index.html for additional information for the CCIE Collaboration candidate.

Jeany Lindberg, 2014

1.00 Cisco Collaboration Infrastructure

Cisco UC Deployment Models

Deploying Unified Communications and Collaboration

From its beginnings with Voice over IP (VoIP) and IP Telephony 10 to 15 years ago, and continuing today with Unified Communications and Collaboration, users expect to be able to meet and communicate in a variety of ways using a range of devices with differing capabilities. Today, a Unified Communications and Collaboration system could start with the deployment of Jabber Clients for IM and Presence only and incrementally add voice, video, web conferencing, mobile voice applications, social media, video conferencing, and TelePresence as required. A tightly integrated Unified Communications architecture is required as the number of devices and forms of communication available to a single Unified Communications user increases. Cisco's Unified Communications and Collaboration architecture has the flexibility and scale to meet the demands of a rapidly changing and expanding Unified Communications environment that will become more URI-centric as users wish to be identified by a single user name irrespective of the form of communication.

Collaboration as a Service

As Unified Communications has become more commonplace, the deployment options for Cisco Unified Communications and Collaboration have increased to address the demand for "collaboration as a service" as well as traditional on-premises Unified Communications and Collaboration deployments. The main focus of this chapter is to provide the reader with design guidance for on-premises Unified Communications and Collaboration deployments, but it also includes a description of systems such as Cisco Hosted Collaboration Solution (HCS) and Cisco WebEx, which can be deployed as managed cloud-based Unified Communications and Collaboration services. The choice of using an on-premise, cloud-based, or hybrid solution may be determined by many factors. For example, cloud-based solutions require less on-site expertise but might lack the deployment flexibility that many enterprises need. Hybrid designs can also be deployed, where some Unified Communications functions such as call control are provided on-premises and others are provided as a cloud-based service.

Cisco offers the following "collaboration as a service" products that can be used to augment or replace functionality provided in an on-premises Unified Communications deployment:

Cisco WebEx, offering the following cloud-based services:

- Cisco WebEx Meetings
- Cisco WebEx Social
- Cisco WebEx TelePresence
- Cisco WebEx Connect, offering instant messaging (IM)

Whereas WebEx Meetings and WebEx Social are unique cloud-based services, WebEx TelePresence and instant messaging can be deployed as WebEx cloud services or as on-premises services using Cisco Unified CM, Cisco Video Communication Server (VCS), and Cisco IM and Presence.

Cisco Hosted Collaboration Solution provides the following range of applications and services:

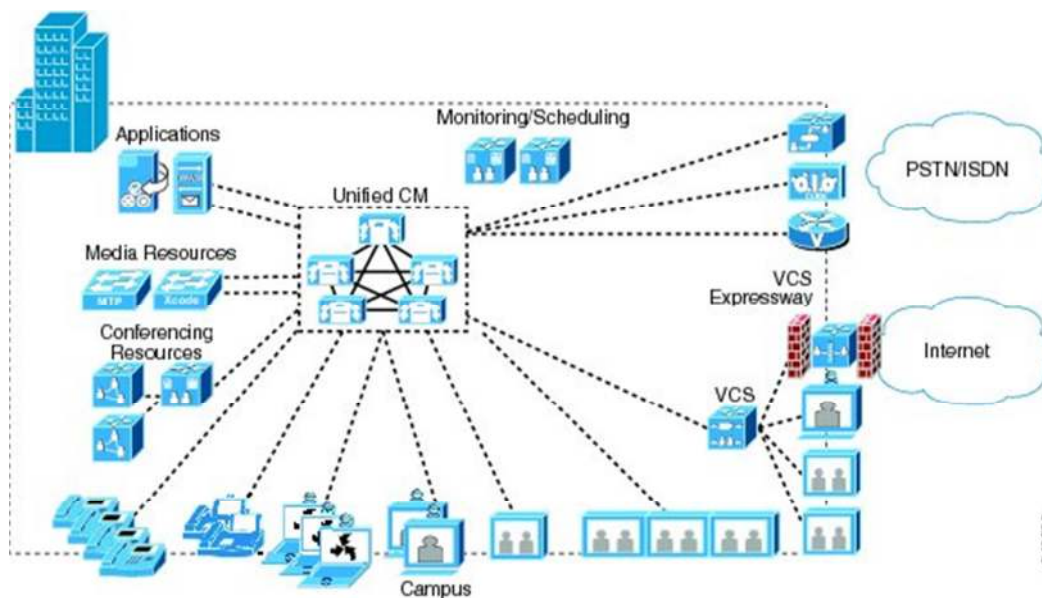
- Cisco Unified Communications Manager (Unified CM)
- Cisco Unified Attendant Consoles
- Cisco Unity Connection
- Cisco IM and Presence
- Cisco WebEx

Campus

In this call processing deployment model, the Unified Communications services and the endpoints are co-located in the campus, and the QoS-enabled network between the service nodes, the endpoints, and applications is considered highly available, offering virtually unlimited bandwidth with less than 15 ms of latency end-to-end. Likewise, the quality and availability of power are very high, and services are hosted in an appropriate data center environment. Communications between the endpoints traverses a LAN or a MAN, and communications outside the enterprise goes over an

external network such as the PSTN. An enterprise would typically deploy the campus model over a single building or over a group of buildings connected by a LAN or MAN.

Example of a Campus Deployment



The campus model typically has the following design characteristics:

- Single Cisco Unified CM cluster. Some campus call processing deployments may require more than one Unified CM cluster, for instance, if scale calls for more endpoints than can be serviced by a single cluster or if a cluster needs to be dedicated to an application such as a call center.
- Alternatively for smaller deployments, Cisco Business Edition 6000 may be deployed in the campus.
- Maximum of 40,000 configured and registered Skinny Client Control Protocol (SCCP) or Session Initiation Protocol (SIP) IP phones, video endpoints, mobile clients, and Cisco Virtualization Experience Clients (VXC) per Unified CM cluster.
- Maximum of 2,100 gateways and trunks (that is, the total number of H.323 gateways, H.323 trunks, digital MGCP devices, and SIP trunks) per Unified CM cluster.

- Trunks and/or gateways (IP or PSTN) for all calls to destinations outside the campus. Co-located digital signal processor (DSP) resources for conferencing, transcoding, and media termination point (MTP).
- Other Unified Communications services, such as messaging (voicemail), presence, and mobility are typically co-located.
- Interfaces to legacy voice services such as PBXs and voicemail systems are connected within the campus, with no operational costs associated with bandwidth or connectivity.
- SIP-based video ISDN gateways are needed to communicate with videoconferencing devices on the public ISDN network.
- Cisco TelePresence Video Communication Server (VCS) is used for room-based TelePresence conferencing systems as well as SIP, H323, and third-party TelePresence endpoints.
- Cisco TelePresence Video Communication Server Expressway (VCS Expressway) provides secure business-to-business TelePresence and video communications over the internet.
- High-bandwidth audio is available (for example, G.711 or G.722) between devices within the site.
- High-bandwidth video (for example, 384 kbps to 1.5 Mbps) is available between devices within the site.

Best Practices for the Campus Model

Follow these guidelines and best practices when implementing the single-site model:

- Ensure that the infrastructure is highly available, enabled for QoS, and configured to offer resiliency, fast convergence, and inline power.
- Know the calling patterns for your enterprise. Use the campus model if most of the calls from your enterprise are within the same site or to PSTN users outside your enterprise.
- Use G.711 codecs for all endpoints. This practice eliminates the consumption of digital signal processor (DSP) resources for transcoding, and those resources can be allocated to other functions such as conferencing and media termination points (MTPs).

- Implement the recommended network infrastructure for high availability, connectivity options for phones (in-line power), Quality of Service (QoS) mechanisms, and security.

Multisite with Centralized Call Processing

In this call processing deployment model, endpoints are remotely located from the call processing service, across a QoS-enabled Wide Area Network. Due to the limited quantity of bandwidth available across the WAN, a call admission control mechanism is required to manage the number of calls admitted on any given WAN link, to keep the load within the limits of the available bandwidth. On-net communication between the endpoints traverses either a LAN/MAN (when endpoints are located in the same site) or a WAN (when endpoints are located in different sites). Communication outside the enterprise goes over an external network such as the PSTN, through a gateway or Cisco Unified Border Element (CUBE) session border controller (SBC) that can be co-located with the endpoint or at a different location (for example, when using a centralized gateway at the main site or when doing Tail End Hop Off (TEHO) across the enterprise network).

The IP WAN also carries call control signaling between the central site and the remote sites. This graphic below illustrates a typical centralized call processing deployment, with a Unified CM cluster as the call processing agent at the central site and a QoS-enabled IP WAN to connect all the sites. In this deployment model, other Unified Communications services such as voice messaging, presence and mobility are often hosted at the central site as well to reduce the overall costs of administration and maintenance. In situations where the availability of the WAN is unreliable or when WAN bandwidth costs are high, it is possible to consider decentralizing some Unified Communications services such as voice messaging (voicemail) so that the service's availability is not impacted by WAN outages.