CISCO

CCNP Voice CIPT1 642-447

Second Edition

Quick Reference

David Bateman

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CCNP Voice CIPT1 Quick Reference

As a final exam preparation tool, the *CCNP Voice CIPT1 Quick Reference* provides a concise review of all objectives on the CIPT1 exam (642-447). This digital Quick Reference provides you with detailed, graphical-based information, highlighting only the key topics in cram-style format.

With this document as your guide, you review topics on deploying a Cisco Unified Communications Manager (CUCM) to support single site and centralized call processing models. This fact-filled Quick Reference enables you to get all-important information at a glance, helping you focus your study on areas of weakness and enhance memory retention of essential exam concepts.

About the Author

David Bateman is a Certified Cisco Systems Instructor (CCSI) with more the 20 years of internetworking experience. For more than 10 years he was a senior LAN/WAN engineer, working on both small and worldwide networks. David has always enjoyed sharing his knowledge, and in 2000, he added to his list of accomplishments by becoming a Cisco instructor for Skyline Advanced Technology Systems. In addition to teaching he is involved in author-ing courses and books including *Configuring Cisco CallManager and Unity* released by Cisco Press. He is often called on to develop and deliver specialized custom Cisco voice courses for Fortune 100 customers. David is currently the director of curriculum development for Skyline-ATS. His years of real-world technical and business knowledge enable him to bring a unique perspective to the classroom, where he not only delivers critical technical knowledge, but can also explain how technologies can be used to address various business needs.

About the Technical Editor

Alex Hannah, CCIE Voice No. 25853, is a certified Cisco instructor, specializing in teaching the Cisco Advanced IP Communications product line. He has over 7 years consulting experience in Cisco Unified Communications for SMB through Enterprise spaces. He is president of Hannah Technologies LLC, a Richmond, Virginia based Cisco consulting firm specializing in Cisco Advanced IP Communications and application development using Microsoft technologies. He holds a bachelor's degree in Information System from Virginia Commonwealth University with a minor in Business. Additionally he is the founder of UCCX.net a video based training website for the Cisco UC product line. In his spare time, you can find Alex on his boat wakeboarding with his family and friends

CHAPTER 3
Single-Site On-Net Calling

This chapter covers the configuration of CUCM for the support of on-cluster calling.

Endpoints

This section focuses on the features and characteristics of the H.323, Skinny Client Control Protocol (SCCP), and Session Initiation Protocol (SIP) endpoints that interwork with CUCM.

Endpoint Support

Cisco IP phones support SCCP and SIP and split into two categories:

- Type A phones: 7905, 7912, 7940, and 7960
- Type B phones: Most other modern Cisco phones

Cisco also has a software-based phone: the Cisco IP Communicator. Other Cisco endpoints are the Cisco Unified IP Phone 7985 (a desktop video phone), the Cisco Unified IP Phone 7920 and 7921 models (Wi-Fi phones), and the Cisco Unified IP Phone 7935 and 7936 models (conference stations). Third-party products are available for all supported protocols.

Feature Support

Use these categories to define feature support:

- **SCCP:** A Cisco proprietary protocol; offers a rich set of telephony features
- **Standard SIP or H.323:** Limited feature support when compared to SCCP
- **Cisco Unified CM SIP support for Cisco IP Phones:** Approximates the number of SCCP-supported features

Functions Found with Cisco IP Phones

- Cisco Discovery Protocol (CDP)
- DHCP
- MAC address-based device identification
- TFTP
- Power over Ethernet (PoE)
- PC port

Cisco SCCP IP Phone Startup Process

Step 1. The phone obtains power from the switch.

Step 2. The phone loads the stored image.

Step 3. The VLAN is configured using CDP.

Step 4. The phone obtains an IP address using DHCP (option 150) or is assigned statically.

Step 5. The phone requests the configuration file.

Step 6. The phone checks load file.

Step 7. The phone registers with CUCM.

Cisco SIP Phone Startup Process

The first four steps are the same as for the Cisco SCCP IP Phone startup. Then these steps occur:

Step 1. The phone boots and tries to download a control (CTL) file.

Step 2. The phone requests its SEP<mac>.cnf or the default configuration file from the Cisco TFTP server.

Step 3. The phone requests the .Loads file.

Step 4. If configured the phone downloads the dial-rule file.

Step 5. The phone registers with the highest priority CUCM server.

H.323 Phone Configuration Requirements

The H.323 phone has to be added to the CUCM with IP address and directory numbers specified. The H.323 endpoint then has to be configured to point to the IP address of the CUCM.

Following are features not supported for H.323 phones:

- MAC address registration
- Phone button templates
- Softkey templates
- Certain telephony features such as IP phone services and Cisco Unified Presence, Cisco Unified Video Advantage, Call Pickup, and Barge.

Third-Party SIP Phone Configuration Requirements

Step 1. Configure the end user in CUCM.

Step 2. Add the third-party SIP phone in CUCM, and configure its directory numbers.

Step 3. Associate the third-party SIP phone with the end user configured in Step 1.

Step 4. Configure the third-party SIP phone with the IP address of CUCM, end-user ID, digest credentials, and directory numbers.

Third-party SIP phones use digest authentication to register to the CUCM .

Features not supported for third-party SIP endpoints follow:

- MAC address registration
- Phone button templates
- Softkey templates
- Certain telephony features such as IP phone services, Cisco Unified Presence, Cisco Unified Video Advantage, Call Pickup, and Barge.

IP Phones

This section guides you through implementing SCCP and SIP phones in CMCM.

Endpoint Configuration Tools

Following are four main ways to add IP Phones to CUCM:

- Autoregistration
- CUCM Bulk Administration Tool (BAT)
- CUCM Auto-Register Phone Tool (commonly referred to as TAPS)
- Manual configuration

Phone Network Time Protocol Reference

A phone Network Time Protocol (NTP) reference can be configured in CUCM to ensure SIP phones receive accurate time and date information from an NTP server.

Date and Time Group

The date and time group defines the following:

- Time zone
- Date format
- Time format
- NTP reference

Device Pool Configuration

A device pool is associated to each device and defines various device settings. The following parameters are required:

- CM group
- Date/Time group
- Region
- Softkey template
- SRST reference

To create the device pool, use **System > Device Pool**. Then click the **Add New** button.

CM Groups

A CUCM group is a prioritized list of CUCMs, This list determines which phone attempts to register to. If the first CUCM in the list is unreachable, the device attempts to register to the second CUCM in the list. A maximum of three CUCMs are configurable per group.

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