



CCNA Voice 640-461

Official Certification Guide

Second Edition

- ✓ Master **Cisco** exam topics with the official study guides
- ✓ Assess your knowledge with **chapter-opening quizzes**
- ✓ Review key concepts with **exam preparation tasks**
- ✓ Practice with **realistic exam questions** on the CD-ROMs

CCNA Voice 640-461

Official Cert Guide

Jeremy Cioara, CCIE No. 11727

Mike Valentine, CCNA, CCNP, CCVP

Cisco Press

800 East 96th Street

Indianapolis, IN 46240

CCNA Voice 640-461 Official Cert Guide

Jeremy Cioara, CCIE No. 11727; Mike Valentine, CCNA, CCNP, CCVP

Copyright© 2012 Pearson Education, Inc.

Published by:

Cisco Press

800 East 96th Street

Indianapolis, IN 46240 USA

All rights reserved. No part of this book may be reproduced or transmitted in any form or by any means, electronic or mechanical, including photocopying, recording, or by any information storage and retrieval system, without written permission from the publisher, except for the inclusion of brief quotations in a review.

Printed in the United States of America

First Printing July 2011

Library of Congress Cataloging-in-Publication Data is on file.

ISBN-10: 1-58720-417-7

ISBN-13: 978-1-58720-417-3

Warning and Disclaimer

This book is designed to provide information about the 640-461 ICOMM exam certification exam. Every effort has been made to make this book as complete and as accurate as possible, but no warranty or fitness is implied.

The information is provided on an “as is” basis. The authors, Cisco Press, and Cisco Systems, Inc., shall have neither liability nor responsibility to any person or entity with respect to any loss or damages arising from the information contained in this book or from the use of the discs or programs that may accompany it.

The opinions expressed in this book belong to the authors and are not necessarily those of Cisco Systems, Inc.

Trademark Acknowledgments

All terms mentioned in this book that are known to be trademarks or service marks have been appropriately capitalized. Cisco Press or Cisco Systems, Inc., cannot attest to the accuracy of this information. Use of a term in this book should not be regarded as affecting the validity of any trademark or service mark.

Corporate and Government Sales

The publisher offers excellent discounts on this book when ordered in quantity for bulk purchases or special sales, which may include electronic versions and/or custom covers and content particular to your business, training goals, marketing focus, and branding interests. For more information, please contact:

U.S. Corporate and Government Sales

1-800-382-3419 corpsales@pearsontechgroup.com

For sales outside the United States, please contact:

International Sales

international@pearsoned.com

Feedback Information

At Cisco Press, our goal is to create in-depth technical books of the highest quality and value. Each book is crafted with care and precision, undergoing rigorous development that involves the unique expertise of members from the professional technical community.

Readers' feedback is a natural continuation of this process. If you have any comments regarding how we could improve the quality of this book, or otherwise alter it to better suit your needs, you can contact us through e-mail at feedback@ciscopress.com. Please make sure to include the book title and ISBN in your message.

We greatly appreciate your assistance.

Publisher: Paul Boger

Cisco Representative: Anthony Wolfenden

Executive Editor: Brett Bartow

Development Editor: Deadline Driven Publishing

Copy Editor: Sheri Cain

Editorial Assistant: Vanessa Evans

Composition: Mark Shirar

Proofreader: Water Crest Publishing

Associate Publisher: Dave Dusthimer

Cisco Press Program Manager: Jeff Brady

Managing Editor: Sandra Schroeder

Project Editor: Mandie Frank

Technical Editors: Brion Washington, John Swartz

Designer: Gary Adair

Indexer: Brad Herriman

About the Authors

Jeremy D. Cioara, CCIE No. 11727, works in many facets of the Cisco networking realm. As an author, he has written multiple books for Cisco Press and Exam Cram. As an instructor, he teaches at Interface Technical Training (www.interfacett.com) in Phoenix, Arizona. Likewise, Jeremy has recorded many E-Learning titles at CBTNuggets (www.cbtnuggets.com). Finally, Jeremy is the CIO of AdTEC Networks and works as a network consultant, focusing on Cisco network and VoIP implementations. Jeremy also casually blogs about Cisco topics at Tekcert (www.tekcert.com) in his “free time.” Thankfully, he is married to the Certified Best Wife in the World (CBWW), who helps him manage his time and priorities and prevents him from getting an enormous Cisco logo tattooed across his chest.

Michael Valentine has 15 years of experience in the IT field, specializing in network design and installation. Currently, he is a Cisco trainer with Skyline Advanced Technology Services and specializes in Cisco Unified Communications, CCNA, and CCNP classes. His accessible, humorous, and effective teaching style has demystified Cisco for hundreds of students since he began teaching in 2002. Mike holds a Bachelor of Arts degree from the University of British Columbia and currently holds CCNA, CCDA, CCNP, CCVP, and CCSI No. 31461 certifications. Mike has developed courseware and labs for Cisco and its training partners. Mike is the coauthor of *CCNA Exam Cram (Exam 640-802)*, Third Edition (Que 2008), authored the *CCNA Voice Quick Reference Guide*, and has served as technical editor and contributor on several Cisco Press titles.

About the Technical Reviewers

Brion S. Washington, CCNA, is a senior voice engineer consultant in Atlanta, GA. He has more than ten years of Cisco experience, with the last five years dedicated to VoIP; he has worked with all the Cisco VoIP products. Brion has done many large projects involving VoIP, from complete network design, implementation, and the last level of escalation. He is currently finishing up his CCVP.

John Swartz, CCIE No. 4426, is the founder of Boson Software and training, 3DSNMP, Purple Penguin and Inner Four. Currently focused on cloud technologies using the VBLOCK Infrastructure Platform by VCE.

He is also focused on mobile technology his company has published over 500 apps for the iPhone. John created the original Cisco Press CCNA Network simulator, the Boson Netsim, and numerous practice tests. He has been a Cisco instructor for 12 years, starting with basic courses and now teaching Unified Computing, Nexus switching and other data center technology. John lives in Florida with his wife and three kids.

Dedication

From Jeremy D. Cioara:

This book is dedicated to you. Yes...the person reading this *right now*. No, I'm not being cheesy, I'm serious! The only real way people are truly successful and fulfilled in this career is to love what they're doing. Because of that, I put much effort (within grammatical boundaries) into not just communicating technical mumbo jumbo—hey! Microsoft Word didn't correct that! Who knew “mumbo jumbo” was a real word?—but making it fun and interesting to read. I hope this book sparks something in you that blooms into an interesting, fun, and fulfilling career.

(In case you're curious, dictionary.com defines “mumbo jumbo” as *senseless or pretentious language, usually designed to obscure an issue, confuse a listener, or the like*. It also says that mumbo jumbo is a masked man who combats evil in the western Sudan. I don't think either of these was my intention...)

From Mike Valentine:

This book is dedicated to my wife Liana, without whose unflinching support, it might never have happened. You and me, love.

In memory of my Dad.

Acknowledgments

Jeremy D. Cioara: When you go see a movie, ever notice how the credits roll for about 5 minutes with hundreds of names? It's the same with this book. There are probably hundreds of names you'll never see that had some part in making this book possible. My thanks goes to all of them!

Personally, I give thanks to Jesus Christ who is...well, everything! Without Christ, my world of color quickly fades to a dull, boring grey. Thanks to my wife, who tirelessly homeschools our three kiddos and puts up with my countless Matrix analogies to explain anything under the sun. Finally, thanks to Interface Technical Training (www.interfacett.com), CBTNuggets (www.cbtnuggets.com), and Pearson (www.pearson.com) for allowing me to communicate my love for all things networking to people everywhere.

Mike Valentine: In fear of forgetting someone, let me try to list all the people who helped make this book happen:

Brett Bartow: For asking, answering, and adapting. Thank you, sir.

Jeremy Cioara: For trusting me with all the hard stuff...kidding, man.

Dayna, Ginny, Chris, and all the unknown soldiers at Cisco Press: They tempered, refined, redrew, and otherwise helped create what you are holding. Professionals, all; I salute them.

Toby Sauer, Dave Schulz, and Dave Bateman: My colleagues at Skyline and, most importantly, my good friends; for their opinions, their commiseration and support, and for making me a better instructor and author. Thank you, my friends. (Please go buy their books, too; you will not regret it.)

Andy de Maria: Thank you for your empathy, flexibility, and your trust.

Ed Misely: A good friend and terrifyingly capable technical resource, for his assistance with my labs.

My family: Thank you so much for your support, patience, your love, and your belief in me.

The readers and posters on the Cisco Learning Community: For your early input and support. Here it is, finally. I sincerely hope you enjoy it.

Contents at a Glance

Introduction xxv

Part I Voice Perspectives

- Chapter 1 Traditional Voice Versus Unified Voice 3
- Chapter 2 Understanding the Pieces of Cisco Unified Communications 27
- Chapter 3 Understanding the Cisco IP Phone Concepts and Registration 49

Part II Managing Endpoints and End Users

- Chapter 4 Getting Familiar with CME Administration 69
- Chapter 5 Managing Endpoints and End Users with CME 81
- Chapter 6 Understanding the CME Dial-Plan 105

Part III Handling Calls

- Chapter 7 Configuring Cisco Unified CME Voice Productivity 167
- Chapter 8 Administrator and End-User Interfaces 213

Part IV Telephony Feature Management

- Chapter 9 Managing Endpoints and End Users in CUCM 231
- Chapter 10 Understanding CUCM Dial-Plan Elements and Interactions 259

Part V Voicemail and Presence Solutions

- Chapter 11 Enabling Telephony Features with CUCM 279
- Chapter 12 Enabling Mobility Features in CUCM 313

Part VI Voice Network Management and Troubleshooting

- Chapter 13 Voicemail Integration with Cisco Unity Connection 333
- Chapter 14 Enabling Cisco Unified Presence Support 367
- Chapter 15 Common CME Management and Troubleshooting Issues 381
- Chapter 16 Management and Troubleshooting of Cisco Unified Communications Manager 399

Chapter 17	Monitoring Cisco Unity Connection	423
Chapter 18	Final Preparation	441
Appendix A	Answers Appendix	447
Appendix B	640-461 CCNA Voice Exam Updates, Version 1.0	451
	Glossary	453
	Index	463

Contents

Introduction xxv

Part I Voice Perspectives

Chapter 1 Traditional Voice Versus Unified Voice 3

- “Do I Know This Already?” Quiz 3
- Foundation Topics 6
- Where It All Began: Analog Connections 6
- The Evolution: Digital Connections 9
 - Moving from Analog to Digital 9
 - Channel Associated Signaling* 11
 - Common Channel Signaling* 12
- Understanding the PSTN 13
 - Pieces of the PSTN 13
 - Understanding PBX and Key Systems 14
 - Connections to and Between the PSTN 14
 - PSTN Numbering Plans 16
- The New Yet Not-So-New Frontier: VoIP 17
 - VoIP: Why It Is a Big Deal for Businesses 17
 - The Process of Converting Voice to Packets 18
 - Role of Digital Signal Processors 22
 - Understanding RTP and RTCP 23
- Exam Preparation Tasks 25
 - Review All the Key Topics 25
 - Definitions of Key Terms 25

Chapter 2 Understanding the Pieces of Cisco Unified Communications 27

- “Do I Know This Already?” Quiz 27
- Foundation Topics 30
- Did Someone Say Unified? 30
- Understanding Cisco Unified Communications Manager Express 31
 - CME Key Features 32
 - CME Interaction with Cisco IP Phones 32
 - A Match Made in Heaven: CME and CUE 35
- Understanding Cisco Unified Communications Manager 37
 - CUCM Key Features 37
 - CUCM Database Replication and Interacting with Cisco IP Phones 38

	Understanding Cisco Unity Connection	41
	Cisco Unity Connection Key Features	42
	Cisco Unity Connection and CUCM Interaction	43
	Understanding Cisco Unified Presence	44
	Cisco Unified Personal Communicator	45
	Exam Preparation Tasks	46
	Review All the Key Topics	46
	Definitions of Key Terms	46
Chapter 3	Understanding the Cisco IP Phone Concepts and Registration	49
	“Do I Know This Already?” Quiz	49
	Foundation Topics	52
	Connecting and Powering Cisco IP Phones	52
	Cisco Catalyst Switch PoE	54
	Powering the IP Phone Using a Power Patch Panel or Coupler	54
	Powering the IP Phone with a Power Brick	55
	VLAN Concepts and Configuration	55
	VLAN Review	55
	VLAN Trunking/Tagging	56
	Understanding Voice VLANs	58
	VLAN Configuration	59
	Understanding the Cisco IP Phone Boot Process	61
	Configuring a Router-Based DHCP Server	61
	Setting the Clock of a Cisco Device with NTP	63
	IP Phone Registration	65
	Exam Preparation Tasks	67
	Review All the Key Topics	67
	Definitions of Key Terms	67
Part II	Managing Endpoints and End Users	
Chapter 4	Getting Familiar with CME Administration	69
	“Do I Know This Already?” Quiz	69
	Foundation Topics	71
	Managing CME Using the Command Line	71
	Managing CME Using a Graphic User Interface	73
	Exam Preparation Tasks	79
	Review All the Key Topics	79
Chapter 5	Managing Endpoints and End Users with CME	81
	“Do I Know This Already?” Quiz	81

Foundation Topics	84
Ensuring the Foundation	84
Voice VLAN	85
DHCP Services	85
TFTP Services	86
Base CME Configuration	87
Ephone and Ephone-DN—The Keys to Ringing Phones	88
Understanding and Configuring Ephone-DNs	89
Understanding and Configuring Ephones	90
Associating Ephones and Ephone-DNs	92
Adding Directory Numbers, Phones, and Users with CCP	95
Exam Preparation Tasks	102
Review All the Key Topics	102

Chapter 6 Understanding the CME Dial-Plan 105

“Do I Know This Already?” Quiz	105
Foundation Topics	108
Configuring Physical Voice Port Characteristics	108
Configuring Analog Voice Ports	108
<i>Foreign Exchange Station Ports</i>	108
<i>Foreign Exchange Office Ports</i>	111
Configuring Digital Voice Ports	112
Understanding and Configuring Dial Peers	117
Voice Call Legs	118
Configuring POTS Dial Peers	120
Configuring VoIP Dial Peers	124
Using Dial Peer Wildcards	126
Private Line Automatic Ringdown	128
Understanding Router Call Processing and Digit Manipulation	130
Matching Inbound and Outbound Dial Peers	132
Using Digit Manipulation	135
<i>Practical Scenario 1: PSTN Failover Using the prefix Command</i>	136
<i>Practical Scenario 2: Directing Operator Calls to the Receptionist</i>	138
<i>Practical Scenario 3: Specific POTS Lines for Emergency Calls</i>	139
<i>Practical Scenario 4: Using Translation Profiles</i>	141
Using CCP to Configure a CME Dial-Plan	145
Understanding and Implementing CME Class of Restriction	146
Quality of Service	152

	Understanding the Enemy	153
	Requirements for Voice, Video, and Data Traffic	154
	<i>Network Requirements for Voice and Video</i>	154
	<i>Network Requirements for Data</i>	154
	Using Cisco AutoQoS	155
	Exam Preparation Tasks	163
	Review All the Key Topics	163
	Complete the Tables and Lists from Memory	164
	Definitions of Key Terms	164
Part III	Handling Calls	
Chapter 7	Configuring Cisco Unified CME Voice Productivity Features	167
	“Do I Know This Already?” Quiz	167
	Foundation Topics	171
	Configuring a Voice Network Directory	171
	Configuring Call Forwarding	175
	Forwarding Calls from the IP Phone	175
	Forwarding Calls from the CLI	177
	Using the call-forward pattern Command to Support H.450.3	177
	Configuring Call Transfer	180
	Configuring Call Park	181
	Configuring Call Pickup	186
	Configuring Intercom	189
	Configuring Paging	192
	Configuring After-Hours Call Blocking	195
	Configuring CDRs and Call Accounting	199
	Configuring Music on Hold	203
	Configuring Single Number Reach	204
	Enabling the Flash-Based CME GUI	206
	Exam Preparation Tasks	210
	Review All the Key Topics	210
	Definitions of Key Terms	210
Chapter 8	Administrator and End-User Interfaces	213
	“Do I Know This Already?” Quiz	213
	Foundation Topics	216
	Describe the CUCM GUI and CLI	216
	Cisco Unified Communications Manager Administration Interface	216

Cisco Unified Serviceability Administration Interface	217
Cisco Unified Operating System Administration Interface	219
Disaster Recovery System Interface	220
Cisco Unified Reporting Interface	220
CLI	220
User Management in CUCM: Roles and Groups	221
<i>Roles</i>	221
<i>Groups</i>	222
Describe the CUC GUI and CLI	223
<i>Cisco Unity Connection Administration</i>	224
<i>Cisco Unity Connection Serviceability</i>	225
Describe the Cisco Unified Presence Server GUI and CLI	226
<i>Cisco Unified Presence Administration Interface</i>	227
<i>Cisco Unified Presence Serviceability</i>	227
Exam Preparation Tasks	228
Review All the Key Topics	228
Definitions of Key Terms	228

Part IV Telephony Feature Management

Chapter 9 Managing Endpoints and End Users in CUCM 231

“Do I Know This Already?” Quiz	231
Foundation Topics	234
Implementing IP Phones in CUCM	234
Special Functions and Services Used by IP Phones	234
<i>NTP</i>	234
<i>CDP</i>	235
<i>DHCP</i>	235
<i>PoE</i>	235
<i>TFTP</i>	235
<i>DNS</i>	235
<i>IP Phone Registration Process</i>	236
SIP Phone Registration Process	236
Preparing CUCM to Support Phones	236
<i>Service Activation</i>	237
<i>DHCP Server Configuration</i>	237
<i>Configuring DHCP in Router IOS</i>	239
IP Phone Configuration Requirements in CUCM	240

<i>Device Pool</i>	240
<i>Device Defaults</i>	242
<i>Softkey Template and Phone Button Template</i>	242
<i>Profiles</i>	243
Describe End Users in CUCM	243
End Users Versus Application Users	243
Credential Policy	244
Features Interacting with User Accounts	244
User Locale	245
Device Association	245
Implementing End Users in CUCM	246
Manual Entry	246
Bulk Import Using BAT	246
LDAP Integration	246
<i>LDAP Synchronization</i>	248
<i>LDAP Authentication</i>	248
<i>LDAP Integration Considerations</i>	248
<i>LDAP Sync Agreements</i>	250
<i>LDAP Sync Mechanism</i>	251
<i>LDAP Custom Filters</i>	251
Configure LDAP Sync	251
Activate DirSync	251
Configure the LDAP System	252
Configure the LDAP Directory	252
Verify LDAP Sync	253
Configuring LDAP Authentication	254
Verify LDAP Authentication	254
Create LDAP Custom Filters	255
Exam Preparation Tasks	256
Review All the Key Topics	256
Chapter 10 Understanding CUCM Dial-Plan Elements and Interactions	259
“Do I Know This Already?” Quiz	259
Foundation Topics	263
CUCM Call Flows	263
Call Flow in CUCM if DNS Is Used	263
Call Flow in CUCM if DNS Is Not Used	263
Centralized Remote Branch Call Flow	265

Centralized Deployment PSTN Backup Call Flow	267
<i>Centralized Deployment Considerations and Limitations</i>	267
<i>PSTN Backup Using CAC</i>	268
Distributed Deployment Call Flow	268
Call-Routing Sources in CUCM	270
Call-Routing Destinations in CUCM	270
Call-Routing Configuration Elements	271
<i>Route Pattern</i>	271
<i>Route List</i>	271
<i>Route Group</i>	272
<i>Gateways and Trunks</i>	272
Call-Routing Behavior	273
<i>Digit Analysis</i>	273
<i>Hunt Groups</i>	274
Class of Control	274
<i>Partition</i>	275
<i>Calling Search Space</i>	275
<i>Interaction of Partitions and Calling Search Spaces</i>	275
<i>Line-Device Configuration</i>	276
Exam Preparation Tasks	277
Review All the Key Topics	277

Part V Voicemail and Presence Solutions

Chapter 11 Enabling Telephony Features with CUCM 279

“Do I Know This Already?” Quiz	279
Foundation Topics	282
Describe Extension Mobility in CUCM	282
Enable EM in CUCM	283
Step 1: Activate the EM Service	283
Step 2: Configure EM Service Parameters	283
Step 3: Add the EM Service	284
Step 4: Create Default Device Profiles	285
Step 5a: Create Device Profiles	285
Step 5b: Subscribe Device Profiles to the EM Service	286
Step 6: Associate Users with Device Profiles	287
Step 7a: Enable EM for Phones	288
Step 7b: Subscribe Phones to EM Service	289

Describe Telephony Features in CUCM	290
Call Coverage	290
<i>Call Forward</i>	291
<i>Shared Lines</i>	291
<i>Barge and Privacy</i>	291
<i>Call Pickup</i>	291
<i>Call Hunting</i>	292
<i>Call Park</i>	292
Intercom	293
CUCM Native Presence	293
<i>Presence Architecture</i>	293
Enable Telephony Features in CUCM	294
Enabling Call Coverage	295
<i>Configuring Shared Lines</i>	295
<i>Configuring Barge</i>	295
<i>Configuring Call Pickup</i>	297
<i>Configuring Call Park and Directed Call Park</i>	299
<i>Configuring Call Hunting</i>	301
Configuring Intercom Features	304
Configure CUCM Native Presence	305
<i>Configuring BLF Speed Dials</i>	305
<i>Configuring Presence-Enabled Call Lists</i>	306
<i>Configuring Custom Presence Groups</i>	307
Exam Preparation Tasks	311
Review All the Key Topics	311

Chapter 12 Enabling Mobility Features in CUCM 313

“Do I Know This Already?” Quiz	313
Foundation Topics	316
Understanding CUCM Mobility Features	316
Describe Mobile Connect	316
Unified Mobility Architecture	317
<i>Access Lists</i>	317
<i>Time-of-Day Access</i>	317
<i>Mobile Voice Access</i>	318
Implementing Mobility Features in CUCM	319
Configuring Mobile Connect	319
<i>Step 1: Configure Softkey Templates</i>	319
<i>Step 2: Configure User Accounts for Mobility</i>	319

<i>Step 3: Configure the IP Phone to Support Mobility Features</i>	320
<i>Step 4: Create Remote Destination Profiles</i>	320
<i>Step 5: Add Remote Destinations to Remote Destination Profiles</i>	321
<i>Step 6: Configure Ring Schedules for Each Remote Destination</i>	322
<i>Step 7: Configure Access Lists</i>	322
<i>Step 8: Apply Access Lists</i>	323
<i>Step 9: Configure Service Parameters</i>	325
Configuring MVA	326
<i>Step 1: Activate the MVA Service</i>	327
<i>Step 2: Configure Service Parameters</i>	327
<i>Step 3: Enable MVA for Each User</i>	328
<i>Step 4: Configure the MVA Media Resource</i>	329
<i>Step 5: Configure the MVA VXML Application at the IOS Gateway</i>	329

Exam Preparation Tasks	331
Review All the Key Topics	331
Definition of Key Terms	331

Part VI Voice Network Management and Troubleshooting

Chapter 13 Voicemail Integration with Cisco Unity Connection 333

“Do I Know This Already?” Quiz	333
Foundation Topics	336
Describe Cisco Unity Connection	336
Overview of Cisco Unity Connection	336
Single-Site and Multisite Deployment Considerations	336
CUC Integration Overview	337
CUC Integration with CUCM Using SCCP	337
CUC Integration Using SIP	338
CUC Features	338
System Settings	338
Call Handlers	340
Call Routing	340
Direct Routing Rules	341
Forwarded Routing Rules	341
Call Routing Rule Filters	341
Distribution Lists	342
Authentication Rules	342
Dial-Plan	342

Describe Cisco Unity Connection Users and Mailboxes	343
User Templates	343
<i>User Template Basics</i>	343
<i>Password Settings</i>	344
<i>Roles</i>	344
<i>Transfer Rules and Greetings</i>	344
<i>Call Actions</i>	345
<i>Message Settings, Message Actions, and Caller Input</i>	345
<i>TUI Settings</i>	345
CUC End Users	345
<i>Extension and Call Forward Options</i>	345
<i>Voice Messaging with SRST and AAR</i>	346
<i>Voicemail Box</i>	346
<i>Private Distribution Lists</i>	346
<i>Notification Devices</i>	346
User Creation Options	346
CUC Voicemail Boxes	347
<i>Message Aging Policy and Mailbox Quotas</i>	347
Implement Cisco Unity Connection Users and Mailboxes	347
Configure End User Templates	347
<i>User Template Basics</i>	348
<i>Password Settings</i>	349
<i>Roles</i>	350
<i>Message Settings</i>	350
<i>Message Actions</i>	351
<i>Phone Menu</i>	351
<i>Playback Message Settings</i>	352
<i>Notification Devices</i>	354
Configure CUC End Users	355
<i>Manual Process</i>	355
<i>Alternate Extensions and Names</i>	355
<i>Private DLs</i>	357
Importing End Users in to CUC	357
<i>Importing Users from CUCM</i>	358
<i>Importing Users from LDAP</i>	358
<i>Bulk Administration Import of CUC Users</i>	361
Managing the CUC Message Store	362
<i>Mailbox Stores Membership</i>	362

Message Aging Policy 363

Mailbox Quotas 363

Exam Preparation Tasks 365

Review All the Key Topics 365

Definitions of Key Terms 365

Chapter 14 Enabling Cisco Unified Presence Support 367

“Do I Know This Already?” Quiz 367

Foundation Topics 370

Describe Cisco Unified Presence Features 370

Cisco Unified Personal Communicator 370

CUPC Operating Modes 370

Enterprise Instant Messaging 371

Voice Calls 372

Video Calls 372

Integration Support 372

CUPC System Requirements 373

Cisco Unified Client Services Framework 373

Cisco Unified Communications Manager IP Phone Service 373

Cisco IP Phone Messenger 373

Describe Cisco Unified Presence Architecture 374

Integration with Microsoft Office Communications Server 374

Integration with LDAP 374

Integration with Cisco Unity Connection 375

Integration with Conferencing Resources 375

Integration with Calendar Resources 375

Architecture and Call Flow: Softphone Mode 376

Architecture and Call Flow: Deskphone Control Mode 376

Compliance and Persistent Chat 376

CUPS and QoS Considerations 377

Exam Preparation Tasks 379

Review All the Key Topics 379

Definitions of Key Terms 379

Chapter 15 Common CME Management and Troubleshooting Issues 381

“Do I Know This Already?” Quiz 381

Foundation Topics 384

Troubleshooting 384

Troubleshooting Common CME Registration Issues 385

Troubleshooting Dial-Plan and QoS Issues 389

Dial-Plan Issues 389

QoS Issues 392

Exam Preparation Tasks 396

Review All the Key Topics 396

Definitions of Key Terms 396

Chapter 16 Management and Troubleshooting of Cisco Unified Communications Manager 399

“Do I Know This Already?” Quiz 399

Foundation Topics 402

Describe How to Provide End-User Support for Connectivity and Voice

Quality Issues 402

Troubleshooting 402

Troubleshooting IP Phone Registration Problems 403

Deleting Unassigned Directory Numbers Using the Route
Plan Report 405

Describe CUCM Reports and How They Are Generated 406

Generating Reports 406

Analyzing Reports 407

Understanding CUCM CDR Analysis and Reporting Tool Reports 408

Activate CAR-Related Services 408

Configure CDR Service Parameters 409

CAR Tool Users 410

CDR and CMR Architecture 410

CAR System Parameters 410

Exporting CDR and CMR Records 411

Generating CDR Reports 411

Example Report Generation 412

Generating System Reports 414

Generating Device Reports 415

Describe Cisco Unified RTMT 416

RTMT Interface 416

Monitoring CUCM with RTMT 417

CallManager Summary 417

Gateway Activity 417

Device Search 417

Database Summary 417

Describe the Disaster Recovery System 418

Using the DRS 419

Set Up a Backup Device 419

Create a Scheduled Backup 419

Perform a Restore 420

Exam Preparation Tasks 421

Review All the Key Topics 421

Definitions of Key Terms 421

Chapter 17 Monitoring Cisco Unity Connection 423

“Do I Know This Already?” Quiz 423

Foundation Topics 426

Generating and Accessing Cisco Unity Connection Reports 426

Cisco Unity Connection Serviceability Reports 426

Cisco Unified Serviceability: Serviceability Reports Archive 429

Analyzing Cisco Unity Connection Reports 430

Troubleshooting and Maintenance Operations Using Cisco Unity
Connection Reports 433

Reports to Support Routine Maintenance 435

Exam Preparation Tasks 438

Review All the Key Topics 438

Definitions of Key Terms 438

Chapter 18 Final Preparation 441

Tools for Final Preparation 441

Pearson Cert Practice Test Engine and Questions on the CD 441

Install the Software from the CD 441

Activate and Download the Practice Exam 442

Activating Other Exams 442

Premium Edition 442

Cisco Learning Network 443

Chapter-Ending Review Tools 443

Suggested Plan for Final Review/Study 443

Using the Exam Engine 444

Summary 445

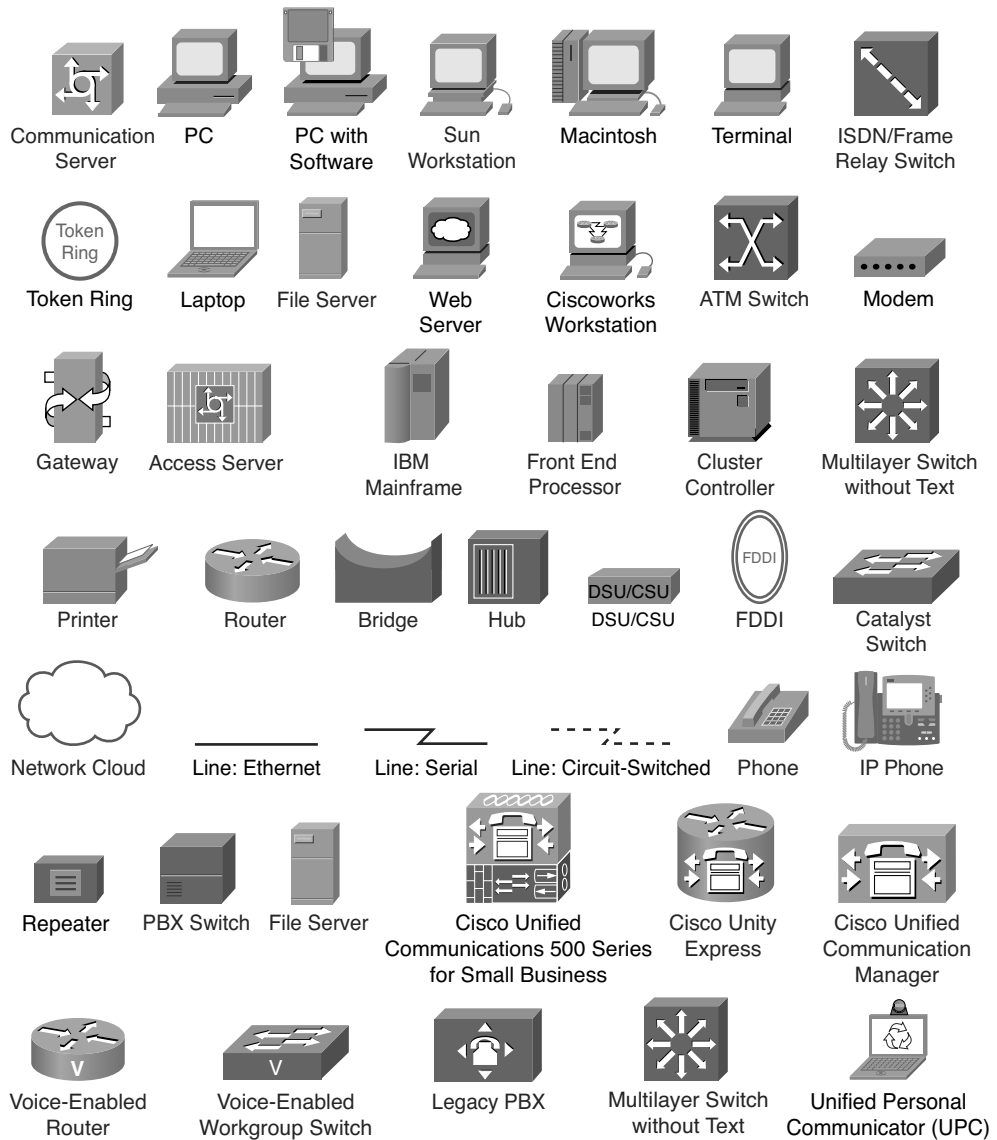
Appendix A Answers Appendix 447

Appendix B 640-461 CCNA Voice Exam Updates, Version 1.0 451

Glossary 453

Index 463

Icons Used in This Book



Command Syntax Conventions

The conventions used to present command syntax in this book are the same conventions used in the IOS Command Reference. The Command Reference describes these conventions as follows:

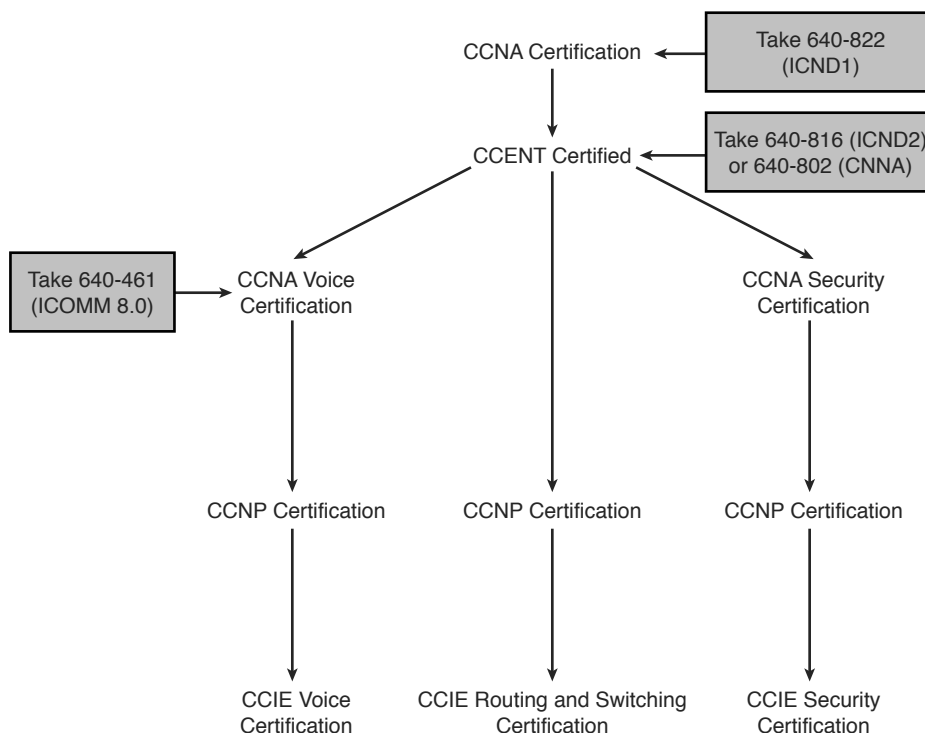
- **Boldface** indicates commands and keywords that are entered literally as shown. In actual configuration examples and output (not general command syntax), boldface indicates commands that are manually input by the user (such as a **show** command).
- *Italics* indicate arguments for which you supply actual values.
- Vertical bars (|) separate alternative, mutually exclusive elements.
- Square brackets [] indicate optional elements.
- Braces { } indicate a required choice.
- Braces within brackets [{ }] indicate a required choice within an optional element.

Introduction

Welcome to the world of CCNA Voice! As technology continues to evolve, the realm of voice, which was traditionally kept completely separate from data, has now begun to merge with the data network. This brings together two different worlds of people: data technicians—historically accustomed to working with routers, switches, servers, and the like—and voice technicians, historically accustomed to working with PBX systems, digital handsets, and trunk lines. Regardless of your background, one of the primary goals of the new CCNA Voice certification is to bridge these two worlds together.

In June 2008, Cisco announced new CCNA specialties, including CCNA Security, CCNA Wireless, and CCNA Voice. These certifications, released ten years after the initial CCNA, represent Cisco's growth into new and emerging industries. Certification candidates can now specialize in specific areas of study. Figure I-1 shows the basic organization of the certifications and exams used to achieve your CCNA Voice certification.

Figure I-1 *Cisco Certifications and CCNA Voice Certification Path*



As you can see from Figure I-1, a traditional CCNA certification is a prerequisite before you venture into the CCNA Voice certification.

Goals and Methods

The most important and somewhat obvious goal of this book is to help you pass the Implementing Introducing Cisco Voice and Unified Communications Administration v8.0 (ICOMM 8.0) exam (640-461). In fact, if the primary objective of this book were different, the book's title would be misleading. The methods used in this book help you pass the ICOMM 8.0 exam and make you much more knowledgeable about how to do your job.

This book uses several key methodologies to help you discover the exam topics that you need to review in more depth, to help you fully understand and remember those details, and to help you prove to yourself that you have retained your knowledge of those topics. So, this book does not try to help you pass by memorization, but helps you truly learn and understand the topics. The CCNA Voice exam is the foundation for many of the Cisco professional certifications, and it would be a disservice to you if this book did not help you truly learn the material. Therefore, this book helps you pass the CCNA Voice exam by using the following methods:

- Helping you discover which test topics you have not mastered
- Providing explanations and information to fill in your knowledge gaps
- Supplying exercises and scenarios that enhance your ability to recall and deduce the answers to test questions
- Providing practice exercises on the topics and the testing process via test questions on the CD-ROM

In addition, this book uses a different style from typical certification-preparation books. The newer Cisco certification exams have adopted a style of testing that essentially says, "If you don't know how to do it, you won't pass this exam." This means that most of the questions on the certification exam require you to deduce the answer through reasoning or configuration rather than just memorizing facts, figures, or syntax from a book. To accommodate this newer testing style, the authors have written this book as a real-world explanation of Cisco VoIP topics. Most concepts are explained using real-world examples rather than showing tables full of syntax options and explanations, which are freely available on Cisco.com. As you read this book, you definitely get a feeling of, "This is how I can *do* this," which is exactly what you need for the newer Cisco exams.

Who Should Read This Book?

The purpose of this book is twofold. The primary purpose is to tremendously increase your chances of passing the CCNA Voice certification exam. The secondary purpose is to provide the information necessary to manage a VoIP solution using Cisco Unified Communication Manager Express (CME), Cisco Unified Communications Manager (CUCM), Cisco Unity Connection, or Cisco Unified Presence. Cisco's new exam approach provides an avenue to write the book with both a real-world and certification-study approach at the same time. As you read this book and study the configuration examples and exam tips, you have a true sense of understanding how you could deploy a VoIP system, while at the same time feeling equipped to pass the CCNA Voice certification exam.

Strategies for Exam Preparation

Strategies for exam preparation will vary depending on your existing skills, knowledge, and equipment available. Of course, the ideal exam preparation would consist of building a small voice lab with a Cisco Integrated Services Router, virtualized lab versions of CUCM, Unity Connection, and Presence servers, a switch, and a few IP Phones, which you could then use to work through the configurations as you read this book. However, not everyone has access to this equipment, so the next best step you can take is to read the chapters and jot down notes with key concepts or configurations on a separate notepad. Each chapter begins with a “Do I Know This Already?” quiz, which is designed to give you a good idea of the chapter’s content and your current understanding of it. In some cases, you might already know most of or all the information covered in a given chapter.

After you read the book, look at the current exam objectives for the CCNA Voice exam listed on Cisco.com (www.cisco.com/certification). If there are any areas shown in the certification exam outline that you would still like to study, find those sections in the book and review them.

When you feel confident in your skills, attempt the practice exam included on the CD with this book. As you work through the practice exam, note the areas where you lack confidence and review those concepts or configurations in the book. After you have reviewed the areas, work through the practice exam a second time and rate your skills. Keep in mind that the more you work through the practice exam, the more familiar the questions will become, so the practice exam will become a less accurate judge of your skills.

After you work through the practice exam a second time and feel confident with your skills, schedule the real ICOMM 8.0 (640-461) exam through Vue (www.vue.com). You should typically take the exam within a week from when you consider yourself ready to take the exam, so that the information is fresh in your mind.

Keep in mind that Cisco exams are very difficult. Even if you have a solid grasp of the information, many other factors play into the testing environment (stress, time constraints, and so on). If you pass the exam on the first attempt, fantastic! If not, know that this commonly happens. The next time you attempt the exam, you will have a major advantage: You already experienced the exam first-hand. Although future exams may have different questions, the topics and general “feel” of the exam remain the same. Take some time to study areas from the book where you felt weak on the exam. Retaking the exam the same or following day from your first attempt is a little aggressive; instead, schedule to retake it within a week, while you are still familiar with the content.

640-461 ICOMM 8.0 Exam Topics

Table I-1 lists the exam topics for the 640-461 ICOMM 8.0 exam. This table also lists the book parts in which each exam topic is covered.

Table I-1 640-461 ICOMM 8.0 Exam Topics**Chapter Where Topic Is Covered Exam Topic****Describe the characteristics of a Cisco Unified Communications solution**

Chapter 2	Describe the Cisco Unified Communications components and their functions
-----------	--

Chapter 2	Describe call signaling and media flows
-----------	---

Chapter 6	Describe quality implications of a VoIP network
-----------	---

Provision end users and associated devices

Chapter 5, Chapter 9	Describe user creation options for Cisco Unified Communications Manager and Cisco Unified Communications Manager Express
----------------------	--

Chapter 9	Create or modify user accounts for Cisco Unified Communications Manager
-----------	---

Chapter 5	Create or modify user accounts for Cisco Unified Communications Manager Express using the GUI
-----------	---

Chapter 9	Create or modify endpoints for Cisco Unified Communications Manager
-----------	---

Chapter 5	Create or modify endpoints for Cisco Unified Communications Manager Express using the GUI
-----------	---

Chapter 6, Chapter 10	Describe how calling privileges function and how calling privileges impact system features
-----------------------	--

Chapter 5, Chapter 9	Create or modify directory numbers
----------------------	------------------------------------

Chapter 7, Chapter 11, Chapter 12	Enable user features and related calling privileges for extension mobility, call coverage, intercom, native presence, and unified mobility remote destination configuration
-----------------------------------	---

Chapter 14	Enable end users for Cisco Unified Presence
------------	---

Chapter 7, Chapter 11, Chapter 12	Verify user features are operational
-----------------------------------	--------------------------------------

Configure voice messaging and presence

Chapter 13	Describe user creation options for voice messaging
------------	--

Chapter 13	Create or modify user accounts for Cisco Unity Connection
------------	---

Chapter 14	Describe Cisco Unified Presence
------------	---------------------------------

Chapter 14	Configure Cisco Unified Presence
------------	----------------------------------

Table I-1 640-461 ICOMM 8.0 Exam Topics**Chapter Where Topic Is Covered Exam Topic****Maintain Cisco Unified Communications system**

Chapter 16	Generate CDR and CMR reports
Chapter 16	Generate capacity reports
Chapter 16	Generate usage reports
Chapter 16	Generate RTMT reports to monitor system activities
Chapter 17	Monitor voicemail usage
Chapter 16	Remove unassigned directory numbers
Chapter 16	Perform manual system backup

Provide end user support

Chapter 15, Chapter 16	Verify PSTN connectivity
Chapter 15, Chapter 16	Define fault domains using information gathered from end user
Chapter 15, Chapter 16	Troubleshoot endpoint issues
Chapter 17	Identify voicemail issues and resolve issues related to user mailboxes
Chapter 15, Chapter 16	Describe causes and symptoms of call quality issues
Chapter 5, Chapter 9	Reset single devices
Chapter 11	Describe how to use phone applications

How This Book Is Organized

Although this book could be read cover-to-cover, it is designed to be flexible and allow you to easily move between chapters and sections of chapters to cover just the material that you need more work with. If you do intend to read all the chapters, the order in the book is an excellent sequence to use.

The core chapters, Chapters 1 through 17, cover the following topics:

- **Chapter 1, “Traditional Voice Versus Unified Voice.”** This chapter discusses what would be known as the traditional telephony world. It begins where the telephone system originally started: analog connectivity. It then moves into the realm of digital connections and considerations and concludes the traditional voice discussion with the primary pieces that you need to know from the public switched telephone network (PSTN). Chapter 1 then moves into the unified voice realm, discussing the benefits of VoIP, the process of coding and decoding audio, digital signal processors (DSP), and the core VoIP protocols.

- **Chapter 2, “Understanding the Pieces of Cisco Unified Communications.”** This chapter primarily focuses on the components of a Cisco VoIP network. By breaking down the voice infrastructure into four distinct areas, each component can be categorized and described. These components include endpoints, call processing agents, applications, and network infrastructure devices.
- **Chapter 3, “Understanding the Cisco IP Phone Concepts and Registration.”** This chapter discusses the preparation and base configuration of the LAN infrastructure to support VoIP devices. This preparation includes support for Power over Ethernet (PoE), voice VLANs, a properly configured DHCP scope for VoIP devices, and the Network Time Protocol (NTP).
- **Chapter 4, “Getting Familiar with CME Administration.”** This chapter familiarizes you with Cisco Unified Communication Manager Express (CME) administration by unpacking the two primary administrative interfaces of CME: command-line and the Cisco Configuration Professional (CCP) GUI.
- **Chapter 5, “Managing Endpoint and End Users with CME.”** This chapter focuses on the process to create and assign directory numbers (DN) and user accounts to Cisco IP Phones. The chapter walks through these configurations in both the command-line and CCP interfaces.
- **Chapter 6, “Understanding the CME Dial-Plan.”** Now that the internal VoIP network is operational through the CME configuration, this chapter examines connections to the outside world through the PSTN or over an IP network. Concepts covered in this chapter include the configuration of physical voice port characteristics, dial peers, digit manipulation, class of restriction (COR), and quality of service (QoS).
- **Chapter 7, “Configuring Cisco Unified CME Voice Productivity Features.”** This chapter examines feature after feature supported by the CME router. By the time you’re done with this chapter, you’ll understand how to configure features such as intercom, paging, Call Park and pickup, and many others.
- **Chapter 8, “Administrator and End-User Interfaces.”** This chapter introduces the administration interfaces for CUCM, CUC, and CUP. From the administrative GUI for each application to the common Unified Serviceability interface, disaster recovery, and CLI, the fundamentals of navigation and configuration are laid out in a clear and logical sequence.
- **Chapter 9, “Managing Endpoints and End Users in CUCM.”** The configuration and management of users and phones is covered in this chapter, including integration with LDAP.
- **Chapter 10, “Understanding CUCM Dial-Plan Elements and Interactions.”** The guts of the call-routing system in CUCM are explained with simplicity and clarity. Call flows in different deployments and under different conditions of use and failure (including CAC and AAR) are demonstrated and compared, and the great mystery of partitions and calling search spaces (CSS) is revealed for the simple truth it really is.

- **Chapter 11, “Enabling Telephony Features with CUCM.”** A small but excellent sample of the billions* (*approximately) of features available in CUCM, including Extension Mobility and call coverage.
- **Chapter 12, “Enabling Mobility Features in CUCM.”** A step-by-step guide to enabling some of the most popular and powerful features in CUCM: Mobile Connect and Mobile Voice Access.
- **Chapter 13, “Voicemail Integration with Cisco Unity Connection.”** The power, stability and wealth of features available in CUC are examined, followed by a look at the configuration of user accounts and their mail boxes.
- **Chapter 14, “Enabling Cisco Unified Presence Support.”** The capabilities, features, and basic configuration of the CUP server and clients are covered, giving an introduction to one of the most powerful additions to the Unified Communications capabilities of any business.
- **Chapter 15, “Common CME Management and Troubleshooting Issues.”** This chapter takes the CME concepts you learned and builds them into troubleshooting scenarios. The chapter begins by discussing a general troubleshooting process you can employ for any technical troubleshooting situation, then walks through many common CME troubleshooting situations dealing with IP phone registration. The chapter concludes by discussing dial-plan and QoS troubleshooting methods.
- **Chapter 16, “Management and Troubleshooting of Cisco Unified Communications Manager.”** This chapter reviews the tools available to administrators to assist in the care and feeding of their CUCM servers. From the myriad of built-in reporting tools to the power of the RTMT, the administrator is introduced to his arsenal of tools to monitor the health and performance of the system.
- **Chapter 17, “Monitoring Cisco Unity Connection.”** The wealth of built-in reporting and monitoring tools for CUC are reviewed in this chapter.

In addition to the 17 main chapters, this book includes tools to help you verify that you are prepared to take the exam. Chapter 18, “Final Preparation,” includes guidelines that you can follow in the final days before the exam. Also, the CD-ROM includes quiz questions and memory tables that you can work through to verify your knowledge of the subject matter.



This chapter includes the following topics:

- **Connecting and Powering Cisco IP Phones:** To provide a centralized power system, the Cisco IP Phones must receive their power from a centralized source using PoE. This section discusses the different options for PoE and the selection criterion of each.
- **VLAN Concepts and Configuration:** VLANs allow you to break the switched network into logical pieces to provide management and security boundaries between the voice and data network. This section discusses the concepts and configuration behind VLAN.
- **Understanding Cisco IP Phone Boot Process:** This section discusses the foundations of the Cisco IP Phone boot process. Understanding this process is critical to troubleshooting issues with the IP Telephony system.
- **Configuring a Router-Based DHCP Server:** This section discusses configuring a Cisco router as a DHCP server for your network.
- **Setting the Clock of a Cisco Device with NTP:** Because a VoIP network heavily depends on accurate time, the sole focus of this section is keeping the clocks accurate on Cisco devices by using NTP.
- **IP Phone Registration:** Once the Cisco IP Phone receives all its network configuration settings, it is ready to speak to a call processing agent. This section describes the process and protocols that make it happen.

Understanding the Cisco IP Phone Concepts and Registration

You walk into the new corporate headquarters for Fizzmo Corp. On the top of each desk is a Cisco 7945G IP Phone, glowing with a full-color display and two line instances. Smiling, courteous agents are busy taking phone calls from callers excited to purchase the latest Fizzmo wares. Samantha (located in the north corner) is checking her visual voicemail, while Emilio (located in the south hall) is getting the latest weather report through an XML IP phone service.

How did we get here? How do you take a newly constructed building and transform it into a bustling call center? That's what this chapter is all about. We walk through the key concepts and technologies used to build a Cisco VoIP network. By the time you are done with this chapter, you will have all the conceptual knowledge you need to have in place before you can move into the installation and configuration of the Cisco VoIP system.

“Do I Know This Already?” Quiz

The “Do I Know This Already?” quiz allows you to assess whether you should read this entire chapter or simply jump to the “Exam Preparation Tasks” section for review. If you are in doubt, read the entire chapter. Table 3-1 outlines the major headings in this chapter and the corresponding “Do I Know This Already?” quiz questions. You can find the answers in Appendix A, “Answers Appendix.”

Table 3-1 “Do I Know This Already?” Foundation Topics Section-to-Question Mapping

Foundation Topics Section	Questions Covered in This Section
Connecting and Powering Cisco IP Phones	1–2
VLAN Concepts and Configuration	3–8
Understanding Cisco IP Phone Boot Process	9
Configuring a Router-Based DHCP Server	10
Setting the Clock of a Cisco Device with NTP	11
IP Phone Registration	12

1. Which of the following is an industry standard used for powering devices using an Ethernet cable?
 - a. Cisco Inline Power
 - b. 802.1Q
 - c. 802.3af
 - d. Local power brick
2. Which of the following are valid methods for powering a Cisco IP Phone? (Select all that apply.)
 - a. Power brick
 - b. Crossover coupler
 - c. PoE
 - d. Using pins 1, 2, 3, and 4
3. Which of the following terms are synonymous with a VLAN? (Choose two.)
 - a. IP subnet
 - b. Port security
 - c. Broadcast domain
 - d. Collision domain
4. Which of the following trunking protocols would be used to connect a Cisco switch to a non-Cisco switch device?
 - a. VTP
 - b. 802.3af
 - c. 802.1Q
 - d. ISL
5. How should you configure a port supporting voice and data VLANs that is connected to a Cisco IP Phone?
 - a. Access
 - b. Trunk
 - c. Dynamic
 - d. Dynamic Desired
6. How does a device attached to a Cisco IP Phone send data to the switch?
 - a. As tagged (using the voice VLAN)
 - b. As untagged
 - c. As tagged (using the data VLAN)
 - d. As tagged (using the CoS value)

7. Which of the following commands should you use to configure a port for a voice VLAN 12?
 - a. switchport mode voice vlan 12
 - b. switchport trunk voice vlan 12
 - c. switchport voice vlan 12
 - d. switchport vlan 12 voice
8. Which of the following commands would you use to forward DHCP requests from an interface connected to the 172.16.1.0/24 subnet to a DHCP server with the IP address 172.16.100.100?
 - a. forward-protocol 172.16.1.0 255.255.255.0 172.16.100.100
 - b. forward-protocol dhcp 172.16.1.0 255.255.255.0 172.16.100.100
 - c. ip helper-address 172.16.1.0 172.16.100.100
 - d. ip helper-address 172.16.100.100
9. How does the Cisco switch communicate voice VLAN information after a Cisco IP Phone has received PoE and started the boot process?
 - a. Through CDP
 - b. Using 802.1Q
 - c. Using the proprietary ISL protocol
 - d. Voice VLAN information must be statically entered on the Cisco IP Phone.
10. Which DHCP option provides the IP address of a TFTP server to a Cisco IP Phone?
 - a. Option 10
 - b. Option 15
 - c. Option 150
 - d. Option 290
11. Which of the following NTP stratum numbers would be considered the best?
 - a. Stratum 0
 - b. Stratum 1
 - c. Stratum 2
 - d. Stratum 3
12. Which of the following protocols could be used for Cisco IP Phone registration? (Choose two.)
 - a. SCCP
 - b. SIP
 - c. DHCP
 - d. H.323

Foundation Topics

Connecting and Powering Cisco IP Phones

Before we can get to the point of plugging in phones and having happy users placing and receiving calls, we must first lay the foundational infrastructure of the network. This includes technologies such as Power over Ethernet (PoE), voice VLANs, and Dynamic Host Configuration Protocol (DHCP). The network diagram shown in Figure 3-1 represents the placement of these technologies. As you read this chapter, each section will act as a building block to reach this goal. The first item that must be in place is power for the Cisco IP Phones.

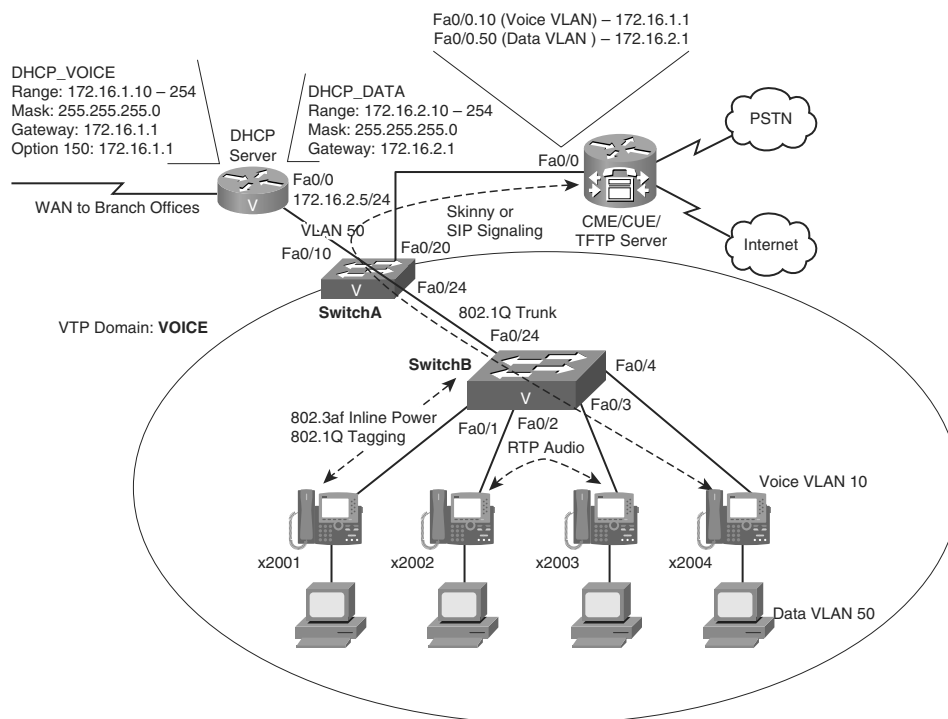


Figure 3-1 VoIP Network

Cisco IP Phones connect to switches just like any other network device (such as PCs, IP-based printers, and so on). Depending on the model of IP phone you are using, it may also have a built-in switch. Figure 3-2 illustrates the connections on the back of a Cisco 7960 IP Phone.

The ports shown in Figure 3-2 are as follows:

- **RS232**: Connects to a expansion module (such as a 7914, 7915, or 7916)
- **10/100 SW**: Used to connect the IP phone to the network
- **10/100 PC**: Used to connect a co-located PC (or other network device) to the IP Phone

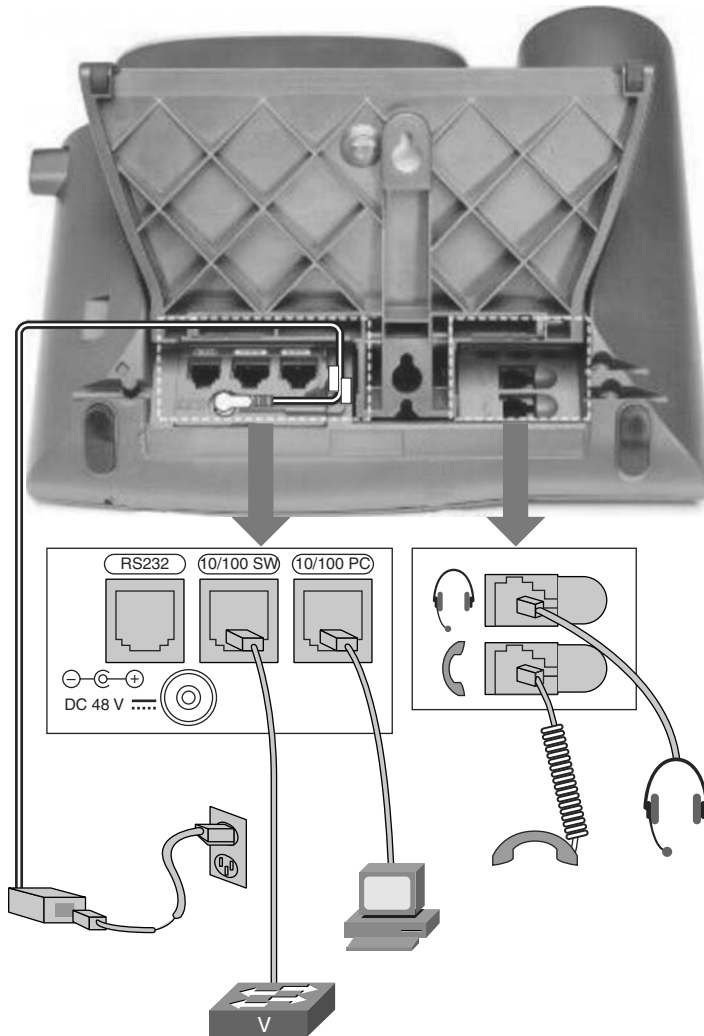


Figure 3-2 *Cisco IP Phone Ethernet Connections*

After you physically connect the IP phone to the network, it needs to receive power in some way. There are three potential sources of power in a Cisco VoIP network:

- Cisco Catalyst Switch PoE (Cisco prestandard or 802.3af power)
- Power Patch Panel PoE (Cisco prestandard or 802.3af power)
- Cisco IP Phone Power Brick (wall power)

Let's dig deeper into each one of these power sources.

Cisco Catalyst Switch PoE

If you were to create an Ethernet cable (Category 5 or 6), you would find that there are eight wires (four pairs of wires) to crimp into an RJ-45 connector on each end of the connection. Further study reveals that only four of the wires are used to transmit data. The other four remain unused and idle...until now.

The terms inline power and PoE describe two methods you can use to send electricity over the unused Ethernet wires to power a connected device. There is now a variety of devices that can attach solely to an Ethernet cable and receive all the power they need to operate. In addition to Cisco IP Phones, other common PoE devices include wireless access points and video surveillance equipment.

Powering devices through an Ethernet cable offers many advantages over using a local power supply. First, you have a centralized point of power distribution. Many users expect the phone system to continue to work even if the power is out in the company offices. By using PoE, you can connect the switch powering the IP phones to an uninterruptible power supply (UPS) instead of placing a UPS at the location of each IP phone. PoE also enables you to power devices that are not conveniently located next to a power outlet. For example, it is a common practice to mount wireless access points in the ceiling, where power is not easily accessible. Finally, PoE eliminates much of the “cord clutter” at employees’ desks.

PoE became an official standard (802.3af) in 2003. However, the IP telephony industry was quickly developing long before this. To power the IP phones without an official PoE standard, some proprietary methods were created, one such method being Cisco Inline Power.

Note: The IEEE standards body has recently created the 802.3at PoE standard (also called PoE Plus), the goal of which is to increase the current maximum PoE wattage from 15.4W to 25.5W. In addition, some proprietary implementations of PoE have reached 51W of power by using all four pairs of wire in the Ethernet cable.

Powering the IP Phone Using a Power Patch Panel or Coupler

Many companies already have a significant investment in their switched network. To upgrade all switches to support PoE would be a significant expense. These organizations may choose to install intermediary devices, such as a patch panel, that are able to inject PoE on the line. The physical layout for this design is demonstrated in Figure 3-3.

By using the power patch panel, you still gain the advantage of centralized power and backup without requiring switch upgrades.

Note: Keep in mind that Cisco switches must also provide quality of service (QoS) and voice VLAN support capabilities, which may require switch hardware upgrades. Be sure your switch supports these features before you consider a power patch panel solution.

Inline PoE injectors provide a low-cost PoE solution for single devices (one device per coupler). These are typically used to support wireless access points or other “single spot” PoE solutions. Using inline PoE couplers for a large IP Phone network would make a mess

of your wiring infrastructure and exhaust your supply of electrical outlets (because each inline PoE coupler requires a dedicated plug).

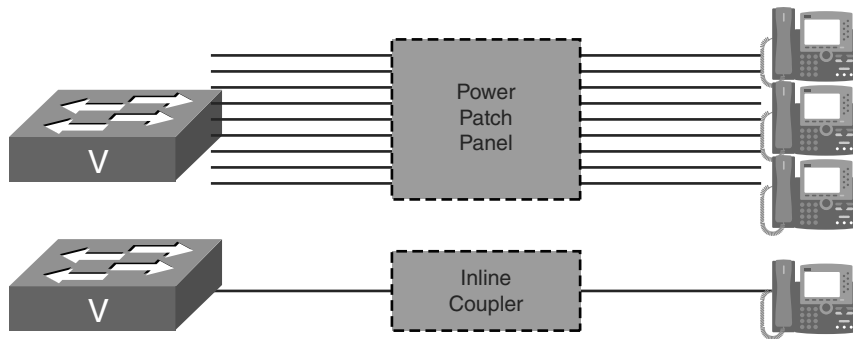


Figure 3-3 *Design for Power Patch Panels or Inline Couplers*

Powering the IP Phone with a Power Brick

Using a power brick to power a device is so simple that it warrants only brief mention. Thus, the reason for this section is primarily to mention that most Cisco IP Phones do not ship with power supplies. Cisco assumes most VoIP network deployments use PoE. If you have to choose between purchasing power bricks and upgrading your switch infrastructure, it's wise to check the prices of the power bricks. The average Cisco IP Phone power brick price is between \$30–\$40 USD. When pricing out a 48-switchport deployment, purchasing power bricks for all the IP phones may very well be in the same price range as upgrading the switch infrastructure.

Note: Some devices exceed the power capabilities of the 802.3af PoE standard. For example, when you add a sidcar module to a Cisco IP Phone (typically to support more line buttons), PoE connections can no longer support the device. These devices will need a power brick adapter.

VLAN Concepts and Configuration

After the IP phone has received power, it must determine its VLAN assignment. Because of security risks associated with having data and voice devices on the same network, Cisco recommends isolating IP phones in VLANs dedicated to voice devices. To understand how to implement this recommendation, let's first review a few key VLAN concepts.

VLAN Review

When VLANs were introduced a number of years ago, the concept was so radical and beneficial that it was immediately adopted into the industry. Nowadays, it is rare to find any reasonably sized network that is not using VLANs in some way.

VLANs allow you to break up switched environments into multiple broadcast domains. Here is the basic summary of a VLAN:

A VLAN = A Broadcast Domain = An IP Subnet

There are many benefits to using VLANs in an organization, some of which include the following:

- **Increased performance:** By reducing the size of the broadcast domain, network devices run more efficiently.
- **Improved manageability:** The division of the network into logical groups of users, applications, or servers allows you to understand and manage the network better.
- **Physical topology independence:** VLANs allow you to group users regardless of their physical location in the campus network. If departments grow or relocate to a new area of the network, you can simply change the VLAN on their new ports without making any physical network changes.
- **Increased security:** A VLAN boundary marks the end of a logical subnet. To reach other subnets (VLANs), you must pass through a routed (Layer 3) device. Any time you send traffic through a router, you have the opportunity to add filtering options (such as access lists) and other security measures.

VLAN Trunking/Tagging

VLANs are able to transcend individual switches, as shown in Figure 3-4.

If a member of VLAN_GRAY sends a broadcast message, it goes to all VLAN_GRAY ports on both switches. The same holds true for VLAN_WHITE. To accommodate this, the connection between the switches must carry traffic for multiple VLANs. This type of port is known as a trunk port.

Trunk ports are often called tagged ports because the switches send frames between each other with a VLAN “tag” in place. Figure 3-5 illustrates the following process:

1. HostA (in VLAN_GRAY) wants to send data to HostD (also in VLAN_GRAY). HostA transmits the data to SwitchA.
2. SwitchA receives the data and realizes that HostD is available through the FastEthernet 0/24 port (because HostD’s MAC address has been learned on this port). Because FastEthernet 0/24 is configured as a trunk port, SwitchA puts the VLAN_GRAY tag in the IP header and sends the frame to SwitchB.
3. SwitchB processes the VLAN_GRAY tag because the FastEthernet 0/24 port is configured as a trunk. Before sending the frame to HostD, the VLAN_GRAY tag is removed from the header.
4. The tagless frame is sent to HostD.

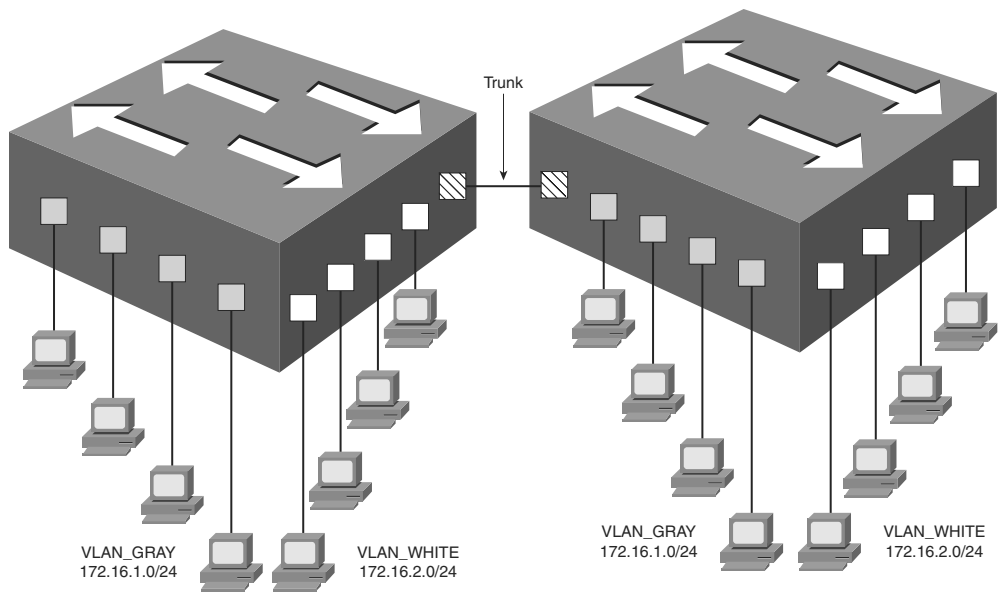


Figure 3-4 *VLANs Move Between Switches*

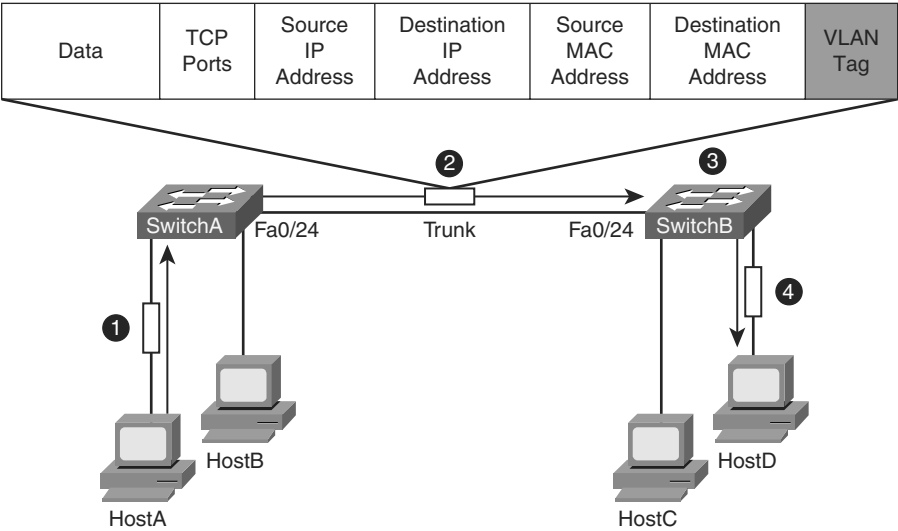


Figure 3-5 *VLAN Tags*



Using this process, the PC never knows what VLAN it belongs to. The VLAN tag is applied when the incoming frame crosses a trunk port. The VLAN tag is removed when exiting the port to the destination PC. Always keep in mind that VLANs are a switching concept; the PCs never participate in the VLAN tagging process.

VLANs are not a Cisco-only technology. Just about all managed switch vendors support VLANs. In order for VLANs to operate in a mixed-vendor environment, a common trunking or “tagging” language must exist between them. This language is known as 802.1Q. All vendors design their switches to recognize and understand the 802.1Q tag, which is what allows us to trunk between switches in any environment.

Understanding Voice VLANs

It is a common and recommended practice to separate voice and data traffic by using VLANs. There are already easy-to-use applications available, such as Wireshark and Voice Over Misconfigured Internet Telephones (VOMIT), that allow intruders to capture voice conversations on the network and convert them into WAV data files. Separating voice and data traffic using VLANs provides a solid security boundary, preventing data applications from reaching the voice traffic. It also gives you a simpler method to deploy QoS, prioritizing the voice traffic over the data.

One initial difficulty you can encounter when separating voice and data traffic is the fact that PCs are often connected to the network using the Ethernet port on the back of a Cisco IP Phone. Because you can assign a switchport to only a single VLAN, it initially seems impossible to separate voice and data traffic. That is, until you see that Cisco IP Phones support 802.1Q tagging.

The switch built into Cisco IP Phones has much of the same hardware that exists inside of a full Cisco switch. The incoming switchport is able to receive and send 802.1Q tagged packets. This gives you the capability to establish a type of trunk connection between the Cisco switch and IP phone, as shown in Figure 3-6.

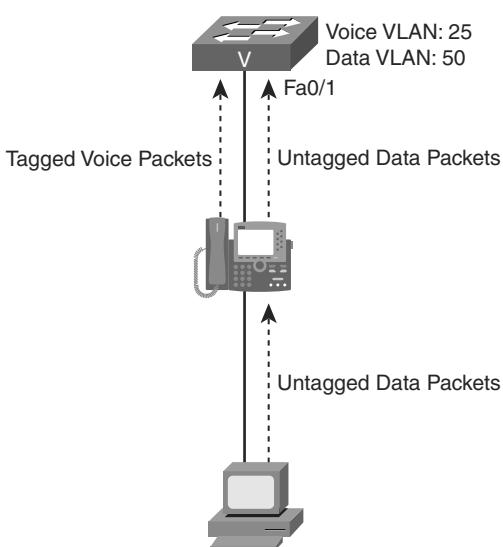


Figure 3-6 Separating Voice and Data Traffic Using VLANs

You might call the connection between the switch and IP phone a “mini-trunk” because a typical trunk passes a large number of VLANs (if not all VLANs). In this case, the IP phone tags its own packets with the correct voice VLAN (VLAN 25, in the case of Figure 3-6). Because the switch receives this traffic on a port supporting tagged packets (our mini-trunk), the switch can read the tag and place the data in the correct VLAN. The data packets pass through the IP phone and into the switch untagged. The switch assigns these untagged packets to whatever VLAN you have configured on the switchport for data traffic.

Note: Traditionally, a switchport on a Cisco switch that receives tagged packets is referred to as a trunk port. However, when you configure a switchport to connect to a Cisco IP Phone, you configure it as an access port (for the untagged data from the PC) while supporting tagged traffic from the IP phone. So, think of these ports as “access ports supporting tagged voice VLAN traffic.”



VLAN Configuration

Configuring a Cisco switch to support Voice VLANs is a fairly simple process. First, you can add the VLANs to the switch, as shown in Example 3-1.

Example 3-1 Adding and Verifying Data and Voice VLANs

```
Switch#configure terminal
Switch(config)#vlan 10
Switch(config-vlan)#name VOICE
Switch(config-vlan)#vlan 50
Switch(config-vlan)#name DATA
Switch(config-vlan)#end
Switch#show vlan brief
```

VLAN	Name	Status	Ports
1	default	active	Fa0/2, Fa0/3, Fa0/4, Fa0/5 Fa0/6, Fa0/7, Fa0/8, Fa0/9 Fa0/10, Fa0/11, Fa0/12, Fa0/13 Fa0/14, Fa0/15, Fa0/16, Fa0/17 Fa0/18, Fa0/19, Fa0/20, Fa0/21 Fa0/22, Fa0/23, Fa0/24, Gi0/1 Gi0/2
10	VOICE	active	
50	DATA	active	
1002	fddi-default	act/unsup	
1003	token-ring-default	act/unsup	
1004	fddinet-default	act/unsup	
1005	trnet-default	act/unsup	



Sure enough, VLANs 10 (VOICE) and 50 (DATA) now appear as valid VLANs on the switch. Now that the VLANs exist, you can assign the ports attaching to Cisco IP Phones (with PCs connected to the IP Phone) to the VLANs, as shown in Example 3-2.



Example 3-2 Assigning Voice and Data VLANs

```
Switch#configure terminal
Switch(config)#interface range fa0/2 - 24
Switch(config-if-range)#switchport mode access
Switch(config-if-range)#spanning-tree portfast
Switch(config-if-range)#switchport access vlan 50
Switch(config-if-range)#switchport voice vlan 10
Switch(config-if-range)#end
Switch#show vlan brief
```

VLAN	Name	Status	Ports
1	default	active	Gi0/1, Gi0/2
10	VOICE	active	Fa0/2, Fa0/3, Fa0/4, Fa0/5 Fa0/6, Fa0/7, Fa0/8, Fa0/9 Fa0/10, Fa0/11, Fa0/12, Fa0/13 Fa0/14, Fa0/15, Fa0/16, Fa0/17 Fa0/18, Fa0/19, Fa0/20, Fa0/21 Fa0/22, Fa0/23, Fa0/24
50	DATA	active	Fa0/2, Fa0/3, Fa0/4, Fa0/5 Fa0/6, Fa0/7, Fa0/8, Fa0/9 Fa0/10, Fa0/11, Fa0/12, Fa0/13 Fa0/14, Fa0/15, Fa0/16, Fa0/17 Fa0/18, Fa0/19, Fa0/20, Fa0/21 Fa0/22, Fa0/23, Fa0/24
1002	fddi-default	act/unsup	
1003	token-ring-default	act/unsup	
1004	fddinet-default	act/unsup	
1005	trnet-default	act/unsup	

Note: When connecting Cisco IP Phones to a switch, you should also enable portfast (using **spanning-tree portfast**, as shown in Example 3-2), because the IP phones boot quickly and request a DHCP assigned address before a typical port with spanning-tree enabled would go active. Also, keep in mind that port Fa0/1 does not appear in the Example 3-2 output because it is configured as a trunk port (ports 2–24 are not considered trunks by Cisco IOS).

The ports are now configured to support a voice VLAN of 10 and a data VLAN of 50. This syntax is a newer form of configuration for IP Phone connections. In the “old days,” you would configure the interface as a trunk port because the switch was establishing a trunking relationship between it and the IP phone. This was less secure because a hacker could remove the IP phone from the switchport and attach their own device (another managed

switch or PC) and perform a VLAN-hopping attack. The more modern syntax configures the port as a “quasi-access port,” because an attached PC will be able to access only VLAN 50. Only an attached Cisco IP Phone will be able to access the voice VLAN 10.

Note: Keep in mind that Cisco IP phones will be able to receive this voice VLAN configuration from the switch via CDP. After it receives the voice VLAN number, the IP Phone begins tagging its own packets. Non-Cisco IP Phones cannot understand CDP packets. This typically requires you to manually configure each of the non-Cisco IP Phones with its voice VLAN number from a local phone configuration window (on the IP phone).

Understanding the Cisco IP Phone Boot Process

Now that you learned about the VLAN architecture used with Cisco IP Phones, we can turn our attention to the IP Phones themselves. By understanding the IP Phone boot process, you can more fully understand how the Cisco IP Phone operates (which aids significantly in troubleshooting Cisco IP Phone issues). Here is the Cisco IP Phone boot process, start to finish:

1. The Cisco IP Phone connects to an Ethernet switchport. If the IP phone and switch support PoE, the IP phone receives power through either Cisco-proprietary PoE or 802.3af PoE.
2. As the Cisco IP Phone powers on, the Cisco switch delivers voice VLAN information to the IP phone using CDP as a delivery mechanism. The Cisco IP Phone now knows what VLAN it should use.
3. The Cisco IP Phone sends a DHCP request asking for an IP address on its voice VLAN.
4. The DHCP server responds with an IP address offer. When the Cisco IP Phone accepts the offer, it receives all the DHCP options that go along with the DHCP request. DHCP options include items such as default gateway, DNS server information, domain name information, and so on. In the case of Cisco IP Phones, a unique DHCP option is included, known as Option 150. This option directs the IP phone to a TFTP server. (You learn more about this in the upcoming section, “Configuring a Router-Based DHCP Server.”)
5. After the Cisco IP Phone has the IP address of the TFTP server, it contacts the TFTP server and downloads its configuration file. Included in the configuration file is a list of valid call processing agents (such as Cisco Unified Communications Manager or Cisco Unified Communications Manager Express CME agents).
6. The Cisco IP Phone attempts to contact the first call processing server (the primary server) listed in its configuration file to register. If this fails, the IP phone moves to the next server in the configuration file. This process continues until the IP phone registers successfully or the list of call processing agents is exhausted.

Configuring a Router-Based DHCP Server

We currently made it to Step 4 in the preceding IP phone boot process. The phones in our network now need to receive IP address and TFTP server information. In the network design scenario used in this chapter, we use the WAN branch router as the DHCP server.

Using a router as a DHCP server is a somewhat common practice in smaller networks. Once you move into larger organizations, DHCP services are typically centralized onto server platforms. Either DHCP option is capable of sending TFTP server information to the IP phones.

Example 3-3 shows the syntax used to configure a WAN branch router as a DHCP server.

Example 3-3 *Configuring Router-Based DHCP Services*

```
WAN_RTR#configure terminal
WAN_RTR(config)#ip dhcp excluded-address 172.16.1.1 172.16.1.9
WAN_RTR(config)#ip dhcp excluded-address 172.16.2.1 172.16.2.9
WAN_RTR(config)#ip dhcp pool DATA_SCOPE
WAN_RTR(dhcp-config)#network 172.16.2.0 255.255.255.0
WAN_RTR(dhcp-config)#default-router 172.16.2.1
WAN_RTR(dhcp-config)#dns-server 4.2.2.2
WAN_RTR(dhcp-config)#exit
WAN_RTR(config)#ip dhcp pool VOICE_SCOPE
WAN_RTR(dhcp-config)#network 172.16.1.0 255.255.255.0
WAN_RTR(dhcp-config)#default-router 172.16.1.1
WAN_RTR(dhcp-config)#option 150 ip 172.16.1.1
WAN_RTR(dhcp-config)#dns-server 4.2.2.2
```

Note: This example uses a Cisco router as a DHCP server. I (Jeremy) took this approach because using a router as a DHCP server is simple and stable. That being said, most people use a Windows server or some other centralized device for DHCP services. Even Cisco Unified Communications Manager includes DHCP server capabilities. In these cases, you typically need to configure an **ip helper-address** <central DHCP server IP address> to forward DHCP requests to the central DHCP server for the voice VLAN devices.

The way in which Cisco routers approach DHCP configurations is slightly different from how many other DHCP servers do so. Most DHCP servers allow you to specify a range of IP addresses that you would like to hand out to clients. Cisco routers take the opposite approach: you first specify a range of addresses that you do not want to hand out to clients (using the **ip dhcp excluded-address** syntax from global configuration mode). Configuring the excluded addresses before you configure the DHCP pools ensures that the Cisco router does not accidentally hand out IP addresses before you have a chance to exclude them from the range. The DHCP service on the router will begin handing out IP addresses from the first nonexcluded IP address in the network range. In Example 3-3, this is 172.16.1.10 for the voice scope and 172.16.2.10 for the data scope.

Tip: Notice a DNS server of 4.2.2.2 is assigned to both the data and voice devices. This is a well-known, open DNS server on the Internet. This IP address works fantastically to test connectivity and DNS services in new network deployments because it is such a simple IP address to remember.

Also notice that the VOICE_SCOPE DHCP pool includes the option 150 syntax. This creates the custom TFTP server option to be handed out to the Cisco IP Phones along with their IP address information. In this case, the TFTP server of the IP phones is the same as the default gateway because we use the CME router as a call processing agent. As mentioned in the section, “Understanding the Cisco IP Phone Boot Process,” the TFTP server holds the configuration files for the phones. When you configure a Cisco IP Phone in Cisco Unified Communications Manager (CUCM) or CME, an XML configuration file is generated and stored on a TFTP server. These CML configuration files have a filename format of SEP<IP Phone MAC Address>.cnf.xml and contain a base configuration for the IP phone (specifying language settings, URLs, and so on). Most importantly, these XML files contain a list of up to three CUCM server or CME IP addresses the Cisco IP Phone uses for registration. After the IP phone receives the XML file, it attempts to register with the first CUCM or CME server listed in the file. If it is unable to reach that server, it moves down to the next until the list is exhausted (at which point the IP phone reboots and tries it all over again).



Note: If the Cisco IP Phone has not yet been configured in CUCM or CME (no SEP<MAC>.cnf.xml file exists on the TFTP server), the IP Phone requests a file named XMLDefault.cnf.xml. This is a base configuration file typically used for a feature called Auto-Registration (allowing phones to register without being configured).

Tip: Many people often wonder the meaning of SEP at the beginning of the configuration filename. SEP stands for Selsius Ethernet Phone. Selsius was the name of the company Cisco acquired when they first began manufacturing VoIP technology.

Setting the Clock of a Cisco Device with NTP

The final task to prepare the network infrastructure to support a Cisco VoIP network is to set the time. Having an accurate time on Cisco devices is important for many reasons. Here is a quick list of just some of the reasons why you want an accurate clock on your network devices:

- It allows Cisco IP Phones to display the correct date and time to your users.
- It assigns the correct date and time to voicemail tags.
- It gives accurate times on Call Detail Records (CDR), which are used to track calls on the network.
- It plays an integral part in multiple security features on all Cisco devices.
- It tags logged messages on routers and switches with accurate time information.

When Cisco devices boot, many of them default their date and time to noon on March 1, 1993. You have two options in setting the clock: manually, using the **clock set** command from the privileged EXEC mode, or automatically, using the Network Time Protocol (NTP).

Devices setting the clock using NTP always have a more accurate time clock than a manually set clock. Likewise, all the NTP devices on your network will have the exact same

time. These advantages make NTP the preferred clock-setting method. The accuracy of the clock on your device depends on the stratum number of the NTP server. A stratum 1 time server is one that has a radio or atomic clock directly attached. The device that receives its time from this server via NTP is considered a stratum 2 device. The device that receives its time from this stratum 2 device via NTP is considered a stratum 3 device, and so on. There are many publicly accessible stratum 2 and 3 (and even some stratum 1) devices on the Internet.

Note: You can obtain a list of publicly accessible NTP servers at www.ntp.org.

After you obtain one or more NTP servers to use, you can configure NTP support on your Cisco devices by using the syntax in Example 3-4.

Example 3-4 *Configuring a Cisco Router to Receive Time via NTP*

```
WAN_RTR#configure terminal
WAN_RTR(config)#ntp server 64.209.210.20
WAN_RTR(config)#clock timezone ARIZONA -7
```

The first command, **ntp server <ip address>**, configures your Cisco device to use the specified NTP server; 64.209.210.20 is one of many publicly accessible NTP servers. If this is the only command you enter, your clock on your device will set itself to the Universal Time Coordinated (UTC) time zone. To accurately adjust the time zone for your device, use the **clock timezone <name> <hours>** command. The previous syntax example set the time zone for Arizona to -7 hours from UTC.

Now that we configured the router to synchronize with an NTP server, we can verify the NTP associations and the current time and date using the commands shown in Example 3-5.

Example 3-5 *Verifying NTP Configurations*

```
WAN_RTR#show ntp associations
      address          ref clock      st  when  poll reach  delay  offset  disp
*~64.209.210.20      138.23.180.126    3   14    64  377    65.5    2.84    7.6
* master (syncd), # master (unsyncd), + selected, - candidate, ~ configured
WAN_RTR#show clock
11:25:48.542 CA1_DST Mon Dec 13 2010
```

The key information from the **show ntp associations** command is just to the left of the configured NTP server address. The asterisk indicates that your Cisco device has synchronized with this server. You can configure multiple NTP sources for redundancy, but the Cisco device will only choose one master NTP server to use at a time.

After you configure the Cisco router to synchronize with an NTP server, you can configure it to provide date and time information to a CUCM server, which can then provide that date and time information to the Cisco IP Phones in your network. To allow other

devices (such as a CUCM server) to pull date and time information from a Cisco router using NTP, use the **ntp master <stratum number>** command from global configuration mode. For example, entering **ntp master 4** instructs the Cisco router to deliver date and time information to requesting clients, marking it with a stratum number of 4.

Note: Example 3-4 illustrates configuring a Cisco router to support NTP. This is necessary if you are supporting a Cisco IP Telephony network using Communication Manager Express (CME). If you were using a full CUCM solution, you'd configure NTP on the CUCM server.

IP Phone Registration

Now that the Cisco IP Phone has gone through the complete process, it is ready to register with the call-management system (CME or CUCM). Before we discuss this final step, keep in mind what the phone has gone through up to this point:

1. The phone has received Power over Ethernet (PoE) from the switch.
2. The phone has received VLAN information from switch via CDP.
3. The phone has received IP information from the DHCP server (including Option 150).
4. The phone has downloaded its configuration file from the TFTP server.

The Cisco IP Phone is now looking at a list of up to three call processing servers (depending on how many you have configured) that it found in the configuration file it retrieved from the TFTP server. The phone tries to register with the first call processing server. If that fails, it continues down the list it received from the TFTP server until the phone makes it through all the listed call processing servers (at which point it reboots if it finds no servers online).

If the IP phone finds an active server in the list, it goes through the registration process using either the Skinny Client Control Protocol (SCCP) or Session Initiation Protocol (SIP). The protocol the phone uses depends on the firmware it is using. Today, most Cisco IP Phones use the SCCP, which is Cisco proprietary. However, as the SIP protocol matures, widespread support continues to grow. Because SIP is an industry standard, using it across your network provides benefits such as vendor neutrality and inter-vendor operation.



Note: The SIP standard is moving so quickly, by the time you read this, SCCP may not be the most popular protocol for Cisco IP Telephony networks. SCCP will most likely take its place in the proprietary protocol history books (which contain other items, such as the InterSwitch Link [ISL] trunking protocol and the Cisco original inline power method).

Regardless of the protocol used, the registration process is simple: The Cisco IP Phone contacts the call processing server and identifies itself by its MAC address. The call processing server looks at its database and sends the operating configuration to the phone. The operating configuration is different than the settings found in the configuration XML file located on the TFTP server. The TFTP server configuration is “base level settings,” including items such as device language, firmware version, call processing server IP addresses, port numbers, and so on. The operating configuration contains items such as

directory/line numbers, ring tones, softkey layout (on-screen buttons), and so on. Although the TFTP server configuration is sent using the TFTP protocol, the operating configuration is sent using SIP or SCCP.

These protocols (SIP or SCCP) are then used for the vast majority of the phone functionality following the registration. For example, as soon as a user picks up the handset of the phone, it sends a SCCP or SIP message to the call processing server indicating an off-hook condition. The server quickly replies with a SCCP or SIP message to play dial tone and collect digits. As the user dials, digits are transmitted to the call processing server using SCCP or SIP; call progress tones, such as ringback or busy, are delivered from the call processing server to the phone using SCCP or SIP. Hopefully, you get the idea: The Cisco IP Phone and call processing server have a dumb terminal and mainframe style of relationship, and the “language of love” between them is SCCP or SIP.

Exam Preparation Tasks

Review All the Key Topics

Review the most important topics in the chapter, noted with the key topics icon in the outer margin of the page. Table 3-2 lists and describes these key topics and identifies the page number on which each is found.

Table 3-2 *Key Topics for Chapter 3*

Key Topic Element	Description	Page Number
Figure 3-5	Trunking tag concepts	56
Figure 3-6	Separating voice and data traffic using VLANs	58
Examples 3-1 and 3-2	Configuring voice and data VLANs	59-60
Note	CDP delivers Voice VLAN information	59
Text	Cisco phones receive DHCP Option 150 to download an .xml configuration file via TFTP.	63
Text	Two primary signaling protocols to Cisco IP Phones are SIP and SCCP.	65

Definitions of Key Terms

Define the following key terms from this chapter, and check your answers in the Glossary:

802.3af Power over Ethernet (PoE), Cisco Inline Power, Cisco Discovery Protocol (CDP), virtual LAN (VLAN), trunking, 802.1Q, Dynamic Trunking Protocol (DTP), Skinny Client Control Protocol (SCCP), Session Initiation Protocol (SIP), Network Time Protocol (NTP)

Index

Symbols

no] digit-strip command, 135

μ-law, 20

Numbers

640-461 ICOMM exam, 443

A

a-law, 20

AAR (Automated Alternate Routing),
346

access layer switchpoints, AutoQoS,
157-158

access lists

CUCM (Cisco Unified
Communications Manager), 317-318
configuring, 322-325
empty, 318

administration interface, CUCM
(Cisco Unified Communications
Manager), 216

Advanced Features menu (CM
Administration interface), 216

after-hours call blocking, *configuring*,
CME (Communications Manager
Express), 195-199

Alarm menu (Cisco Unified
Serviceability interface), 217

alerts reports, CUC (Cisco Unity
Connection), 432

analog connections, 6-9

analog signals

digital signals, converting to, 9-11
repeaters, 9

analog telephones, PSTN (public
switched telephone network), 13

analog voice ports, *configuring*, 108

analog waveforms, 6

Application menu (CM Administration
interface), 217

application rules, CUC (Cisco Unity
Connection), 342

Application Users, versus Endpoint
Users, 243-244

applications

best-effort applications, 154
mission-critical applications, 154
scavenger applications, 155
transactional applications, 154

assigning, COR (Class of Restriction)
lists, 150

associating, ephone/ephone-dn, 92-95

attribute mapping, LDAP (Lightweight
Directory Access Protocol),
249-250

audio codec bandwidth, MOS (Mean
Opinion Score), 21

authentication, LDAP (Lightweight
Directory Access Protocol), 248,
254-255

auto qos voip cisco-phone command,
162

auto qos voip command, 162
auto qos voip trust command, 162
AutoQoS, 155-162
 access layer switchpoints, 157-158
 configuration points, 157
 router interfaces, 160-161
 switch router uplink, 159-160
 syntax variations, 162

B

bandwidth, 21
 audio codec, MOS (Mean Opinion Score), 21
 VoIP, 153
barge feature (CUCM), 291
 configuring, 295-296
base configuration, CME (Communications Manager Express), 87-88
BAT (Bulk Administration Tool), End Users, 246
best-effort applications, 154
BLF Speed Dials (CUCM), configuring, 305-306
boot process, Cisco IP phones, 84
Bulk Administration menu (CM Administration interface), 217
bulk import, End Users, BAT (Bulk Administration Tool), 246
businesses, VoIP, benefits, 17-18

C

cabling, VoIP, cost savings, 17
call accounting, configuring, CME (Communications Manager Express), 199-203
call actions, voicemail, 345
call coverage, CUCM (Cisco Unified Communications Manager), 290
 enabling, 295
call flow, CUCM (Cisco Unified Communications Manager), 263
 call routing, 270-273
 centralized deployment PSTN backup call flow, 267
 centralized remote branch call flow, 265-266
 CoC (Class of Service), 274-275
 CSS (Calling Search Space), 275
 digit analysis, 273-274
 distributed deployment call flow, 268-270
 DNS (Domain Name System), 263
 gateways, 272-273
 Hunt Group, 274
 line-device configuration, 276
 Partition, 275
 PSTN backup using CAC, 268
 Route Group, 272
 Route List, 271-272

trunks, 272-273

without DNS, 263-265

call forwarding

configuring, CME (Communications Manager Express), 175-179

CUCM (Cisco Unified Communications Manager), 291

voicemail, 345

call handlers, CUC (Cisco Unity Connection), 340

call hunting, CUCM (Cisco Unified Communications Manager), 292

configuring, 301-303

call lists, presence-enabled call lists, configuring, 306-307

call park

call routing, 270

configuring, CME (Communications Manager Express), 181-186

CUCM (Cisco Unified Communications Manager), 292

configuring, 299-300

call pickup

CME (Communications Manager Express), configuring, 186-189

CUCM (Cisco Unified Communications Manager), 291-292

configuring, 297-298

call processing

CME (Communications Manager Express), 32

CUCM (Cisco Unified Communications Manager), 40

call routing

CUC (Cisco Unity Connection), 340-341

CUCM (Cisco Unified Communications Manager)

behavior, 273

configuration elements, 271

destinations, 270-271

sources, 270

Call Routing menu (CM Administration interface), 216

call routing rule filters, CUC (Cisco Unity Connection), 341-342

call transfer, configuring, CME (Communications Manager Express), 180-181

caller ID, CCP (Cisco Configuration Professional), 174

CallManager. *See* CUCM (Cisco Unified Communications Manager)

CAR (Call Detail Record Analysis and Reporting) tool, CUCM (Cisco Unified Communications Manager), 408-416

catalyst switch PoE, Cisco IP phones, 54

CCP (Cisco Configuration Professional), 95-101

caller ID, 174

CME (Communications Manager Express), configuring dial-plans, 145-146

directory sorting, 174

CCS (common channel signaling), 12

CDP (Cisco Discovery Protocol), IP phones, 235

CDRs (Call Detail Records), 408-416

configuring, CME (Communications Manager Express), 199-203

centralized deployment PSTN backup

call flow, CUCM (Cisco Unified Communications Manager), 267

centralized remote branch call flow, CUCM (Cisco Unified Communications Manager), 265-266

CFS (Client Services Framework), 373

Cisco AutoQoS, 155-162

access layer switchpoints, 157-158

configuration points, 157

- router interfaces, 160-161
- switch router uplink, 159-160
- syntax variations, 162
- Cisco IP phones, 48-49**
 - boot process, 84
 - call routing, 270
 - catalyst switch PoE, 54
 - CDP (Cisco Discovery Protocol), 235
 - CM Groups, 241
 - CME (Communications Manager Express), interaction, 32-35
 - connecting, 52-55
 - CUCM (Cisco Unified Communications Manager), interaction, 38-41
 - Date/Time Groups, 241
 - defaults, 242
 - Device Pools, 240-242
 - DHCP (Dynamic Host Configuration Protocol), 235
 - DNS (Domain Name System), 235, 265
 - EM (Extension Mobility), enabling for, 288-289
 - forwarding calls from, 175
 - Hunt Groups, 274
 - implementing, CUCM (Cisco Unified Communications Manager), 234-243
 - locations, 241
 - Mobility features, configuring for, 320
 - NTP (Network Time Protocol), 234, 241-242
 - phone buttons template, 243
 - PoE (Power over Ethernet), 235
 - powering, 52-55
 - profiles, 243
 - regions, 241
 - registration, 65-66, 236
 - CME (Communications Manager Express)*, 385-389
 - CUCM (Cisco Unified Communications Manager)*, 403-405
 - service activation, 237
 - softkey template, 242
 - TFTP (Trivial File Transfer Protocol), 235
 - VLANs, 55-61
- Cisco Learning Network, 443**
- Cisco Unified Communications Manager (CUCM). See CUCM (Cisco Unified Communications Manager)**
- Cisco Unified Communications Manager Express (CME). See CME (Communications Manager Express)**
- Cisco Unified Operating System, interface, 219**
- Cisco Unified Presence, 44-45**
- Cisco Unified Reporting, interface, 220**
- Cisco Unified Serviceability, interface, 217**
- Cisco Unity Connection, 26, 41-44**
 - CUCM (Cisco Unified Communications Manager), interaction, 43-44
- Class of Restriction (COR) lists. See COR (Class of Restriction) lists**
- CLI, forwarding calls from, 177**
- CLI (command line interface), CUCM (Cisco Unified Communications Manager), 220-221**
- Client Services Framework (CSF), 373**
- clocks, setting, NTP, 63-65**
- CM Administration interface, 216**
- CM Groups, IP phones, 241**
- CME (Communications Manager Express), 26, 30-31-37, 68-69, 166-167, 380-381**
 - after-hours call blocking, configuring, 195-199
 - call accounting, configuring, 199-203

- call forwarding, configuring, 175-179
- call park, configuring, 181-186
- call pickup, configuring, 186-189
- call processing, 32
- call transfer, configuring, 180-181
- CCP (Cisco Configuration Professional), 95-101
- CDRs (Call Detail Records), configuring, 199-203
- Cisco IP phones, interaction, 32-35
- configuration, base, 87-88
- CTI (Computer Telephony Integration), 32
- CUE (Cisco Unity Express), 32-37
- device control, 32
- dial-plans, 103-104
 - configuring dial peers, 117-130*
 - configuring physical voice port characteristics, 108-117*
 - COR (Class of Restriction) lists, 146-152*
 - digit manipulation, 135-146*
 - QoS (Quality of Service), 152-162*
 - router call processing, 130-146*
 - troubleshooting, 389-392*
- end users, 80-81
- endpoints, 80-81
- ephone-dn, 95
- ephones, 95
- flash-based GUI, enabling, 206-209
- foundation, ensuring, 84-88
- intercom, configuring, 189-192
- local directory service, 32
- managing
 - command line, 71-73*
 - GUIs (Graphic User Interface), 73-77*
- MoH (Music on Hold), configuring, 203-204

- paging, configuring, 192-195
- single number reach, configuring, 204-206
- TFTP services, 86-87
- troubleshooting, 384-385
 - dial-plans, 389-392*
 - QoS (Quality of Service), 392-395*
 - registration, 385-389*
- voice network directories, configuring, 171-175
- CO switches, PSTN (public switched telephone network), 13
- CoC (Class of Service), CUCM (Cisco Unified Communications Manager), call flow, 274-275
- codecs
 - audio bandwidth, MOS (Mean Opinion Score), 21
 - complexity, 23
 - G.711, 20
 - G.729, 21
- command line, CME (Communications Manager Express), managing, 71-73
- commands
 - [no] digit-strip, 135
 - auto qos voip, 162
 - auto qos voip cisco-phone, 162
 - auto qos voip cisco-softphone, 162
 - auto qos voip trust, 162
 - debug voip dialpeer, 122-391
 - destination-pattern, 126
 - digits set digits, 136
 - forward-digits number, 135
 - num-exp, 139
 - num-exp match, 136
 - park slot, 184
 - prefix, 136-138
 - prefix digits, 135
 - show dial-peer voice summary, 390

- show logging, 200
- show policy-map interface, 394
- voice-translation profile, 136
- common channel signaling (CCS), 12**
- Communications Manager Express (CME). See CME (Communications Manager Express)**
- configuration**
 - CME (Communications Manager Express)
 - after-hours call blocking, 195-199*
 - base, 87-88*
 - call accounting, 199-203*
 - call forwarding, 175-179*
 - call park, 181-186*
 - call pickup, 186-189*
 - call transfer, 180-181*
 - CDRs (Call Detail Records), 199-203*
 - dial-plans, 108-130*
 - intercom, 189-192*
 - MoH (Music on Hold), 203-204*
 - paging, 192-195*
 - single number reach, 204-206*
 - voice network directories, 171-175*
- CUCM (Cisco Unified Communications Manager)
 - barge, 295-296*
 - BLF Speed Dials, 305-306*
 - call hunting, 301-303*
 - call park, 299-300*
 - call pickup, 297-298*
 - intercom, 304-305*
 - native presence, 305*
 - presence groups, 307-309*
 - presence-enabled call lists, 306-307*
 - service parameters, 325-326*
 - shared lines, 295*

- ephone-dns, 89-90
- ephones, 90-92
- LDAP (Lightweight Directory Access Protocol), 252-253
- router-based DHCP servers, 61-63
- routers, DHCP scope, 85
- VLANs, 59-61
- voice VLANs, 85
- connections**
 - analog, 6-9
 - digital, 9-12
 - FXO (Foreign Exchange Office) , 111
 - FXS (Foreign Exchange Station) , 108
 - PSTN (public switched telephone network), 14-15
- COR (Class of Restriction) lists, CME (Communications Manager Express), 146-152**
- CoS (Class of Service), 162**
- Credential Policy, End Users, 244**
- CSS (Calling Search Space), CUCM (Cisco Unified Communications Manager), 275**
- CTI (Computer Telephony Integration), CME (Communications Manager Express), 32**
- CUC (Cisco Unity Connection)**
 - application rules, 342
 - call handlers, 340
 - call routing, 340-341
 - call routing rule filters, 341-342
 - dial-plans, 342
 - direct routing rules, 341
 - DLs (Distribution Lists), 342
 - forwarded routing rules, 341
 - interfaces, 223-225
 - monitoring, 422-423

notification devices, 354-355

reports

alerts reports, 432

analyzing, 430-433

generating and accessing, 426-433

*Phone Interface Failed Logon
report, 433*

Port Activity report, 435

serviceability reports, 426-427

*troubleshooting and maintenance,
433-437*

User Lockout reports, 433

system settings

General Configuration page, 339

Roles page, 339

Unified Serviceability, 225

voicemail, 332-333, 336-347

*AAR (Automated Alternate
Routing), 346*

call actions, 345

call forwarding, 345

end users, 355-363

extensions, 345

greetings, 344

mailbox, 346-363

message actions and settings, 345

message aging policy, 347

password settings, 344

private DLs, 346

SCCP, 337-338

*SIP (Session Initiation Protocol),
338*

*SRST (Survivable Remote Site
Telephony), 346*

transfer rules, 344

user creation, 346

User Templates, 343-344

**CUCM (Cisco Unified Communications
Manager), 26, 31, 37-41**

barge feature, 291

configuring, 295-296

BLF Speed Dials, configuring, 305-306

call coverage, 290

enabling, 295

call flow, 263

call routing, 273

*centralized deployment PSTN
backup call flow, 267*

*centralized remote branch call
flow, 265-266*

CoC (Class of Service), 274-275

CSS (Calling Search Space), 275

digit analysis, 273-274

*distributed deployment call flow,
268-270*

DNS (Domain Name System), 263

gateways, 272-273

Hunt Group, 274

line-device configuration, 276

Partition, 275

PSTN backup using CAC, 268

Route Group, 272

Route List, 271-272

trunks, 272-273

without DNS, 263-265

call forwarding, 291

call hunting, 292

configuring, 301-303

call park, 292

configuring, 299-300

call pickup, 291-292

configuring, 297-298

call processing, 40

- call routing
 - configuration elements*, 271
 - destinations*, 270-271
 - sources*, 270
- CAR (Call Detail Record Analysis and Reporting) tool, 408-416
- CDRs (Call Detail Records), 408-416
- Cisco IP phones, interaction, 38-41
- Cisco Unity Connection, interaction, 43-44
- database replication, 38-41
- dial-plans, 259
- digit-by-digit analysis, 274
- EM (Extension Mobility), 282-306
 - enabling in*, 283-289
- End Users, 230-231, 243-246
 - implementing*, 246-256
- endpoints, 230-231
- groups, 222-223
- intercom, 293
 - configuring*, 304-305
- interfaces
 - administration*, 216
 - Cisco Unified Reporting*, 220
 - Cisco Unified Serviceability*, 217
 - CLI (command line interface)*, 220-221
 - DRS (Disaster Recovery System)*, 220
 - Unified Operating System*, 219
- IP phones
 - configuration requirements*, 240-243
 - implementing*, 234-243
 - service activation*, 237
- LDAP (Lightweight Directory Access Protocol)
 - attribute mapping*, 249-250
 - integration*, 246-249
 - synchronization*, 248
- Mobility features, 312-313
 - access lists*, 317-318, 322-325
 - implementing*, 319-329
 - IP phones*, 320
 - Mobile Connect*, 316-317
 - MVA (Mobile Voice Access)*, 318
 - service parameters*, 325-326
 - user accounts*, 319-320
- MVA (Mobile Voice Access), configuring, 326-329
- native presence, 293
 - configuring*, 305
- presence architecture, 293-294
- presence groups, configuring, 307-309
- presence-enabled call lists, configuring, 306-307
- privacy feature, 291
- reports
 - analyzing*, 407
 - generating*, 406
- roles, 221-222
- shared lines, 291
 - configuring*, 295
- troubleshooting, 398-399, 402, 405
 - DRS (Disaster Recovery System)*, 418-420
 - IP phone registration*, 403-405
 - reports*, 406-408
 - RTMT (Real-Time Monitoring Tool)*, 416-417
- unassigned directory numbers, deleting, 405

CUE (Cisco Unity Express)

CME (Communications Manager Express), 32-37

modules, 36

CUPC (Cisco Unified Personal Communicator), 370-373

instant messaging, 371-372

video calls, 372

voice calls, 372

CUPS (Cisco Unified Presence Server), 366-367, 370, 374

CUC (Cisco Unity Connection) integration, 375

interfaces, 226-227

LDAP (Lightweight Directory Access Protocol) integration, 374-375

Microsoft Office Communications Server integration, 374-375

Persistent Chat, 376-377

QoS (Quality of Service), 377-378

custom filters, LDAP (Lightweight Directory Access Protocol), 251, 255

D

data, network requirements, 154-155

database replication, CUCM (Cisco Unified Communications Manager), 38-41

debug voip dialpeer command, 122, 391

delay, VoIP, 153

destination patterns, 127

brackets wildcard, 127

PSTN (public switched telephone network), North America, 127

destination-pattern command, wildcards, 126

device control, CME (Communications Manager Express), 32

Device menu (CM Administration interface), 217

Device Pools, IP phones, 240, 242

DHCP (Dynamic Host Configuration Protocol)

IP phones, 235

router IOS, configuring in, 239-240

server configuration, 237-239

DHCP scope, routers, configuring on, 85

dial peers, 117-118

configuring, 117-130

matching, 132-135

PLAR (Private Line Automatic Ringdown), 128-130

POTS (Plain Old Telephone Service) dial peers, 118

configuring, 120-124

voice call legs, 118-119

VoIP dial peers, 118

Configuring, 124-126

wildcards, 126-128

dial-plans

CME (Communications Manager Express), 103-104

configuring dial peers, 117-130

configuring physical voice port characteristics, 108-117

COR (Class of Restriction) lists, 146-152

digit manipulation, 135-146

QoS (Quality of Service), 152-162

router call processing, 130-146

troubleshooting, 389-392

CUC (Cisco Unity Connection), 342

CUCM (Cisco Unified Communications Manager), 258

call flow, 263-276

digit analysis, CUCM (Cisco Unified Communications Manager), 273-274

digit manipulation, 135-146
 operator calls, directing to receptionist, 138-139
 PSTN failover, prefix command, 136-138
 routers, 136
 translation profiles, 141-145

digital connections, 9-12

digital signal processors (DSPs). See DSPs (digital signal processors)

digital signals
 analog signals, converting from, 9-11
 processors, 22-23

digital telephones, PSTN (public switched telephone network), 14

digital voice ports, configuring, 112-117

digit-by-digit analysis, CUCM (Cisco Unified Communications Manager), 274

digits set digits command, 136

direct routing rules, CUC (Cisco Unity Connection), 341

directory number (DN)
 call routing, 270
 adding, CCP (Cisco Configuration Professional), 95-101

DirSync, activating, 251

distributed deployment call flow, CUCM (Cisco Unified Communications Manager), 268-270

DLs (Distribution Lists)
 CUC (Cisco Unity Connection), 342
 voicemail, 346

DN (directory number), call routing, 270

DNS (Domain Name System)
 CUCM (Cisco Unified Communications Manager), call flows, 263
 IP phones, 235

DRS (Disaster Recovery System), 220
 CUCM (Cisco Unified Communications Manager), 418-420
 scheduled backups, 419

DSPs (digital signal processors), 22-23

E

Edison, Thomas, 6

EM (Extension Mobility)
 CUCM (Cisco Unified Communications Manager), 282-306
enabling in, 283-289
 device profiles
associating users with, 287-288
creating, 285-286
creating defaults, 285
subscribing to EM service, 286
 IP phones, enabling for, 288-289
 service
activating, 283
adding, 284
configuring, 283

emergency calls, POTS (Plain Old Telephone Service) lines, 139-141

end users, 230-231, 243-246
 bulk import, BAT (Bulk Administration Tool), 246
 CME (Communications Manager Express), 80-81
 Credential Policy, 244
 CUC (Cisco Unity Connection), 355-363
 device association, 245
 implementing, CUCM (Cisco Unified Communications Manager), 246-256
 LDAP (Lightweight Directory Access Protocol)
attribute mapping, 249-250

integration, 246-247

synchronization, 248

manual entry, 246

user locale, 245

versus Application Users, 243-244

endpoints, 230-231

CME (Communications Manager Express), 80-81

ephone-dns, 95

associating, 92-95

configuring, 89-90

ephones

associating, 92-95

configuring, 90-92

Extension Mobility (EM). *See* EM (Extension Mobility)

extensions, voicemail, 345

F

filters, 255

LDAP (Lightweight Directory Access Protocol), 251

creating, 255

flash-based GUI, CME (Communications Manager Express), enabling, 206-209

forward-digits number command, 135

forwarded routing rules, CUC (Cisco Unity Connection), 341

FXO (Foreign Exchange Office) ports

configuring, 111-112

connections, 111

FXS (Foreign Exchange Station) ports

configuring, 108-111

connections, 108

ground start, 109

loop start, 109

G

G.711 codec, 20

G.729 codec, 21

gateways

call routing, 270

CUCM (Cisco Unified Communications Manager), call flow, 272-273

General Configuration page (CUC), 339

generating, CUC reports, 426-433

greetings, voicemail, 344

ground start, FXS (Foreign Exchange Station) ports, 109

groups, CUCM (Cisco Unified Communications Manager), 222-223

GUIs (Graphic User Interfaces)

CME (Communications Manager Express), managing, 73-77, 206-209

CUC (Cisco Unity Connection), 223-225

CUCM (Cisco Unified Communications Manager), 216

CUPS (Cisco Unified Presence), 226-227

H

Help menu (Cisco Unified Serviceability interface), 218

Help menu (CM Administration interface), 217

Hunt Groups, 274

hunt pilot, call routing, 270

I-K

inbound dial peers, outbound dial peers, matching, 132-135

incoming COR lists

assigning, 150

creating, 149

instant messaging, CUPC (Cisco Unified Personal Communicator), 371-372
intercom, configuring, CME (Communications Manager Express), 189-192
intercom feature, CUCM (Cisco Unified Communications Manager), 293
 configuring, 304-305
interfaces
 CUC (Cisco Unity Connection), 223-225
 CUCM (Cisco Unified Communications Manager), 216
 administration, 216
 Cisco Unified Reporting, 220
 Cisco Unified Serviceability, 217
 CLI (command line interface), 220-221
 DRS (Disaster Recovery System), 220
 Unified Operating System, 219
 CUPS (Cisco Unified Presence), 226-227
 routers, AutoQoS, 160-161
 RTMT (Real-Time Monitoring Tool), 416-417
IP Phone Messenger (IPPM), 373-374
IP phones, 48-49
 boot process, 84
 call routing, 270
 CDP (Cisco Discovery Protocol), 235
 CM Groups, 241
 CME (Communications Manager Express), interaction, 32-35
 connecting, 52-55
 CUCM (Cisco Unified Communications Manager), 38-41
 Date/Time Groups, 241
 defaults, 242

 Device Pools, 240-242
 DHCP (Dynamic Host Configuration Protocol), 235
 DNS (Domain Name System), 235-265
 EM (Extension Mobility), enabling for, 288-289
 forwarding calls from, 175
 Hunt Groups, 274
 implementing, CUCM (Cisco Unified Communications Manager), 234-243
 locations, 241
 Mobility features, configuring for, 320
 NTP (Network Time Protocol), 234, 241-242
 phone buttons template, 243
 PoE (Power over Ethernet), 235
 profiles, 243
 regions, 241
 registration, 65-66, 236
 CME (Communications Manager Express), 385-389
 CUCM (Cisco Unified Communications Manager), 403-405
 service activation, 237
 softkey template, 242
 TFTP (Trivial File Transfer Protocol), 235
 VLANs (Virtual LANs), 55-61

IP SoftPhones, 18

IPPM (IP Phone Messenger), 373-374

L

LDAP (Lightweight Directory Access Protocol)

 attribute mapping, 249-250
 authentication, 248, 254-255
 configuring, 252-253
 CUPS (Cisco Unified Presence Server), integration, 374-375

- custom filters, 251
 - creating*, 255
- integration, CUCM (Cisco Unified Communications Manager), 246-249
- Sync, 250
 - agreements*, 250
 - mechanisms*, 251
 - verifying*, 253
- synchronization, CUCM (Cisco Unified Communications Manager), 248
- local directory service, CME (Communications Manager Express), 32
- local loops, PSTN (public switched telephone network), 13
- locations, IP phones, 241
- loop start, FXS (Foreign Exchange Station) ports, 109

M

- Mailboxes, voicemail, 346-347
 - CUC (Cisco Unity Connection), 347-363
 - password settings, 349-350
- matching, dial peers, 132-135
- Mean Opinion Score (MOS), audio codec bandwidth, 21
- Media Resources menu (CM Administration interface), 216
- meet-me number, call routing, 270
- message aging policy, voicemail, 347
- Message Settings page (CUC), 350
- mission-critical applications, 154
- Mobile Connect
 - configuring, 319
 - CUCM (Cisco Unified Communications Manager), 316-317

- Remote Destination Profiles, 317
 - creating*, 320-321
- softkey templates, configuring, 319
- Mobility features, CUCM (Cisco Unified Communications Manager), 312-313**
 - access lists, 317-325
 - implementing, 319-329
 - IP phones, 320
 - Mobile Connect, 316-317
 - MVA (Mobile Voice Access), 318, 326-329
 - service parameters, 325-326
 - user accounts, 319-320
- modules, CUE (Cisco Unity Express), 36
- MoH (Music on Hold), configuring, CME (Communications Manager Express), 203-204
- monitoring, CUC (Cisco Unity Connection), 422-423
- MOS (Mean Opinion Score), audio codec bandwidth, 21
- multiple-group paging, 194
- MVA (Mobile Voice Access), CUCM (Cisco Unified Communications Manager), 318
 - configuring, 326-329

N

- native presence, CUCM (Cisco Unified Communications Manager), 293
 - configuring, 305
- network requirements
 - data, 154-155
 - video and voice, 154
- notification devices, CUC (Cisco Unity Connection), 354-355
- NTP (Network Time Protocol), IP phones, 234, 241-242
 - clocks, setting, 63-65

numbering plans, PSTN (public switched telephone network), 16-17
 num-exp command, 139
 num-exp match command, 136
 Nyquist, Harry, 18

O

operator calls, directing to receptionist, 138-139
 outbound dial peers, inbound dial peers, matching, 132-135
 outgoing COR lists
 assigning, 150
 creating, 149

P

packet loss, VoIP, 153
 packets, voice, converting from, 18-21
 paging
 CME (Communications Manager Express), configuring, 192-195
 configuring, CME (Communications Manager Express), 192-195
 park slot command, 184
 Partition, CUCM (Cisco Unified Communications Manager), 275
 password settings, voicemail, 344, 349-350
 PBX system, PSTN (public switched telephone network), 14
 PCPT (Pearson Cert Practice Test) engine, 441-445
 Persistent Chat, 376-377
 Phone Interface Failed Logon report, CUC (Cisco Unity Connection), 433
 Phone menu (CUC), 351

phone rings, 177
 phones, adding, CCP (Cisco Configuration Professional), 95-101
 photograph, invention of, 6
 PLAR (Private Line Automatic Ringdown), dial peers, 128-130
 Playback Message Settings menu (CUC), 352-354
 PoE (Power over Ethernet), IP phones, 235
 Port Activity report, CUC (Cisco Unity Connection), 435
 POTS (Plain Old Telephone Service) dial peers, 118, 139-141
 configuring, 120-124
 wildcards, 126-128
 powering, Cisco IP phones, 52-55
 prefix command, 136-138
 prefix digits command, 135
 presence, 366-367
 CUPC (Cisco Unified Personal Communicator), 370-373
 instant messaging, 371-372
 video calls, 372
 voice calls, 372
 CUPS (Cisco Unified Presence Server), 366-367, 370
 CUC (Cisco Unity Connection) integration, 375
 LDAP (Lightweight Directory Access Protocol), 374-375
 Microsoft Office Communications Server integration, 374-375
 Persistent Chat, 376-377
 QoS (Quality of Service), 377-378
 presence groups, CUCM (Cisco Unified Communications Manager), configuring, 307-309

presence-enabled call lists, CUCM (Cisco Unified Communications Manager), configuring, 306-307

privacy feature, CUCM (Cisco Unified Communications Manager), 291

private DLs, voicemail, 346

Private Line Automatic Ringdown (PLAR). *See* PLAR (Private Line Automatic Ringdown)

private switches, PSTN (public switched telephone network), 13

profiles, IP phones, 243

PSTN (public switched telephone network), 13-17

- backup, CAC, 268
- components, 13-14
- connections, 14-15
- North American destination patterns, 127
- numbering plans, 16-17
- PBX system, 14

public switched telephone network). *See* PSTN (public switched telephone network)

Q

QoS (Quality of Service)

- AutoQoS, 155-162
- CME (Communications Manager Express) dial-plans, 152-162
- CoS (Class of Service), 162
- CUPS (Cisco Unified Presence Server), 377-378
- requirements, 154
- ToS (Type of Service), 162
- troubleshooting, CME (Communications Manager Express), 392-395
- trust boundaries, 155

R

Real-time Transport Control Protocol (RTCP), 23-24

regions, IP phones, 241

registration

- Cisco IP phones, 65-66
- CME (Communications Manager Express), troubleshooting, 385-389
- IP phones, 236
- SIP phones, 236

Remote Destination Profiles, Mobile Connect, 317

- creating, 320-321

repeaters, analog signals, 9

reports

- CUC (Cisco Unity Connection)
 - analyzing*, 430-433
 - generating and accessing*, 427-433
 - Phone Interface Failed Logon report*, 433
 - Port Activity report*, 435
 - troubleshooting and maintenance*, 433-437
 - User Lockout reports*, 433
- CUCM (Cisco Unified Communications Manager)
 - alerts reports*, 432
 - analyzing*, 407
 - generating*, 406, 426-427

roles, CUCM (Cisco Unified Communications Manager), 221-222

Roles page (CUC), 339

Route Group, CUCM (Cisco Unified Communications Manager), 272

Route List, CUCM (Cisco Unified Communications Manager), 271-272

route patterns, call routing, 270

- Route Plan Report, unassigned directory numbers, deleting, 405
- router call processing, 130-146
- router-based DHCP servers, configuring, 61-63
- routers
 - DHCP scope, configuring on, 85
 - digit manipulation, 136
 - interfaces, AutoQoS, 160-161
- RTCP (Real-time Transport Control Protocol), 23-24
- RTMT (Real-Time Monitoring Tool), CUCM (Cisco Unified Communications Manager), 416-417

S

- scavenger applications, 155
- SCCP, CUC (Cisco Unity Connection), voicemail, 337-338
- scheduled backups, DRS (Disaster Recovery System), 419
- servicability reports (CUC), 426-430
- service parameters, CUCM (Cisco Unified Communications Manager), configuring, 325-326
- shared lines, CUCM (Cisco Unified Communications Manager), 291
 - configuring, 295
- show dial-peer voice summary command, 390
- show logging command, 200
- show policy-map interface command, 394
- signaling, CCS (common channel signaling), 12
- signals, converting analog to digital, 9-11
- single number reach, configuring, CME (Communications Manager Express), 204-206
- single-group paging, 193
- SIP (Session Initiation Protocol), CUC (Cisco Unity Connection), voicemail, 338
- SIP phones, registration, 236
- SNMP menu (Cisco Unified Serviceability interface), 218
- softkey templates, Mobile Connect, configuring, 319
- SoftPhones, 18
- SRST (Survivable Remote Site Telephony), 346
- switch router uplink, AutoQoS, 159-160
- Sync, LDAP (Lightweight Directory Access Protocol), 250
 - agreements, 250
 - mechanisms, 251
 - verifying, 253
- syntax variations, Cisco AutoQoS, 162
- System menu (CM Administration interface), 216

T

- T1 CAS PSTN interfaces, configuring, 112-114
- T1 CCS PSTN interfaces, configuring, 115-117
- tags
 - COR (Class of Restriction) lists, defining, 148
 - VLANs, 56
- telephony features, CUCM (Cisco Unified Communications Manager), 290-294
 - barge, 291, 295-296
 - BLF Speed Dials, 305-306
 - call coverage, 290, 295
 - call forwarding, 291

- call hunting, 292, 301-303
- call park, 292, 299-300
- call pickup, 291-292, 297-298
- intercom, 293, 304-305
- native presence, 293, 305
- presence groups, 307-309
- presence-enabled call lists, 306-307
- privacy, 291
- shared lines, 291, 295

TFTP (Trivial File Transfer Protocol)

- IP phones, 235
- services, 86-87

Tools menu (Cisco Unified Serviceability interface), 218

ToS (Type of Service), 162

Trace menu (Cisco Unified Serviceability interface), 217

transactional applications, 154

transfer rules, voicemail, 344

translation patterns, call routing, 270

translation profiles, digit manipulation, 141-145

troubleshooting

- CME (Communications Manager Express), 384-385
 - dial-plans*, 389-392
 - QoS (Quality of Service)*, 392-395
 - registration*, 385-389
- CUCM (Cisco Unified Communications Manager), 398-399, 402-403
 - DRS (Disaster Recovery System)*, 418-420
 - IP phone registration*, 403-405
 - reports*, 406-408
 - RTMT (Real-Time Monitoring Tool)*, 416-417
 - unassigned directory numbers*, 405

trunks

- call routing, 270
- CUCM (Cisco Unified Communications Manager), call flow, 272-273
- PSTN (public switched telephone network), 13

trust boundaries, QoS (Quality of Service), 155

U

unassigned directory numbers, deleting, 405

unified communications

- Cisco Unity Connection, 26
- CME (Communications Manager Express), 26-30
- CUCM (Cisco Unified Communications Manager), 26-31

Unified Operating System, interface, 219

Unified Serviceability, CUC (Cisco Unity Connection), 225

user accounts, Mobility, 319-320

User Lockout reports, CUC (Cisco Unity Connection), 433

User Management menu (CM Administration interface), 217

User Templates, CUC (Cisco Unity Connection), 343-344

users, adding, CCP (Cisco Configuration Professional), 95-101

V

video, network requirements, 154

video calls, CUPC (Cisco Unified Personal Communicator), 372

VLANs (Virtual LANs), 55-61

configuration, 59-61

tags, 56

voice VLANs, 58-59

*configuration, 85***voice**

network requirements, 154

packets, converting to, 18-21

voice call legs, dial peers, 118-119**voice calls, CUPC (Cisco Unified Personal Communicator), 372****voice network directories, configuring, CME (Communications Manager Express), 171-175****voice ports**

analog, configuring, 108

digital, configuring, 112-117

FXO (Foreign Exchange Office) ports, configuring, 111-112

FXS (Foreign Exchange Station) ports, configuring, 108-111

voice telephony³

PSTN (public switched telephone network), 13-17

VoIP, 17-24

voice VLANs, 58-59

configuring, 85

voicemail, CUC (Cisco Unity Connection), 332-333, 338-347

AAR (Automated Alternate Routing), 346

call actions, 345

call forwarding, 345

end users, 355-363

extensions, 345

greetings, 344

mailbox, 346, 363

message actions and settings, 345

message aging policy, 347

password settings, 344

private DLs, 346

SCCP, 337-338

SIP (Session Initiation Protocol), 338

SRST (Survivable Remote Site Telephony), 346

transfer rules, 344

user creation, 346

User Templates, 343-344

voicemail ports, call routing, 270**voice-translation profile command, 136****VoIP (Voice over IP), 3, 17-24**

bandwidth, 153

business benefits, 17-18

cabling, cost savings, 17

converting voice to packets, 18-21

delay, 153

DSPs (digital signal processors), 22-23

IP SoftPhones, 18

packet loss, 153

RTCP (Real-time Transport Control Protocol), 23-24

RTP (Real-time Transport Protocol), 23-24

traffic, enemies, 153-154

VoIP dial peers, 118

configuring, 124-126

wildcards, 126-128

W-Z

wildcards, dial peers, 126-128