Official Cert Guide
Learn, prepare, and practice for exam success

CCNA Voice
640-461

Master CCNA Voice 640-461 exam topics
Assess your knowledge with chapter-opening quizzes
Review key concepts with exam preparation tasks
Practice with realistic exam questions on the CD-ROM

JEREMY CIOARA, CCIE® NO. 11727
MICHAEL VALENTINE

ciscopress.com

FREE SAMPLE CHAPTER
SHARE WITH OTHERS
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.interfacettt.com) in Phoenix, Arizona. Likewise, Jeremy has recorded many E-Learning titles at CBTNuggets (www.cbt nuggets.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.interfacet.com), CBTNuggets (www.cbt nuggets.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

<table>
<thead>
<tr>
<th>Part</th>
<th>Title</th>
<th>Pages</th>
</tr>
</thead>
<tbody>
<tr>
<td>Introduction</td>
<td>xxi</td>
<td></td>
</tr>
<tr>
<td><strong>Part I</strong></td>
<td><strong>Voice Perspectives</strong></td>
<td></td>
</tr>
<tr>
<td>Chapter 1</td>
<td>Traditional Voice Versus Unified Voice</td>
<td>3</td>
</tr>
<tr>
<td>Chapter 2</td>
<td>Understanding the Pieces of Cisco Unified Communications</td>
<td>27</td>
</tr>
<tr>
<td>Chapter 3</td>
<td>Understanding the Cisco IP Phone Concepts and Registration</td>
<td>49</td>
</tr>
<tr>
<td><strong>Part II</strong></td>
<td><strong>Cisco Unified Communications Manager Express</strong></td>
<td></td>
</tr>
<tr>
<td>Chapter 4</td>
<td>Getting Familiar with CME Administration</td>
<td>69</td>
</tr>
<tr>
<td>Chapter 5</td>
<td>Managing Endpoint and End Users with CME</td>
<td>81</td>
</tr>
<tr>
<td>Chapter 6</td>
<td>Understanding the CME Dial-Plan</td>
<td>105</td>
</tr>
<tr>
<td>Chapter 7</td>
<td>Configuring Cisco Unified CME Voice Productivity Features</td>
<td>171</td>
</tr>
<tr>
<td><strong>Part III</strong></td>
<td><strong>Cisco Unified Communications Manager</strong></td>
<td></td>
</tr>
<tr>
<td>Chapter 8</td>
<td>Administrator and End-User Interfaces</td>
<td>217</td>
</tr>
<tr>
<td>Chapter 9</td>
<td>Managing Endpoints and End Users in CUCM</td>
<td>235</td>
</tr>
<tr>
<td>Chapter 10</td>
<td>Understanding CUCM Dial-Plan Elements and Interactions</td>
<td>269</td>
</tr>
<tr>
<td>Chapter 11</td>
<td>Enabling Telephony Features with CUCM</td>
<td>289</td>
</tr>
<tr>
<td>Chapter 12</td>
<td>Enabling Mobility Features in CUCM</td>
<td>323</td>
</tr>
<tr>
<td><strong>Part IV</strong></td>
<td><strong>Voicemail and Presence Solutions</strong></td>
<td></td>
</tr>
<tr>
<td>Chapter 13</td>
<td>Voicemail Integration with Cisco Unity Connection</td>
<td>343</td>
</tr>
<tr>
<td>Chapter 14</td>
<td>Enabling Cisco Unified Presence Support</td>
<td>377</td>
</tr>
<tr>
<td><strong>Part V</strong></td>
<td><strong>Voice Network Management and Troubleshooting</strong></td>
<td></td>
</tr>
<tr>
<td>Chapter 15</td>
<td>Common CME Management and Troubleshooting Issues</td>
<td>397</td>
</tr>
<tr>
<td>Chapter 16</td>
<td>Management and Troubleshooting of Cisco Unified Communications Manager</td>
<td>415</td>
</tr>
<tr>
<td>Chapter 17</td>
<td>Monitoring Cisco Unity Connection</td>
<td>439</td>
</tr>
<tr>
<td>Chapter 18</td>
<td>Final Preparation</td>
<td>457</td>
</tr>
</tbody>
</table>
Appendix A  Answers Appendix  463
Appendix B  640-461 CCNA Voice Exam Updates, Version 1.0  467
Appendix C  Glossary  469
Index  480
## Contents

### Introduction xx

### 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

#### 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
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

Part II Cisco Unified Communications Manager Express

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

Chapter 5 Managing Endpoint 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
Configuring Call Forwarding  179
Forwarding Calls from the IP Phone  179
Forwarding Calls from the CLI  181
Using the call-forward pattern Command to Support H.450.3  181
Configuring Call Transfer  184
Configuring Call Park  185
Configuring Call Pickup  190
Configuring Intercom  193
Configuring Paging  196
Configuring After-Hours Call Blocking  199
Configuring CDRs and Call Accounting  203
Configuring Music on Hold  207
Configuring Single Number Reach  208
Enabling the Flash-Based CME GUI  210
Exam Preparation Tasks  214

Part III  Cisco Unified Communications Manager

Chapter 8  Administrator and End-User Interfaces  217
“Do I Know This Already?” Quiz  217
Foundation Topics  220
  Describe the CUCM GUI and CLI  220
  Cisco Unified Communications Manager
     Administration Interface  220
  Cisco Unified Serviceability Administration Interface  221
  Cisco Unified Operating System Administration Interface  223
  Disaster Recovery System Interface  224
  Cisco Unified Reporting Interface  224
  CLI  224
  User Management in CUCM: Roles and Groups  225
  Describe the CUC GUI and CLI  227
  Describe the Cisco Unified Presence Server GUI and CLI  230
Exam Preparation Tasks  232

Chapter 9  Managing Endpoints and End Users in CUCM  235
“Do I Know This Already?” Quiz  235
Foundation Topics  238
  Implementing IP Phones in CUCM  238
  Special Functions and Services Used by IP Phones  238
  IP Phone Registration Process  240
Chapter 10  Understanding CUCM Dial-Plan Elements and Interactions  269

“Do I Know This Already?” Quiz  269

Foundation Topics  273

  CUCM Call Flows  273
  Call Flow in CUCM if DNS Is Used  273
  Call Flow in CUCM if DNS Is Not Used  273
  Centralized Remote Branch Call Flow  275
  Centralized Deployment PSTN Backup Call Flow  277
  Distributed Deployment Call Flow  278
  Call-Routing Sources in CUCM  280
  Call-Routing Destinations in CUCM  280
  Call-Routing Configuration Elements  281
  Call-Routing Behavior  283
  Class of Control  284

Exam Preparation Tasks  287
Chapter 11  Enabling Telephony Features with CUCM  289

“Do I Know This Already?” Quiz  289

Foundation Topics  292

  Describe Extension Mobility in CUCM  292
  Enable EM in CUCM  293
  Step 1: Activate the EM Service  293
  Step 2: Configure EM Service Parameters  293
  Step 3: Add the EM Service  294
  Step 4: Create Default Device Profiles  295
  Step 5a: Create Device Profiles  295
  Step 5b: Subscribe Device Profiles to the EM Service  296
  Step 6: Associate Users with Device Profiles  297
  Step 7a: Enable EM for Phones  298
  Step 7b: Subscribe Phones to EM Service  299

Describe Telephony Features in CUCM  300

  Call Coverage  300
  Intercom  303
  CUCM Native Presence  303
  Enable Telephony Features in CUCM  304
  Enabling Call Coverage  305
  Configuring Intercom Features  314
  Configure CUCM Native Presence  315

Exam Preparation Tasks  321

Chapter 12  Enabling Mobility Features in CUCM  323

“Do I Know This Already?” Quiz  323

Foundation Topics  326

  Understanding CUCM Mobility Features  326
  Describe Mobile Connect  326
  Unified Mobility Architecture  327
  Implementing Mobility Features in CUCM  329
  Configuring Mobile Connect  329
  Configuring MVA  336

Exam Preparation Tasks  341
Part IV  Voicemail and Presence Solutions

Chapter 13  Voicemail Integration with Cisco Unity Connection  343
“Do I Know This Already?” Quiz  343
Foundation Topics  346
  Describe Cisco Unity Connection  346
  Overview of Cisco Unity Connection  346
  Single-Site and Multisite Deployment Considerations  346
  CUC Integration Overview  347
  CUC Features  349
  Describe Cisco Unity Connection Users and Mailboxes  353
  User Templates  353
  CUC End Users  355
  User Creation Options  356
  CUC Voicemail Boxes  357
Implement Cisco Unity Connection Users and Mailboxes  357
  Configure End User Templates  357
  Configure CUC End Users  365
  Importing End Users in to CUC  367
  Managing the CUC Message Store  372
Exam Preparation Tasks  375

Chapter 14  Enabling Cisco Unified Presence Support  377
“Do I Know This Already?” Quiz  377
Foundation Topics  380
  Describe Cisco Unified Presence Features  380
  Cisco Unified Personal Communicator  380
  Cisco Unified Communications Manager IP Phone Service  383
  Cisco IP Phone Messenger  383
  Describe Cisco Unified Presence Architecture  384
  Integration with Microsoft Office Communications Server  384
  Integration with LDAP  384
  Integration with Cisco Unity Connection  385
  Integration with Conferencing Resources  385
  Integration with Calendar Resources  385
  Architecture and Call Flow: Softphone Mode  386
  Architecture and Call Flow: Deskphone Control Mode  386
  Compliance and Persistent Chat  386
CUPS and QoS Considerations 387
Enabling Cisco Unified Presence 389

Enabling End Users for Cisco Unified Personal Communicator in CUCM 389
Enabling End Users for CUPC in Cisco Unified Presence 391

Troubleshooting CUPC 392

Exam Preparation Tasks 394

Part V Voice Network Management and Troubleshooting

Chapter 15 Common CME Management and Troubleshooting Issues 397

“Do I Know This Already?” Quiz 397

Foundation Topics 400

Troubleshooting 400
Troubleshooting Common CME Registration Issues 401
Troubleshooting Dial-Plan and QoS Issues 405

Dial-Plan Issues 405
QoS Issues 408

Exam Preparation Tasks 412

Chapter 16 Management and Troubleshooting of Cisco Unified Communications Manager 415

“Do I Know This Already?” Quiz 415

Foundation Topics 418

Describe How to Provide End-User Support for Connectivity and Voice Quality Issues 418

Troubleshooting 418
Troubleshooting IP Phone Registration Problems 419
Deleting Unassigned Directory Numbers Using the Route Plan Report 421

Describe CUCM Reports and How They Are Generated 422
Understanding CUCM CDR Analysis and Reporting Tool Reports 424

CDR and CMR Architecture 426
Generating CDR Reports 427
Describe Cisco Unified RTMT 432
RTMT Interface 432
Monitoring CUCM with RTMT 433
Describe the Disaster Recovery System 434

Using the DRS 435

Exam Preparation Tasks 437
Chapter 17 Monitoring Cisco Unity Connection 439

“Do I Know This Already?” Quiz 439

Foundation Topics 442

Generating and Accessing Cisco Unity Connection Reports 442
Cisco Unity Connection Serviceability Reports 442
Cisco Unified Serviceability: Serviceability Reports Archive 445
Analyzing Cisco Unity Connection Reports 446
Troubleshooting and Maintenance Operations Using Cisco Unity Connection Reports 449

Reports to Support Routine Maintenance 451

Exam Preparation Tasks 454

Chapter 18 Final Preparation 457

Tools for Final Preparation 457
Pearson Cert Practice Test Engine and Questions on the CD 457
Cisco Learning Network 459
Chapter-Ending Review Tools 459
Suggested Plan for Final Review/Study 459
Using the Exam Engine 460
Summary 461

Appendix A Answers Appendix 463

Appendix B 640-461 CCNA Voice Exam Updates, Version 1.0 467

Appendix C Glossary 469

Index 480
Icons Used in This Book

Communication Server  PC  PC with Software  Sun Workstation  Macintosh  Terminal  ISDN/Frame Relay Switch

Token Ring  Laptop  File Server  Web Server  Ciscoworks Workstation  ATM Switch  Modem

Gateway  Access Server  IBM Mainframe  Front End Processor  Cluster Controller  Multilayer Switch without Text

Printer  Router  Bridge  Hub  DSU/CSU  DSU/CSU  FDDI  Catalyst Switch

Network Cloud  Line: Ethernet  Line: Serial  Line: Circuit-Switched  Phone  IP Phone

Repeater  PBX Switch  File Server  Cisco Unified Communications 500 Series for Small Business  Cisco Unity Express  Cisco Unified Communication Manager

Voice-Enabled Router  Voice-Enabled Workgroup Switch  Legacy PBX  Multilayer Switch without Text  Unified Personal Communicator (UPC)
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>
<thead>
<tr>
<th>Table I-1</th>
<th>640-461 ICOMM 8.0 Exam Topics</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Chapter Where Topic Is Covered Exam Topic</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Describe the characteristics of a Cisco Unified Communications solution</strong></td>
<td></td>
</tr>
<tr>
<td>Chapter 2</td>
<td>Describe the Cisco Unified Communications components and their functions</td>
</tr>
<tr>
<td>Chapter 2</td>
<td>Describe call signaling and media flows</td>
</tr>
<tr>
<td>Chapter 6</td>
<td>Describe quality implications of a VoIP network</td>
</tr>
<tr>
<td><strong>Provision end users and associated devices</strong></td>
<td></td>
</tr>
<tr>
<td>Chapter 5, Chapter 9</td>
<td>Describe user creation options for Cisco Unified Communications Manager and Cisco Unified Communications Manager Express</td>
</tr>
<tr>
<td>Chapter 9</td>
<td>Create or modify user accounts for Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>Chapter 5</td>
<td>Create or modify user accounts for Cisco Unified Communications Manager Express using the GUI</td>
</tr>
<tr>
<td>Chapter 9</td>
<td>Create or modify endpoints for Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>Chapter 5</td>
<td>Create or modify endpoints for Cisco Unified Communications Manager Express using the GUI</td>
</tr>
<tr>
<td>Chapter 6, Chapter 10</td>
<td>Describe how calling privileges function and how calling privileges impact system features</td>
</tr>
<tr>
<td>Chapter 5, Chapter 9</td>
<td>Create or modify directory numbers</td>
</tr>
<tr>
<td>Chapter 7, Chapter 11, Chapter 12</td>
<td>Enable user features and related calling privileges for extension mobility, call coverage, intercom, native presence, and unified mobility remote destination configuration</td>
</tr>
<tr>
<td>Chapter 14</td>
<td>Enable end users for Cisco Unified Presence</td>
</tr>
<tr>
<td>Chapter 7, Chapter 11, Chapter 12</td>
<td>Verify user features are operational</td>
</tr>
<tr>
<td><strong>Configure voice messaging and presence</strong></td>
<td></td>
</tr>
<tr>
<td>Chapter 13</td>
<td>Describe user creation options for voice messaging</td>
</tr>
<tr>
<td>Chapter 13</td>
<td>Create or modify user accounts for Cisco Unity Connection</td>
</tr>
<tr>
<td>Chapter 14</td>
<td>Describe Cisco Unified Presence</td>
</tr>
<tr>
<td>Chapter 14</td>
<td>Configure Cisco Unified Presence</td>
</tr>
</tbody>
</table>
Table I-1  640-461 ICOMM 8.0 Exam Topics

<table>
<thead>
<tr>
<th>Chapter Where Topic Is Covered</th>
<th>Exam Topic</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maintain Cisco Unified Communications system</td>
<td></td>
</tr>
<tr>
<td>Chapter 16</td>
<td>Generate CDR and CMR reports</td>
</tr>
<tr>
<td>Chapter 16</td>
<td>Generate capacity reports</td>
</tr>
<tr>
<td>Chapter 16</td>
<td>Generate usage reports</td>
</tr>
<tr>
<td>Chapter 16</td>
<td>Generate RTMT reports to monitor system activities</td>
</tr>
<tr>
<td>Chapter 17</td>
<td>Monitor voicemail usage</td>
</tr>
<tr>
<td>Chapter 16</td>
<td>Remove unassigned directory numbers</td>
</tr>
<tr>
<td>Chapter 16</td>
<td>Perform manual system backup</td>
</tr>
<tr>
<td>Provide end user support</td>
<td></td>
</tr>
<tr>
<td>Chapter 15, Chapter 16</td>
<td>Verify PSTN connectivity</td>
</tr>
<tr>
<td>Chapter 15, Chapter 16</td>
<td>Define fault domains using information gathered from end user</td>
</tr>
<tr>
<td>Chapter 15, Chapter 16</td>
<td>Troubleshoot endpoint issues</td>
</tr>
<tr>
<td>Chapter 17</td>
<td>Identify voicemail issues and resolve issues related to user mailboxes</td>
</tr>
<tr>
<td>Chapter 15, Chapter 16</td>
<td>Describe causes and symptoms of call quality issues</td>
</tr>
<tr>
<td>Chapter 5, Chapter 9</td>
<td>Reset single devices</td>
</tr>
<tr>
<td>Chapter 11</td>
<td>Describe how to use phone applications</td>
</tr>
</tbody>
</table>

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 page intentionally left blank
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.
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.”
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
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.

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
Chapter 3: Understanding the Cisco IP Phone Concepts and Registration

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).

![Diagram showing Power Patch Panel and Inline Coupler](image)

**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 sidecar 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

<table>
<thead>
<tr>
<th>Data</th>
<th>TCP Ports</th>
<th>Source IP Address</th>
<th>Destination IP Address</th>
<th>Source MAC Address</th>
<th>Destination MAC Address</th>
<th>VLAN Tag</th>
</tr>
</thead>
</table>

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](image-url)
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**

```plaintext
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

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

```plaintext
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
```

<table>
<thead>
<tr>
<th>VLAN Name</th>
<th>Status</th>
<th>Ports</th>
</tr>
</thead>
<tbody>
<tr>
<td>1 default</td>
<td>active</td>
<td>Gi0/1, Gi0/2</td>
</tr>
<tr>
<td>10 VOICE</td>
<td>active</td>
<td>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</td>
</tr>
<tr>
<td>50 DATA</td>
<td>active</td>
<td>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</td>
</tr>
<tr>
<td>1002 fddi-default</td>
<td>act/unsup</td>
<td></td>
</tr>
<tr>
<td>1003 token-ring-default</td>
<td>act/unsup</td>
<td></td>
</tr>
<tr>
<td>1004 fddinet-default</td>
<td>act/unsup</td>
<td></td>
</tr>
<tr>
<td>1005 trnet-default</td>
<td>act/unsup</td>
<td></td>
</tr>
</tbody>
</table>

**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 (synced), # master (unsynced), + 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

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

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

Numbers

640-461 ICOMM exam, 459

A

AAR (Automated Alternate Routing), 356
access lists
  CUCM (Cisco Unified Communications Manager), 327-328
    configuring, 332-335
  empty, 328
administration interface, CUCM (Cisco Unified Communications Manager), 220-221
Advanced Features menu (CM Administration interface), 220
after-hours call blocking, configuring, CME (Communications Manager Express), 199-204
Alarm menu (Cisco Unified Serviceability interface), 221
alerts reports, CUC (Cisco Unity Connection), 448
algorithms, queuing, 157-158
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, 7
Application menu (CM Administration interface), 221
application rules, CUC (Cisco Unity Connection), 352
Application Users, versus End Users, 254
architecture, CUPS (Cisco Unified Presence Server), 384-388
ARPT (Auto Register Phone Tool), CUCM (Cisco Unified Communications Manager), 253-254
assigning, inbound/outgoing COR lists, 150
associating, ephone/ephone-dn, 92-95
audio codec bandwidth, MOS (Mean Opinion Score), 21
authentication, LDAP (Lightweight Directory Access Protocol), 265-266
AutoQoS, 158-166
autoregistration, IP phones, 251-252

B

Bandwidth, 21
  audio codec, MOS (Mean Opinion Score), 21
barge feature, CUCM (Cisco Unified Communications Manager), 301
configuring, 305-306
base configuration, CME (Communications Manager Express), 87-88
BAT (Bulk Administration Tool), CUCM (Cisco Unified Communications Manager), 252-253
End Users, 258
Best Effort model (QoS), 155
BLF Speed Dials, CUCM (Cisco Unified Communications Manager), configuring, 315-317
boot process, Cisco IP phones, 84
Bulk Administration menu (CM Administration interface), 221
businesses, VoIP, benefits, 17-18

cabling, VoIP, cost savings, 17
CAC, PSTN backup, 278-279
calendars, CUPS (Cisco Unified Presence Server), integration, 385-386
call accounting, configuring, CME (Communications Manager Express), 203-207
call actions, voicemail, 355
call coverage, CUCM (Cisco Unified Communications Manager), 300
enabling, 305
call flow, CUCM (Cisco Unified Communications Manager), 273
call routing, 280-283
centralized deployment PSTN backup call flow, 277
centralized remote branch call flow, 275-276
CoC (Class of Service), 284-285
CSS (Calling Search Space), 285
digit analysis, 283-284
distributed deployment call flow, 278-280
DNS (Domain Name System), 273-274
gateways, 282-283
Hunt Group, 284
line-device configuration, 286
Partition, 285
PSTN backup using CAC, 278-279
Route Group, 282
Route List, 281-282
trunks, 282-283
without DNS, 273-275
call forwarding
configuring, CME (Communications Manager Express), 179-183
CUCM (Cisco Unified Communications Manager), 301
voicemail, 355
call handlers, CUC (Cisco Unity Connection), 350
call hunting, CUCM (Cisco Unified Communications Manager), 302
configuring, 311-313
call lists, presence-enabled call lists, configuring, 316-317
call park, configuring
  CME (Communications Manager Express), 185-191
  CUCM (Cisco Unified Communications Manager), 302, 309-311
call park number, call routing, 280
call pickup
  CME (Communications Manager Express), configuring, 190-193
  CUCM (Cisco Unified Communications Manager), 301-302
    configuring, 307-309
call processing
  CME (Communications Manager Express), 32
  CUCM (Cisco Unified Communications Manager), 40
  routers, 130-146
call progress tones, 110-111
call routing
  CUC (Cisco Unity Connection), 350-351
  CUCM (Cisco Unified Communications Manager)
    behavior, 283
    configuration elements, 281
    destinations, 280-281
    sources, 280
  Call Routing menu (CM Administration interface), 220
call routing rule filters, CUC (Cisco Unity Connection), 351-352
call transfer, configuring, CME
  (Communications Manager Express), 184-186
caller ID, CCP (Cisco Configuration Professional), 178
CallManager. See CUCM (Cisco Unified Communications Manager)
CAR (Call Detail Record Analysis and Reporting) tool, CUCM (Cisco Unified Communications Manager), 424-432
catalyst switch PoE, Cisco IP phones, 54
CBWFQ (Class-Based Weighted Fair Queuing) algorithm, 157
CCMCIP (Cisco Unified Communication Manager IP Phone), CUPC (Cisco Unified Personal Communicator), 383
CCP (Cisco Configuration Professional), 95-101
caller ID, 178
directory sorting, 179
CCS (common channel signaling), 12
CDP (Cisco Discovery Protocol), IP phones, 239
CDRs (Call Detail Records), 424-432
  configuring, CME (Communications Manager Express), 203-207
centralized deployment PSTN backup
call flow, CUCM (Cisco Unified Communications Manager), 277
centralized remote branch call flow,
  CUCM (Cisco Unified Communications Manager), 275-276
Cisco AutoQoS, 158-166
Cisco IP phones, 49
  boot process, 84
call routing, 280
catalyst switch PoE, 54
CDP (Cisco Discovery Protocol), 239
CM Groups, 245
CME (Communications Manager Express), interaction, 32-35
connecting, 52-55
CUCM (Cisco Unified Communications Manager), interaction, 38-41
Date/Time Groups, 246
defaults, 246
Device Pools, 245-246
DHCP (Dynamic Host Configuration Protocol), 239
DNS (Domain Name System), 239, 275
EM (Extension Mobility), enabling for, 298-300
forwarding calls from, 179-180
Hunt Groups, 284
implementing, CUCM (Cisco Unified Communications Manager), 238-247
locations, 245
Mobility features, configuring for, 330
NTP (Network Time Protocol), 238-246
phone buttons template, 247
PoE (Power over Ethernet), 239
powering, 52-55
profiles, 247
regions, 245
registration, 65-66, 240
CME (Communications Manager Express), 401-405
CUCM (Cisco Unified Communications Manager), 419-421
service activation, 241
softkey template, 247
TFTP (Trivial File Transfer Protocol), 239
VLANs, 55-61
Cisco Learning Network, 459
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, 223
Cisco Unified Presence, 44-45
Cisco Unified Reporting, interface, 224
Cisco Unified Serviceability, interface, 221-222
Cisco Unity Connection, 41-44
CUCM (Cisco Unified Communications Manager), interaction, 43-44
Class-Based Weighted Fair Queuing (CBWFQ) algorithm, 157
CUCM (Cisco Unified Communications Manager), 224-225
forwarding calls from, 181
clocks, setting, NTP, 63-65
CM Administration interface, 220-221
CM Groups, IP phones, 245
CME (Communications Manager Express), 30-37, 69, 171, 397
after-hours call blocking, configuring, 199-204
call accounting, configuring, 203-207
call forwarding, configuring, 179-183
call park, configuring, 185-191
call pickup, configuring, 190-193
call processing, 32
call transfer, configuring, 184-186
CCP (Cisco Configuration Professional), 95-101
CDRs (Call Detail Records), configuring, 203-207
Cisco IP phones, interaction, 32-35
configuration, base, 87-88
CTI (Computer Telephony Integration), 32
CUE (Cisco Unity Express), 32, 35-37
device control, 32
dial-plan
  configuring dial peers, 117-130
  COR (Class of Restriction) lists, 104, 146-152
dial-peers, 104
digit manipulation, 104
QoS (Quality of Service), 104, 152-166
router call processing, 104
troubleshooting, 405-408
voice port configuration, 108-117
end users, 81
endpoints, 81
ephone-dn, 95
ephones, 95
flash-based GUI, enabling, 210-213
foundation, ensuring, 84-88
intercom, configuring, 193-196
local directory service, 32
managing
  command line, 71-73
  GUIs (Graphic User Interface), 73-77
MoH (Music on Hold), configuring, 207-208
paging, configuring, 196-200
single number reach, configuring, 208-210
TFTP services, 86-87
troubleshooting, 400-401
dial-plans, 405-408
QoS (Quality of Service), 408-411
registration, 401-405
voice network directories, configuring, 175-180
CO switches, PSTN (public switched telephone network), 13
CoC (Class of Service), CUCM (Cisco Unified Communications Manager), call flow, 284-285
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
  debug voip dialpeer, 407
  park slot, 187-188
  show dial-peer voice summary, 406
  show logging, 204
  show policy-map interface, 410
common channel signaling (CCS), 12
Communications Manager Express (CME). See CME (Communications Manager Express)
conferencing resources, CUPC (Cisco Unified Personal Communicator), 385-386
configuration
CME (Communications Manager Express)
  after-hours call blocking, 199-204
  base, 87-88
  call accounting, 203-207
  call forwarding, 179-183
  call park, 185-191
  call pickup, 190-193
  call transfer, 184-186
  CDRs (Call Detail Records), 203-207
CUC (Cisco Unity Connection), 193-196
MoH (Music on Hold), 207-208
paging, 196-200
single number reach, 208-210
voice network directories, 175-180
CUCM (Cisco Unified Communications Manager)
barge, 305-306
BLF Speed Dials, 315-317
call hunting, 311-313
call park, 309-311
call pickup, 307-309
intercom, 314-316
native presence, 315
presence groups, 317-320
presence-enabled call lists, 316-317
service parameters, 335-336
shared lines, 305
dial peers, CME (Communications Manager Express), 117-130
dphone-dns, 89-90
ephones, 90-92
IP phones, 248-251
router-based DHCP servers, 61-63
routers, DHCP scope, 85
VLANs, 59-61
voice ports, CME (Communications Manager Express), 108-117
voice VLANs, 85
congestion avoidance, QoS (Quality of Service), 156
congestion management, QoS (Quality of Service), 156
connections
analog, 6-9
digital, 9-12
PSTN (public switched telephone network), 14-15
COR (Class of Restriction) lists
CME (Communications Manager Express), 104
defining tags, 148
implementing, 146-152
incoming COR lists, creating, 149
outgoing COR lists, creating, 149
Credential Policy, CUCM (Cisco Unified Communications Manager), 255
CSF (Client Services Framework), CUPC (Cisco Unified Personal Communicator), 383
CSS (Calling Search Space), CUCM (Cisco Unified Communications Manager), 285
CTI (Computer Telephony Integration), CME (Communications Manager Express), 32
CUC (Cisco Unity Connection)
application rules, 352
call handlers, 350
call routing, 350-351
call routing rule filters, 351-352
CUPS (Cisco Unified Presence Server), 385
dial-plans, 352
direct routing rules, 351
DLs (Distribution Lists), 352
forwarded routing rules, 351-352
interfaces, 227-230
monitoring, 439
notification devices, 364-365
reports
alerts reports, 448
analyzing, 446-449
generating and accessing, 442-449
Phone Interface Failed Logon report, 450
Port Activity report, 451
serviceability reports, 442-443
troubleshooting and maintenance, 449-453
User Lockout reports, 450
system settings
General Configuration page, 349
Roles page, 349
Unified Serviceability, 229
voicemail, 343, 346-357
AAR (Automated Alternate Routing), 356
call actions, 355
call forwarding, 355
distributed deployment, 355
end users, 365-374
extensions, 355
greetings, 354
mailboxes, 356-374
message actions and settings, 355
message aging policy, 357
password settings, 354
private DLs, 356
SCCP, 347-348
SIP (Session Initiation Protocol), 348
SRST (Survivable Remote Site Telephony), 356
transfer rules, 354
user creation, 356
User Templates, 353-354

CUCM (Cisco Unified Communications Manager), 31, 37-41
assigning license capabilities, 389
barge feature, 301
configuring, 305-306
BAT (Bulk Administration Tool), 252-253
BLF Speed Dials, configuring, 315-317
call coverage, 300
enabling, 305
call flow, 273
call forwarding, 301
configuring, 311-313
call hunting, 302
configuring, 311-313
call park, 302
configuring, 309-311
call pickup, 301-302
configuring, 307-309
call processing, 40
call routing
configuration elements, 281
destinations, 280-281
sources, 280
CoC (Class of Service), 284-285
CSS (Calling Search Space), 285
digit analysis, 283-284
distributed deployment call flow, 278-280
DNS (Domain Name System), 273-274
gateways, 282-283
Hunt Group, 284
line-device configuration, 286
Partition, 285
PSTN backup using CAC, 278-279
Route Group, 282
Route List, 281-282
trunks, 282-283
without DNS, 273-275
CAR (Call Detail Record Analysis and Reporting) tool, 424-432
CDRs (Call Detail Records), 424-432
Cisco IP phones, interaction, 38-41
Cisco Unity Connection, interaction, 43-44
database replication, 38-41
dial-plans, 269-282
digit-by-digit analysis, 284
EM (Extension Mobility), 292-316
  enabling in, 293-300
End Users, 235, 254-257
  BAT (Bulk Administration Tool), 258
    configuring, 389-390
  Credential Policy, 255
    implementing, 257-266
  LDAP (Lightweight Directory Access Protocol) authentication, 265-266
  LDAP (Lightweight Directory Access Protocol) integration, 258-261
    manual entry, 257-258
endpoints, 235
groups, 226-227
intercom, 303
  configuring, 314-316
interfaces, 220
  administration, 220-221
  Cisco Unified Reporting, 224
  Cisco Unified Serviceability, 221-222
CLI (command line interface), 224-225
DRS (Disaster Recovery System), 224
Unified Operating System, 223
IP phones
  configuration requirements, 244-247
  implementing, 238-247
  service activation, 241
Mobility features, 323
  access lists, 327-328, 332-335
    implementing, 329-339
  IP phones, 330
  Mobile Connect, 326-327
  MVA (Mobile Voice Access), 328
    service parameters, 335-336
    user accounts, 329-331
MVA (Mobile Voice Access), configuring, 336-339
native presence, 303-304
  configuring, 315
phones
  adding, 247-248
  ARPT (Auto Register Phone Tool), 253-254
    autoregistration, 251-252
    manual configuration, 248-251
presence architecture, 303-304
presence groups, configuring, 317-320
presence-enabled call lists, configuring, 316-317
privacy feature, 301
reports
  analyzing, 423
  generating, 422-424
roles, 225-226
shared lines, 301
  configuring, 305
troubleshooting, 415-418
  DRS (Disaster Recovery System), 434-436
  IP phone registration, 419-421
  reports, 422-425
  RTMT (Real-Time Monitoring Tool), 432-433
unassigned directory numbers, deleting, 421
CUE (Cisco Unity Express)
  CME (Communications Manager Express), 32-37
  modules, 36
CUPC (Cisco Unified Personal Communicator), 380-383
  CCMCIP (Cisco Unified Communication Manager IP Phone), 383
  CSF (Client Services Framework), 383
defining CCMCIP profile, 392
desk phone control, 391
  Desktop Control mode, 386
  enabling, 389-393
  Enterprise Instant Messaging, 381-382
  integration support, 382
  IPPM (IP Phone Messenger), 383-384
  LDAP (Lightweight Directory Access Protocol), 384-385
  Microsoft Office Communications Server integration, 384
  Persistent Chat, 386-387
  QoS (Quality of Service), 387-388
  Custom Filters, LDAP (Lightweight Directory Access Protocol), 266

CUPS (Cisco Unified Presence Server), 377-376, 380-383
calendar resource integration, 385-386
  conferencing resources, 385-386
  CUC (Cisco Unity Connection), 385
  CUPC (Cisco Unified Personal Communicator), 380-383
  Desktop Control mode, 386
  enabling, 389-393
  Softphone mode, 386
troubleshooting, 392-393
  interfaces, 230-231
  LDAP (Lightweight Directory Access Protocol), 384-385
  Microsoft Office Communications Server integration, 384
  Persistent Chat, 386-387
  QoS (Quality of Service), 387-388
  Custom Filters, LDAP (Lightweight Directory Access Protocol), 266

D
data, network requirements, 154-155
data traffic requirements, CME (Communications Manager Express), 154
database replication, CUCM (Cisco Unified Communications Manager), 38-41
debug voip dialpeer command, 407
desk phone control, CUPC (Cisco Unified Personal Communicator), 391
  Deskphone mode (CUPC), 380-381
  Desktop Control mode (CUPC), 386
device control, CME (Communications Manager Express), 32
  Device menu (CM Administration interface), 221
Device Pools, IP phones, 245-246

DHCP (Dynamic Host Configuration Protocol)
  IP phones, 239
  router IOS, configuring in, 244
  server configuration, 241-243

DHCP scope, routers, configuring on, 85

dial peers, 130-131
  CME (Communications Manager Express), 104
  configuring, 117-130
  matching inbound and outbound, 132-146
  verifying, 121
  VoIP, configuring, 124-126
  wildcards, 126-128

dial-plans
  CME (Communications Manager Express), 105-104
    configuring dial peers, 117-130
  COR (Class of Restriction) lists, 104, 146-152
  digit manipulation, 104
  QoS (Quality of Service), 104, 152-166
  router call processing, 104
  troubleshooting, 405-408
  voice port configuration, 108-117
  voice-port, 105

CUC (Cisco Unity Connection), 352

CUCM (Cisco Unified Communications Manager), 269-282
  call flow, 273-286

DiffServ (Differentiated Services), QoS (Quality of Service), 155

digit analysis, CUCM (Cisco Unified Communications Manager), 283-284

digit manipulation, 130-146
  CME (Communications Manager Express), 104

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), 284
direct routing rules, CUC (Cisco Unity Connection), 351
directory lookups
  LDAP (Lightweight Directory Access Protocol), CUPC (Cisco Unified Personal Communicator), 391
directory number (DN), call routing, 280
directory numbers, adding, CCP (Cisco Configuration Professional), 95-101

distributed deployment call flow, CUCM (Cisco Unified Communications Manager), 278-280

DLs (Distribution Lists)
  CUC (Cisco Unity Connection), 352
  voicemail, 356

DN (directory number), call routing, 280

DNS (Domain Name System)
  CUCM (Cisco Unified Communications Manager), call flows, 273-274

IP phones, 239

DRS (Disaster Recovery System), 224

CUCM (Cisco Unified Communications Manager), 434-436
  scheduled backups, 435
DSPs (digital signal processors), 22-23
  chips, 22
  quantity, calculating, 22

E

Edison, Thomas, 6

EM (Extension Mobility)
  CUCM (Cisco Unified Communications Manager), 292-316
    enabling in, 293-300
  device profiles
    associating users with, 297-298
    creating, 295-296
    creating defaults, 295
    subscribing to EM service, 296-297
  IP phones, enabling for, 298-299
  service
    activating, 293
    adding, 294
    configuring, 293

End Users
  CME (Communications Manager Express), 81
  CUC (Cisco Unity Connection), 365-374
  CUCM (Cisco Unified Communications Manager), 235, 254-257
    BAT (Bulk Administration Tool), 258
    configuring, 389-390
    implementing, 257-266
    LDAP (Lightweight Directory Access Protocol) authentication, 265-266
    LDAP (Lightweight Directory Access Protocol) integration, 258-261
    manual entry, 257-258
    versus Application Users, 254

endpoints
  CME (Communications Manager Express), 81
  CUCM (Cisco Unified Communications Manager), 235
  Enterprise Instant Messaging, CUPC (Cisco Unified Personal Communicator), 381-382
  ephone-dns, 95
    associating, 92-95
    configuring, 89-90
  ephones, 95
    associating, 92-95
    configuring, 90-92
  Extension Mobility (EM). See EM (Extension Mobility)
  extensions, voicemail, 355

F

flash-based GUI, CME (Communications Manager Express), enabling, 210-213
forwarded routing rules, CUC (Cisco Unity Connection), 351
FXO (Foreign Exchange Office) ports, configuring, 111-117
FXS (Foreign Exchange Station) ports, configuring, 108-111

G

G.711 codec, 20
G.729 codec, 21
gateways
call routing, 280
CUCM (Cisco Unified Communications Manager), call flow, 282-283
General Configuration page (CUC), 349
generating, CUC reports, 442-449
greetings, voicemail, 354
groups, CUCM (Cisco Unified Communications Manager), 226-227
GUIs (Graphic User Interface)
CME (Communications Manager Express), managing, 73-77, 210-213
CUC (Cisco Unity Connection), 227-229
CUCM (Cisco Unified Communications Manager), 220-221
CUPS (Cisco Unified Presence), 230-231

H
header compression, 157
Help menu (Cisco Unified Serviceability interface), 222
Help menu (CM Administration interface), 221
Hunt Groups, 284
hunt pilot, call routing, 280

I-J
IM (Instant Messaging), CUPC (Cisco Unified Personal Communicator), 381-382
inbound dial peers, outbound dial peers, matching, 132-146
incoming COR lists
assigning, 150
creating, 149
Instant Messaging, CUPC (Cisco Unified Personal Communicator), 381-382
intercom
CME (Communications Manager Express), configuring, 193-196
CUCM (Cisco Unified Communications Manager), 303
configuring, 314-316
interfaces
CUC (Cisco Unity Connection), 227-230
CUCM (Cisco Unified Communications Manager), 220
administration, 220-221
Cisco Unified Reporting, 224
Cisco Unified Serviceability, 221-222
CLI (command line interface), 224-225
DRS (Disaster Recovery System), 224
Unified Operating System, 223
CUPS (Cisco Unified Presence), interfaces 230-231
RTMT (Real-Time Monitoring Tool), 432-433
IntServ (Integrated Services) model, QoS (Quality of Service), 155
IP phones, 49-60
adding, CUCM (Cisco Unified Communications Manager), 247-248
autoregistration, 251-252
boot process, 84
call routing, 280
CDP (Cisco Discovery Protocol), 239
CM Groups, 245
CME (Communications Manager Express), interaction, 32-35
connecting, 52-55
CUCM (Cisco Unified Communications Manager), 38-41
Date/Time Groups, 246
defaults, 246
Device Pools, 245-246
DHCP (Dynamic Host Configuration Protocol), 239
DNS (Domain Name System), 239-275
EM (Extension Mobility), enabling for, 298-300
forwarding calls from, 179-180
Hunt Groups, 284
implementing, CUCM (Cisco Unified Communications Manager), 238-247
locations, 245
manual configuration, 248-251
Mobility features, configuring for, 330
NTP (Network Time Protocol), 238, 246
phone buttons template, 247
PoE (Power over Ethernet), 239
profiles, 247
regions, 245
registration, 65-66, 240
CME (Communications Manager Express), 401-405
CUCM (Cisco Unified Communications Manager), 419-421
service activation, 241
softkey template, 247
TFTP (Trivial File Transfer Protocol), 239
VLANs (virtual LANs), 55-61
IP SoftPhones, 18
IPPM (IP Phone Messenger), CUPC (Cisco Unified Personal Communicator), 383-384

K-L

key topics, studying,

LDAP (Lightweight Directory Access Protocol)
CUCM (Cisco Unified Communications Manager)
  authentication, 265-266
  integration, 258-261
  Sync agreements, 261-265
  synchronization, 259
CUPC (Cisco Unified Personal Communicator), directory lookups, 391
CUPS (Cisco Unified Presence Server), 384-385
Custom Filters, 266
LFI (Link Fragmentation and Interleaving), 157
licensing capabilities, CUCM (Cisco Unified Communications Manager), 389
link efficiency, QoS (Quality of Service), 156-157
Link Fragmentation and Interleaving (LFI), 157
LLQ (Low Latency Queuing) algorithm, 157
local directory service, CME (Communications Manager Express), 32
local loops, PSTN (public switched telephone network), 13
locations, IP phones, 245
Low Latency Queuing (LLQ) algorithm, 157
M

mailboxes, voicemail, 356-357
   CUC (Cisco Unity Connection), 357-374
   password settings, 359-360
manual configuration, IP phones, 248-251
manual entry, End Users, 257-258
matching, inbound/outbound data peers, 132-146
Mean Opinion Score (MOS), audio codec bandwidth, 21
Media Resources menu (CM Administration interface), 220
meet-me number, call routing, 280
message aging policy, voicemail, 357
Message Settings page (CUC), 360
Microsoft Office Communications Server, CUPS (Cisco Unified Presence Server), 384
Mobile Connect
   configuring, 329
   CUCM (Cisco Unified Communications Manager), 326-327
Remote Destination Profiles, 327
   creating, 330-332
   softkey templates, configuring, 329-330
Mobility features (CUCM), 323
   access lists, 327-328, 332-335
   implementing, 329-339
   IP phones, 330
   Mobile Connect, 326-327
MVA (Mobile Voice Access), 328, 336-339
   service parameters, 335-336
   user accounts, 329-331
modules, CUE (Cisco Unity Express), 36
MoH (Music on Hold), configuring,
   CME (Communications Manager Express), 207-208
monitoring, CUC (Cisco Unity Connection), 439
MOS (Mean Opinion Score), audio codec bandwidth, 21
multiple-group paging, 198
MVA (Mobile Voice Access), CUCM (Cisco Unified Communications Manager), 328
   configuring, 336-339

N

native presence, CUCM (Cisco Unified Communications Manager), 303-304
   configuring, 315
network requirements
   data, 154-155
   video/voice, 154
notification devices, CUC (Cisco Unity Connection), 364-365
NTP (Network Time Protocol)
   clocks, setting, 63-65
   IP phones, 238, 246
numbering plans, PSTN (public switched telephone network), 16-17
Nyquist, Harry, 18

O

operating modes, CUPC (Cisco Unified Personal Communicator), 380-381
outbound dial peers, inbound dial peers, matching, 132-146
outgoing COR lists
   assigning, 150
   creating, 149
packets, voice, converting from, 18-21
paging, CME (Communications Manager Express), configuring, 196-200
park slot command, 187-188
Partition, CUCM (Cisco Unified Communications Manager), 285
password settings, voicemail, 354, 359-360
payload compression, 157
PBX system, PSTN (public switched telephone network), 14
PCPT (Pearson Cert Practice Test) engine, 457-461
Persistent Chat, CUPS (Cisco Unified Presence Server), 386-387
personal voice mail access, CUPC (Cisco Unified Personal Communicator), 391
Phone Interface Failed Logon report, CUC (Cisco Unity Connection), 450
Phone menu (CUC), 361
phone rings, 181
phones
  adding
    CCP (Cisco Configuration Professional), 95-101
    CUCM (Cisco Unified Communications Manager), 247-248
  autoregistration, 251-252
  manual configuration, 248-251
photograph, invention of, 6
Playback Message Settings menu (CUC), 362-364
PoE (Power over Ethernet), IP phones, 239
Port Activity report, CUC (Cisco Unity Connection), 451

POTS (Plain Old Telephone Service) dial peers, configuring, 120-124
powering, Cisco IP phones, 52-55
Presence, 376-377
  CUPS (Cisco Unified Presence Server), 376-377, 380-383
  architecture, 384-388
  CUPC (Cisco Unified Personal Communicator), 380-383, 389-393
presence groups, CUCM (Cisco Unified Communications Manager), configuring, 317-320
presence-enabled call lists, CUCM (Cisco Unified Communications Manager), configuring, 316-317
privacy feature, CUCM (Cisco Unified Communications Manager), 301
private DLs, voicemail, 356
private line automatic ringdown, 128-130
private switches, PSTN (public switched telephone network), 13
profiles, IP phones, 247
PSTN (public switched telephone network), 13-17
  backup, CAC, 278-279
  components, 13-14
  connections, 14-15
  numbering plans, 16-17
  PBX system, 14

Q

QoS (Quality of Service)
  applying, 158
  Best Effort model, 155
  Cisco AutoQoS, 158-166
  classification and marking tools, 155
CME (Communications Manager Express), 104
dial-plans, 152-166
mechanisms, 155-156
congestion avoidance, 156
congestion management, 156
CUPS (Cisco Unified Presence Server), 387-388
DiffServ (Differentiated Services), 155
IntServ (Integrated Services) model, 155
link efficiency, 156-157
policing, 156
shaping, 156
troubleshooting, CME (Communications Manager Express), 408-411
queuing algorithms, 157-158

Real-time Transport Control Protocol (RTCP). See RTCP (Real-time Transport Control Protocol)
regions, IP phones, 245
registration
Cisco IP phones, 65-66
CME (Communications Manager Express), troubleshooting, 401-405
IP phones, 240
SIP phones, 240
Remote Destination Profiles, Mobile Connect, 327
creating, 330-332
repeaters, analog signals, 9
reports
CUC (Cisco Unity Connection)
  analyzing, 446-449
generating and accessing, 444-449
Phone Interface Failed Logon report, 450
Port Activity report, 451
troubleshooting and maintenance, 449-453
User Lockout reports, 450
CUCM (Cisco Unified Communications Manager)
  alerts reports, 448
  analyzing, 423
  generating, 422-424, 442-443
roles, CUCM (Cisco Unified Communications Manager), 225-226
Roles page (CUC), 349
Route Group, CUCM (Cisco Unified Communications Manager), 282
Route List, CUCM (Cisco Unified Communications Manager), 281-282
route patterns, call routing, 280
Route Plan Report, unassigned directory numbers, deleting, 421
router call processing, CME (Communications Manager Express), 104
router-based DHCP servers, configuring, 61-63
routers
call processing, 130-146
DHCP scope, configuring on, 85
RTCP (Real-time Transport Control Protocol), 23-24
RTMT (Real-Time Monitoring Tool), CUCM (Cisco Unified Communications Manager), 432-433
RTP (Real-time Transport Protocol), 23-24
SCCP (Skinny Call Control Protocol), CUC (Cisco Unity Connection), voicemail, 347-348
scheduled backups, DRS (Disaster Recovery System), 435
servers, CUPS (Cisco Unified Presence Server), 380-383
architecture, 384-388
CUPC (Cisco Unified Personal Communicator), 380-383, 389-393
servicability reports (CUC), 442-447
service parameters, CUCM (Cisco Unified Communications Manager), configuring, 335-336
shaping QoS (Quality of Service), 156
shared lines, CUCM (Cisco Unified Communications Manager), 301
configuring, 305
show dial-peer voice summary command, 406
show logging command, 204
show policy-map interface command, 410
signaling, CCS (common channel signaling), 12
signals, converting analog to digital, 9-11
single number reach, configuring, CME (Communications Manager Express), 208-210
single-group paging, 197
SIP (Session Initiation Protocol)
  CUC (Cisco Unity Connection), voicemail, 348
  phones, registration, 240
SNMP menu (Cisco Unified Serviceability interface), 222
softkey templates, Mobile Connect, configuring, 329-330
Softphone mode (CUPC), 381, 386
SoftPhones, 18
SRST (Survivable Remote Site Telephony), 356
Sync agreements, LDAP (Lightweight Directory Access Protocol), 261-265
System menu (CM Administration interface), 220
T1 CCS PSTN interfaces, configuring, 115-117
tags, VLANs, 57
telephony features, CUCM (Cisco Unified Communications Manager), 300-304
  barge, 301, 305-306
  BLF Speed Dials, 315-317
  call coverage, 300, 305
  call forwarding, 301
  call hunting, 302, 311-313
  call park, 302, 309-311
  call pickup, 301-302, 307-309
  intercom, 303, 314-316
  native presence, 303-304, 315
  presence groups, 317-320
  presence-enabled call lists, 316-317
  privacy, 301
  shared lines, 301, 305
TFTP (Trivial File Transfer Protocol) services, 86-87
IP phones, 239
Tools menu (Cisco Unified Serviceability interface), 222
Trace menu (Cisco Unified Serviceability interface), 221
transfer rules, voicemail, 354
translation patterns, call routing, 280
troubleshooting
CME (Communications Manager Express), 400-401
dial-plans, 405-408
QoS (Quality of Service), 408-411
registration, 401-405
CUCM (Cisco Unified Communications Manager), 415-419
DRS (Disaster Recovery System), 434-436
IP phone registration, 419-421
reports, 422-425
RTMT (Real-Time Monitoring Tool), 432-433
unassigned directory numbers, 421

trunks
call routing, 280
CUCM (Cisco Unified Communications Manager), call flow, 282-283
PSTN (public switched telephone network), 13

U
unassigned directory numbers, deleting, 421
unified communications
CME (Communications Manager Express), 37
CUCM (Cisco Unified Communications Manager), 31
Unified Operating System, interface, 223
Unified Serviceability, CUC (Cisco Unity Connection), 229
user accounts, Mobility, 329-331
user locale, IP phones, 256
User Lockout reports, CUC (Cisco Unity Connection), 450

User Management menu (CM Administration interface), 221
User Templates, CUC (Cisco Unity Connection), 353-354
users, adding, CCP (Cisco Configuration Professional), 95-101

V-Z
verifying dial peers, 121
video calls, CUPC (Cisco Unified Personal Communicator), 382
video requirements
CME (Communications Manager Express), 154
networks, 154
VLANs (virtual LANs), 55-61
configuration, 59-61
tags, 57
voice VLANs, 58-59
configuration, 85
voice, packets, converting to, 18-21
voice call legs, 119-120
voice calls, CUPC (Cisco Unified Personal Communicator), 382
voice network directories, configuring, CME (Communications Manager Express), 175-180
voice ports, configuring, CME (Communications Manager Express), 108-117
voice requirements
CME (Communications Manager Express), 154
networks, 154
voice telephony, 3
PSTN (public switched telephone network), 13-17
VoIP, 17-24
voice VLANs, 58-59
   configuring, 85
voicemail, CUC (Cisco Unity Connection), 349-357
   AAR (Automated Alternate Routing), 356
   call actions, 355
   call forwarding, 355
   end users, 365-374
   extensions, 355
   greetings, 354
   mailboxes, 357-374
      message actions and settings, 355
      message aging policy, 357
      password settings, 354
      private DLs, 356
      SCCP, 347-348
      SIP (Session Initiation Protocol), 348
      SRST (Survivable Remote Site Telephony), 356
      transfer rules, 354
      user creation, 356
      User Templates, 353-354
voicemail ports, call routing, 280
voice-port, CME (Communications Manager Express), 105
VoIP (Voice over IP), 3, 17-24
   business benefits, 17-18
   cabling, cost savings, 17
   converting voice to packets, 18-21
   DSPs (digital signal processors), 22-23
   IP SoftPhones, 18
   RTCP (Real-time Transport Control Protocol), 23-24
   RTP (Real-time Transport Protocol), 23-24