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CCNA Voice 640-461

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MICHAEL VALENTINE

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CCNA Voice 640-461

Official Cert Guide

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Dedication

From Jeremy D. Cioara:

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

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

From Mike Valentine:

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

In memory of my Dad.

Acknowledgments

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

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

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

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

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

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

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

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

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

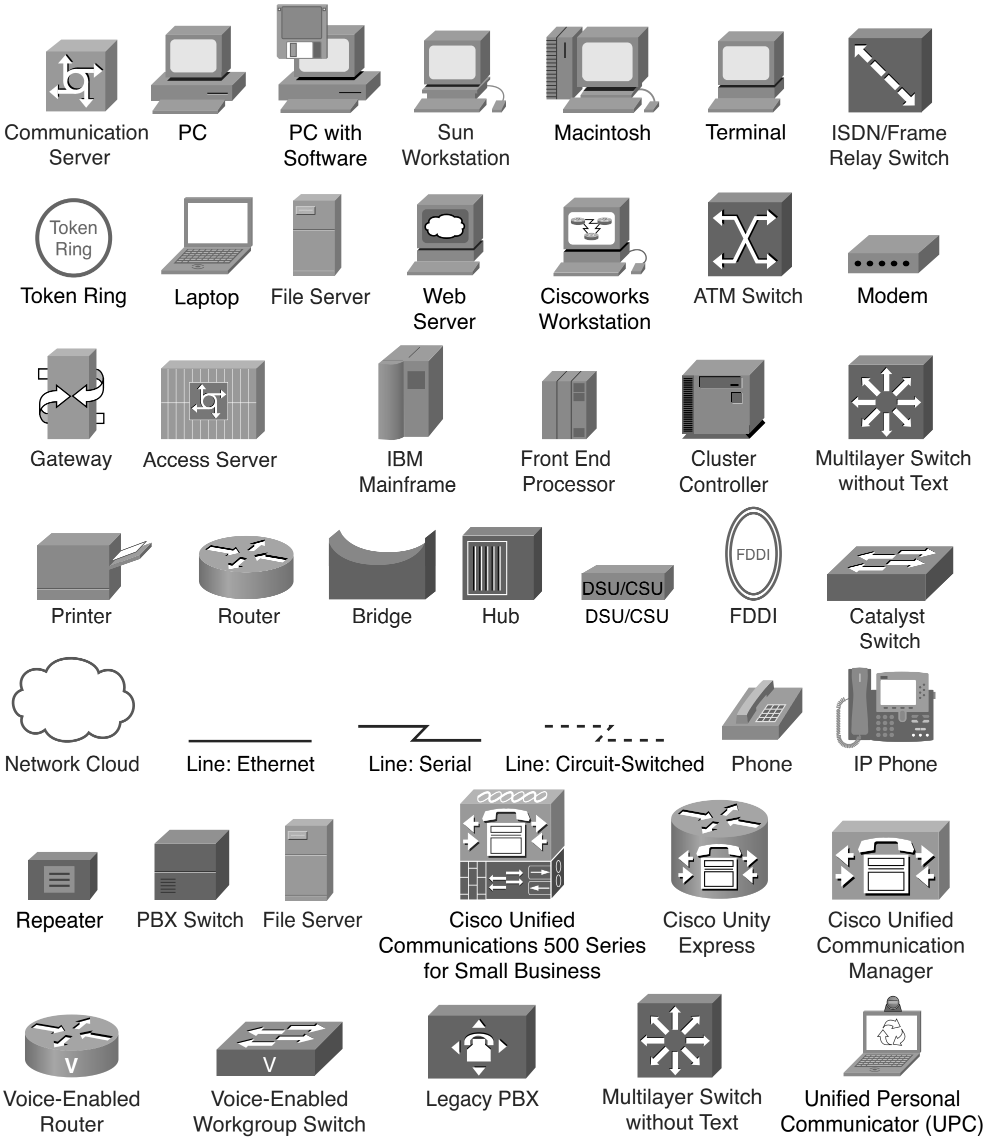
My family: Thank you so much for your support, patience, your love, and your belief in me.

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

Contents at a Glance

Contents

Icons Used in This Book



Command Syntax Conventions

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

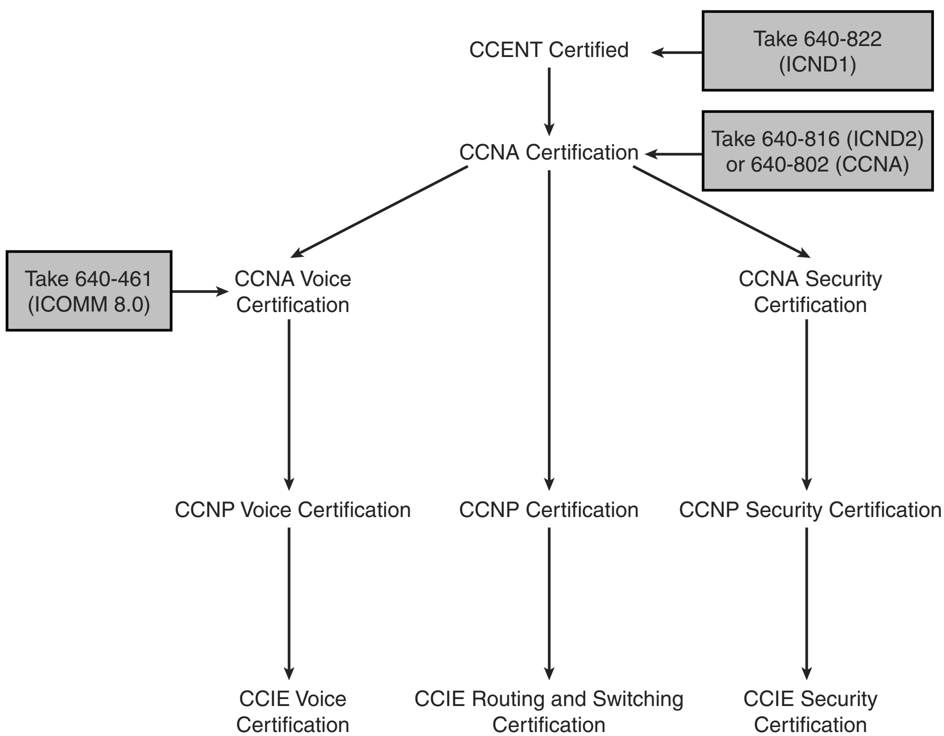
- **Boldface** indicates commands and keywords that are entered literally as shown. In actual configuration examples and output (not general command syntax), boldface indicates commands that are manually input by the user (such as a **show** command).
- *Italics* indicate arguments for which you supply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

Introduction

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

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

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



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

Goals and Methods

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

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

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

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

Who Should Read This Book?

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

Strategies for Exam Preparation

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

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

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

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

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

640-461 ICOMM 8.0 Exam Topics

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

Table I-1 *640-461 ICOMM 8.0 Exam Topics*

Chapter Where Topic Is Covered	Exam Topic
Describe the characteristics of a Cisco Unified Communications solution	
Chapter 2	Describe the Cisco Unified Communications components and their functions
Chapter 2	Describe call signaling and media flows
Chapter 6	Describe quality implications of a VoIP network
Provision end users and associated devices	
Chapter 5, Chapter 9	Describe user creation options for Cisco Unified Communications Manager and Cisco Unified Communications Manager Express
Chapter 9	Create or modify user accounts for Cisco Unified Communications Manager
Chapter 5	Create or modify user accounts for Cisco Unified Communications Manager Express using the GUI
Chapter 9	Create or modify endpoints for Cisco Unified Communications Manager
Chapter 5	Create or modify endpoints for Cisco Unified Communications Manager Express using the GUI
Chapter 6, Chapter 10	Describe how calling privileges function and how calling privileges impact system features
Chapter 5, Chapter 9	Create or modify directory numbers
Chapter 7, Chapter 11, Chapter 12	Enable user features and related calling privileges for extension mobility, call coverage, intercom, native presence, and unified mobility remote destination configuration
Chapter 14	Enable end users for Cisco Unified Presence
Chapter 7, Chapter 11, Chapter 12	Verify user features are operational
Configure voice messaging and presence	
Chapter 13	Describe user creation options for voice messaging
Chapter 13	Create or modify user accounts for Cisco Unity Connection
Chapter 14	Describe Cisco Unified Presence
Chapter 14	Configure Cisco Unified Presence

Table I-1 *640-461 ICOMM 8.0 Exam Topics*

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

How This Book Is Organized

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

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

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

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

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

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

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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.

CHAPTER 3

Understanding the Cisco IP Phone Concepts and Registration

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

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

"Do I Know This Already?" Quiz

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

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

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

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

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

Foundation Topics

Connecting and Powering Cisco IP Phones

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

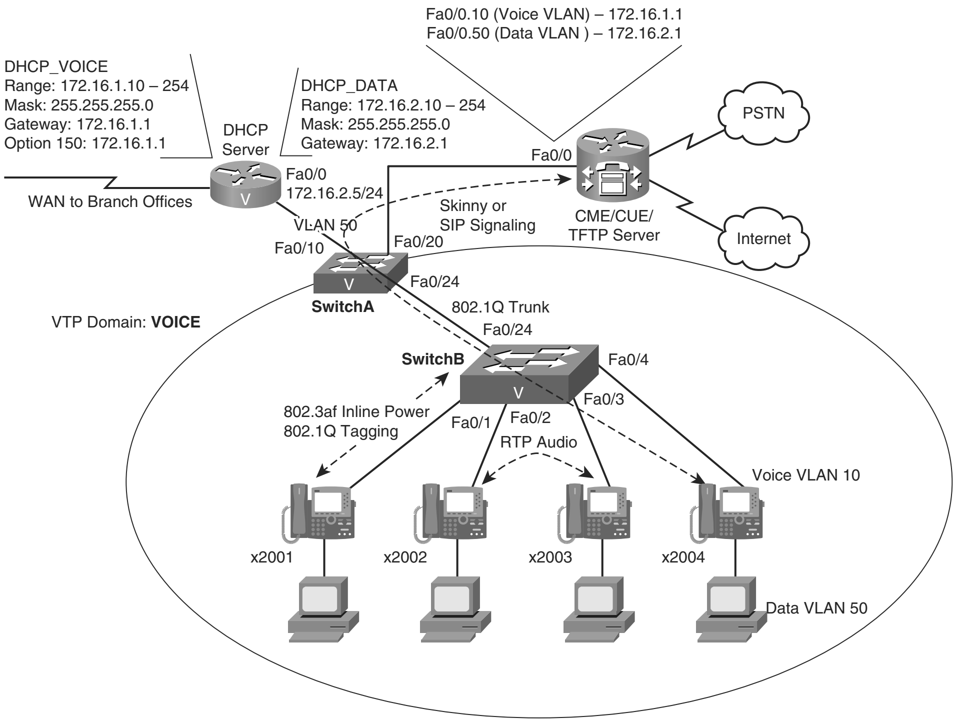


Figure 3-1 *VoIP Network*

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

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

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

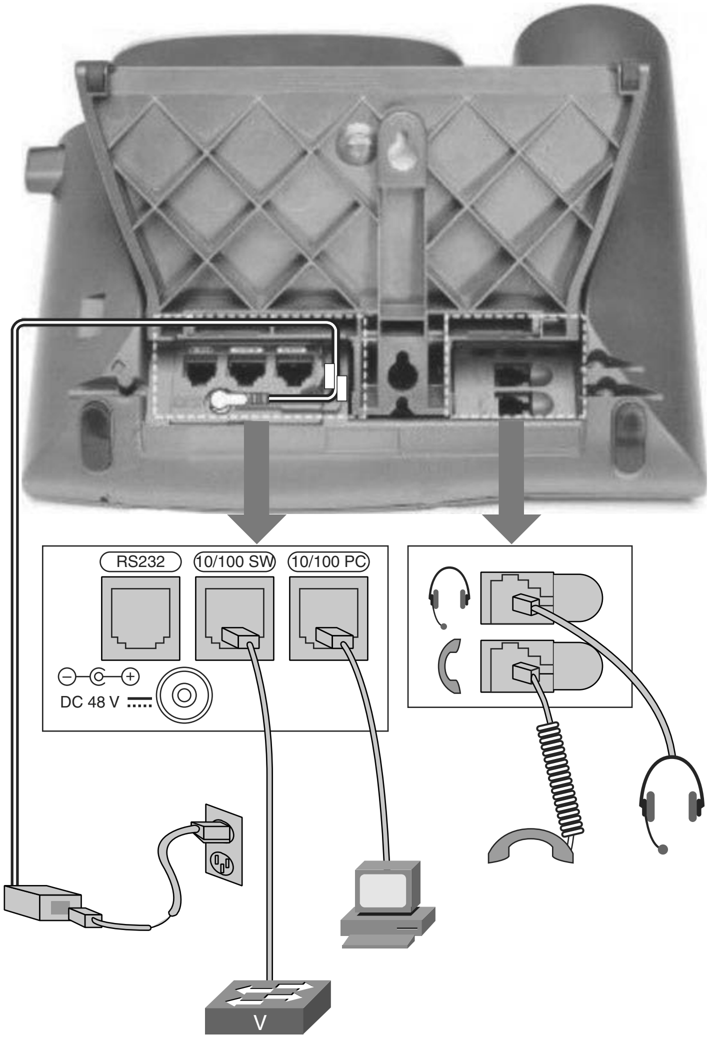


Figure 3-2 *Cisco IP Phone Ethernet Connections*

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

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

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

Cisco Catalyst Switch PoE

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

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

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

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

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

Powering the IP Phone Using a Power Patch Panel or Coupler

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

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

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

Inline PoE injectors provide a low-cost PoE solution for single devices (one device per coupler). These are typically used to support wireless access points or other "single spot" PoE solutions. Using inline PoE couplers for a large IP Phone network would make a mess of your wiring infrastructure and exhaust your supply of electrical outlets (because each inline PoE coupler requires a dedicated plug).

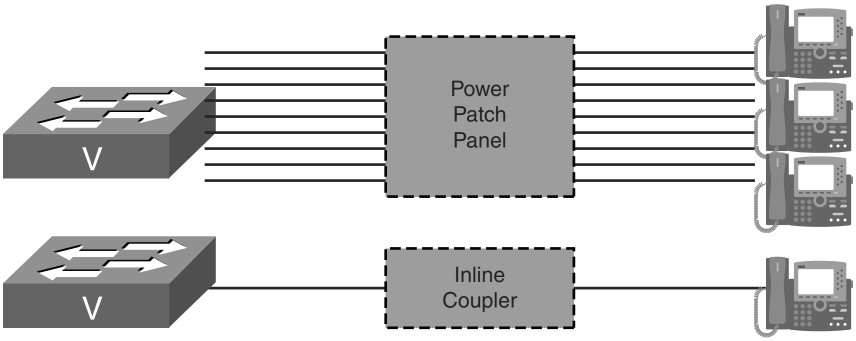


Figure 3-3 *Design for Power Patch Panels or Inline Couplers*

Powering the IP Phone with a Power Brick

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

Note: Some devices exceed the power capabilities of the 802.3af PoE standard. For example, when you add a 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.

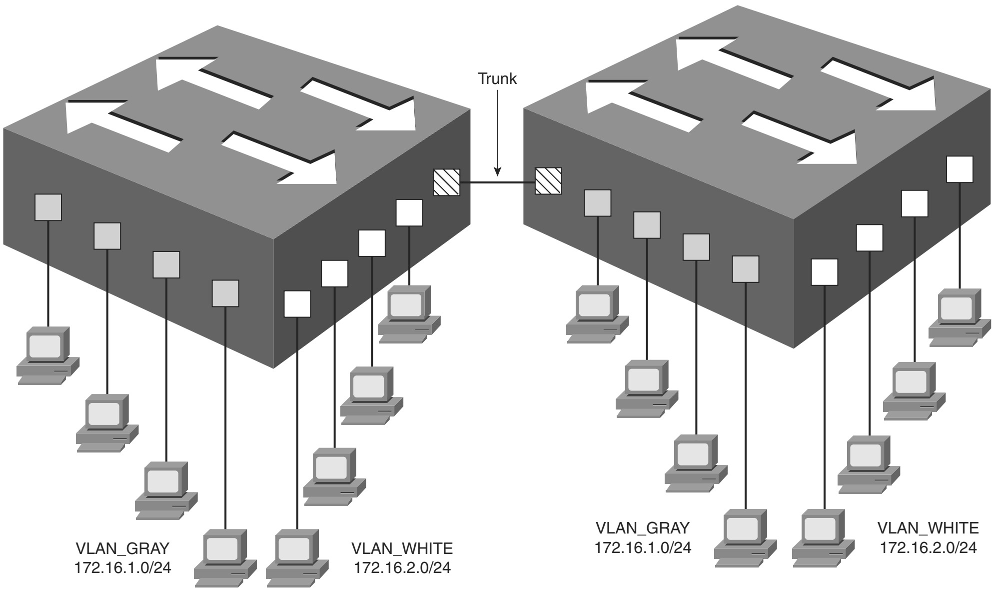
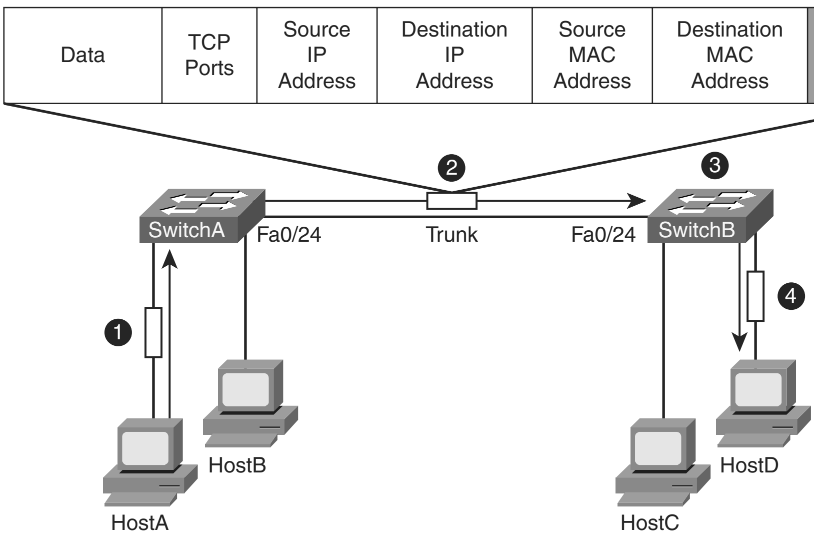


Figure 3-4 *VLANs Move Between Switches*



Key Topic

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.

Key Topic

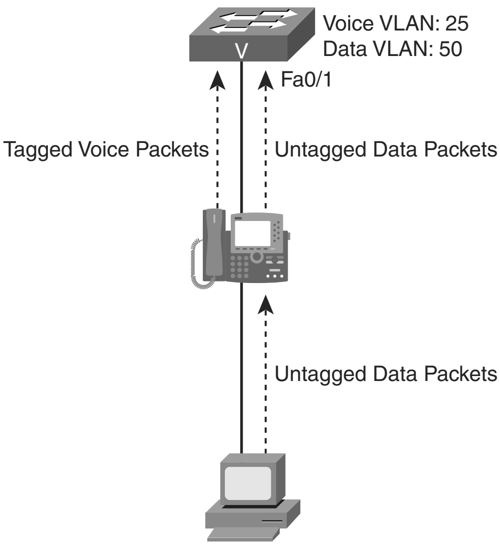


Figure 3-6 *Separating Voice and Data Traffic Using VLANs*

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

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

Key Topic

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*

Key Topic

```
Switch#configure terminal
Switch(config)#vlan 10
Switch(config-vlan)#name VOICE
Switch(config-vlan)#vlan 50
Switch(config-vlan)#name DATA
Switch(config-vlan)#end
Switch#show vlan brief
VLAN Name                             Status    Ports
---- -------------------------------- --------- -------------------------------
1    default                          active    Fa0/2, Fa0/3, Fa0/4, Fa0/5
                                                 Fa0/6, Fa0/7, Fa0/8, Fa0/9
                                                 Fa0/10, Fa0/11, Fa0/12, Fa0/13
                                                 Fa0/14, Fa0/15, Fa0/16, Fa0/17
                                                 Fa0/18, Fa0/19, Fa0/20, Fa0/21
                                                 Fa0/22, Fa0/23, Fa0/24, Gi0/1
                                                 Gi0/2
10   VOICE                            active
50   DATA                             active
1002 fddi-default                     act/unsup
1003 token-ring-default               act/unsup
1004 fddinet-default                  act/unsup
1005 trnet-default                    act/unsup
```

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

Key Topic

Example 3-2 *Assigning Voice and Data VLANs*

```
Switch#configure terminal
Switch(config)#interface range fa0/2 - 24
Switch(config-if-range)#switchport mode access
Switch(config-if-range)#spanning-tree portfast
Switch(config-if-range)#switchport access vlan 50
Switch(config-if-range)#switchport voice vlan 10
Switch(config-if-range)#end
Switch#show vlan brief
VLAN Name                             Status    Ports
—— ————————————————— ————- —————————————————-
1     default                         active    Gi0/1, Gi0/2
10    VOICE                           active    Fa0/2, Fa0/3, Fa0/4, Fa0/5
                                                 Fa0/6, Fa0/7, Fa0/8, Fa0/9
                                                 Fa0/10, Fa0/11, Fa0/12, Fa0/13
                                                 Fa0/14, Fa0/15, Fa0/16, Fa0/17
                                                 Fa0/18, Fa0/19, Fa0/20, Fa0/21
                                                 Fa0/22, Fa0/23, Fa0/24
50    DATA                            active    Fa0/2, Fa0/3, Fa0/4, Fa0/5
                                                 Fa0/6, Fa0/7, Fa0/8, Fa0/9
                                                 Fa0/10, Fa0/11, Fa0/12, Fa0/13
                                                 Fa0/14, Fa0/15, Fa0/16, Fa0/17
                                                 Fa0/18, Fa0/19, Fa0/20, Fa0/21
                                                 Fa0/22, Fa0/23, Fa0/24
1002 fddi-default                    act/unsup
1003 token-ring-default              act/unsup
1004 fddinet-default                 act/unsup
1005 trnet-default                   act/unsup
```

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

The ports are now configured to support a voice VLAN of 10 and a data VLAN of 50. This syntax is a newer form of configuration for IP Phone connections. In the "old days," you would configure the interface as a trunk port because the switch was establishing a trunking relationship between it and the IP phone. This was less secure because a hacker could remove the IP phone from the switchport and attach their own device (another managed switch or PC) and perform a VLAN-hopping attack. The more modern syntax configures the port as a "quasi-access port," because an attached PC will be able to access only VLAN 50. Only an attached Cisco IP Phone will be able to access the voice VLAN 10.

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

Understanding the Cisco IP Phone Boot Process

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

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

Configuring a Router-Based DHCP Server

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

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

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

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

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

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

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

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

Key Topic

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.

Key Topic

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

Regardless of the protocol used, the registration process is simple: The Cisco IP Phone contacts the call processing server and identifies itself by its MAC address. The call processing server looks at its database and sends the operating configuration to the phone. The operating configuration is different than the settings found in the configuration XML file located on the TFTP server. The TFTP server configuration is "base level settings," including items such as device language, firmware version, call processing server IP addresses, port numbers, and so on. The operating configuration contains items such as directory/line numbers, ring tones, softkey layout (on-screen buttons), and so on. Although the TFTP server configuration is sent using the TFTP protocol, the operating configuration is sent using SIP or SCCP.

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

Exam Preparation Tasks

Review All the Key Topics

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

Table 3-2 *Key Topics for Chapter 3*

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

Definitions of Key Terms

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

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

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Index

Numbers

A

B

C

D

E

F

G

H

I-J

K-L

M

N

O

P

Q

R

S

T

U

V-Z