Implementing Cisco Unified Communications Voice over IP and QoS (CVOICE)

Foundation Learning Guide

Foundation Learning for the CCNP® Voice (CVOICE) 642-437 Exam

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Kevin Wallace, CCIE® No. 7945

Implementing Cisco Unified Communications Voice over IP and QoS (CVOICE) Foundation Learning Guide

Fourth Edition

Kevin Wallace, CCIE No. 7945

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Kevin Wallace, CCIE No. 7945, is a certified Cisco instructor and holds multiple Cisco certifications, including the CCSP, CCVP, CCNP, and CCDP, in addition to multiple security and voice specializations. With Cisco experience dating back to 1989 (beginning with a Cisco AGS+ running Cisco IOS 7.x), Kevin has been a network design specialist for the Walt Disney World Resort, a senior technical instructor for SkillSoft/Thomson NETg/KnowledgeNet, and a network manager for Eastern Kentucky University. Kevin holds a bachelor's of science degree in electrical engineering from the University of Kentucky. Also, Kevin has authored multiple books for Cisco Press, including *CCNP TSHOOT 642-832 Official Certification Guide*, *Routing Video Mentor*, and the Video Mentor component of the *TSHOOT 642-832 Cert Kit*, all of which target the current CCNP certification. Kevin lives in central Kentucky with his wife, Vivian, and two daughters, Stacie and Sabrina. You can follow Kevin online through the following social media outlets:

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Dedications

As a young boy, my curiosity drove me to learn, experiment, and build things. Also, I promised myself at a young age that I would never forget what it was like to be a kid. My daughters (Stacie and Sabrina) and my wife (Vivian), who I embarrass on a regular basis, would tell you I've kept that promise.

But it was that hunger to learn more…to play…that led me on my journey of discovery in the networking world. So, I dedicate this book to the child in all of us. May we always be curious.

Acknowledgments

Thanks to all the great folks at Cisco Press, especially Brett Bartow, for their commitment to make this the best book it can be. You guys are totally professional and are a huge asset to Cisco learners everywhere.

My family deserves tremendous credit and acknowledgment for this book. It's a tough balancing act…to be a husband, a father, and an author. Family is definitely number one for me, and if I thought my hours of writing would hurt my family, then I would walk away from the keyboard. Fortunately, though, I am blessed with inexplicable support from my beautiful wife, Vivian, and two amazing daughters, Sabrina and Stacie. And speaking of being blessed, I thank God and His Son Jesus Christ for having a personal relationship with me. I fully realize that readers of this book come from a variety of faiths and traditions. So, I don't make such statements to be "preachy," I simply want you to know from where my strength comes.

Contents at a Glance

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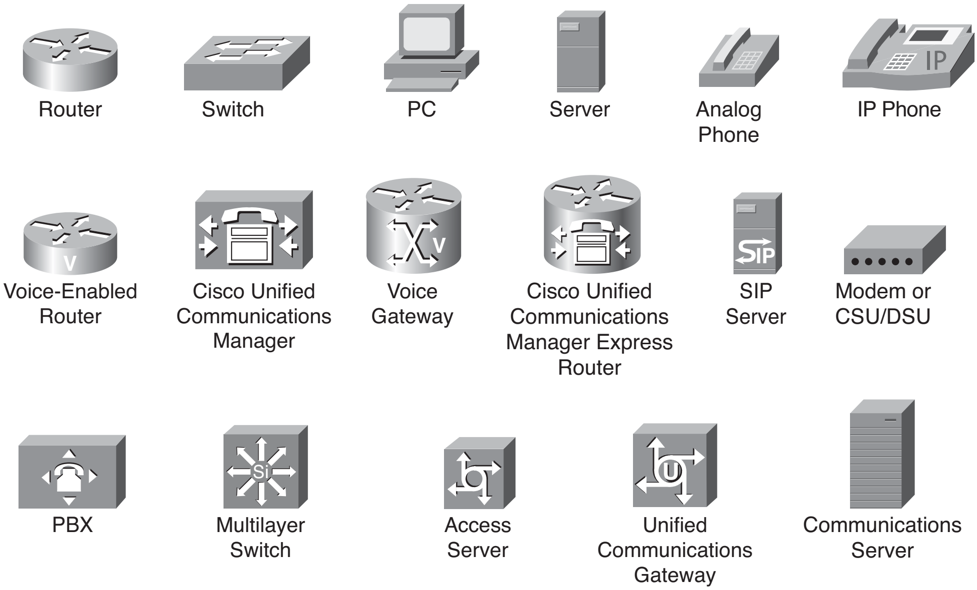


Command Syntax Conventions

The conventions used to present command syntax in this book are the same conventions used in the Cisco 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).
- *Italic* indicates arguments for which you supply actual values.
- Vertical bars (|) separate alternative, mutually exclusive elements.
- Square brackets ([ ]) indicate an optional element.
- Braces ({ }) indicate a required choice.
- Braces within brackets ([{ }]) indicate a required choice within an optional element.

Introduction

With the rapid adoption of Voice over IP (VoIP), many telephony and data network technicians, engineers, and designers are now working to become proficient in VoIP. Professional certifications, such as the CCNP Voice certification, offer validation of an employee's or a consultant's competency in specific technical areas.

This book mirrors the level of detail found in the Cisco CVOICE Version 8.0 course, which many CCNP Voice candidates select as their first course in the CCNP Voice track. Version 8.0 represents a significant update over the previous version, Version 6.0, of the CVOICE course. Specifically, Version 8.0 integrates much of the content previously found in the Implementing Cisco IOS Unified Communications (IIUC) 1.0 and Implementing Cisco QoS (QOS) 2.3 courses. This content includes coverage of Cisco Unified Communications Manager Express (CUCME) and quality of service topics.

A fundamental understanding of traditional telephony, however, would certainly benefit a CVOICE student or a reader of this book. If you think you lack a fundamental understanding of traditional telephony, a recommended companion for this book is the Cisco Press book *Voice over IP First-Step* (ISBN: 978-1-58720-156-1), which is also written by this book's author. *Voice over IP First-Step* is written in a conversational tone and teaches concepts surrounding traditional telephony and how those concepts translate into a VoIP environment.

Additional Study Resources

This book contains a CD with 14 supplemental video lab demonstrations. The video lab titles are as follows:

- Lab 1: DHCP Server Configuration
- Lab 2: CUCME Auto Registration Configuration
- Lab 3: ISDN PRI Configuration for an E1 Circuit
- Lab 4: Configuring a PSTN Dial Plan
- Lab 5: Configuring DID with Basic Digit Manipulation
- Lab 6: H.323 Gateway and VoIP Dial Peer Configuration
- Lab 7: Dial Peer Codec Selection
- Lab 8: Voice Translation Rules and Voice Translation Profiles
- Lab 9: MGCP Gateway Configuration
- Lab 10: Configuring PSTN Failover
- Lab 11: Class of Restriction (COR) Configuration
- Lab 12: Configuring a Gatekeeper
- Lab 13: Configuring a Gateway to Register with a Gatekeeper
- Lab 14: Configuring AutoQoS VoIP

In addition to the 14 video labs, this book periodically identifies bonus videos (a total of 8 bonus videos), which can be viewed on the author's web site (1ExamAMonth.com). These bonus videos review basic telephony theory (not addressed in the course). This telephony review discusses analog and digital port theory and configuration. Other fundamental concepts (that is, dial-peer configuration and digit manipulation) are also addressed. Finally, these bonus videos cover three of the most challenging QoS concepts encountered by students.

With the combination of the 14 video labs on the accompanying CD and the 8 bonus online videos, you have 22 videos to help clarify and expand on the concepts presented in the book.

Goals and Methods

The primary objective of this book is to help the reader pass the 642-437 CVOICE exam, which is a required exam for the CCNP Voice certification.

One key methodology used in this book is 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. This book does not try to help you pass by memorization, but helps you truly learn and understand the topics 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, including detailed illustrations and topologies as well as sample configurations
- Providing exam practice questions to confirm your understanding of core concepts

Who Should Read This Book?

This book is primarily targeted toward candidates of the CVOICE exam. However, because CVOICE is one of the Cisco foundational VoIP courses, this book also serves as a VoIP primer to noncertification readers.

Many Cisco resellers actively encourage their employees to attain Cisco certifications, and seek new employees who already possess Cisco certifications, to obtain deeper discounts when purchasing Cisco products. Additionally, having attained a certification communicates to your employer or customer that you are serious about your craft and have not simply "hung out a shingle" declaring yourself knowledgeable about VoIP. Rather, you have proven your competency through a rigorous series of exams.

How This Book Is Organized

Although the chapters in this book could be read sequentially, the organization allows you to focus your reading on specific topics of interest. For example, if you already possess a strong VoIP background but want to learn more about Cisco Unified

Communications Manager Express, you can jump right to Chapter 3. Alternately, if you are interested in quality of service (QoS), and not necessarily for VoIP purposes, you can read about basic QoS theory in Chapter 7 and see how to configure various QoS mechanisms in Chapter 8. Specifically, the chapters in this book cover the following topics:

- **Chapter 1, "Introducing Voice Gateways":** This chapter describes the characteristics and historical evolution of unified communications networks, the three operational modes of gateways, their functions, and the related call leg types. Also, this chapter explains how gateways route calls and which configuration elements relate to incoming and outgoing call legs. Additionally, Chapter 1 describes how to connect a gateway to traditional voice circuits using analog and digital interfaces. Finally, DSPs and codecs are addressed.
- **Chapter 2, "Configuring Basic Voice over IP":** This chapter describes how VoIP signaling and media transmission differs from traditional voice circuits, and explains how voice is sent over IP networks, including analog-to-digital conversion, encoding, and packetization. Characteristics of the gateway protocols H.323, SIP, and MGCP are presented, along with special considerations for transmitting DTMF, fax, and modem tones. Finally, this chapter introduces the concept of dial peers.
- **Chapter 3, "Supporting Cisco IP Phones with Cisco Unified Communications Manager Express":** This chapter focuses on Cisco Unified Communications Manager Express (CUCME). After a discussion of CUCME theory and components, this chapter covers CUCME configuration.
- **Chapter 4, "Introducing Dial Plans":** This chapter describes the characteristics and requirements of a numbering plan. Also, the components of a dial plan, and their functions, are explained.
- **Chapter 5, "Implementing Dial Plans":** This chapter describes how to configure a gateway for digit manipulation, how to configure a gateway to perform path selection, and how to configure calling privileges on a voice gateway.
- **Chapter 6, "Using Gatekeepers and Cisco Unified Border Elements":** This chapter describes Cisco gatekeeper functionality, along with configuration instructions. Additionally, this chapter addresses how a gatekeeper can be used to perform call admission control (CAC). Also covered in Chapter 6 is Cisco Unified Border Element (UBE) theory and configuration.
- **Chapter 7, "Introducing Quality of Service":** This chapter explains the functions, goals, and implementation models of QoS, and what specific issues and requirements exist in a converged Cisco Unified Communications network. Also addressed in this chapter are the characteristics and QoS mechanisms of the DiffServ QoS model, as contrasted with other QoS models.
- **Chapter 8, "Configuring QoS Mechanisms":** This chapter explains the operation and configuration of various QoS mechanisms, including classification, marking, queuing, congestion avoidance, policing, shaping, Link Fragmentation and Interleaving (LFI), and header compression. Additionally, all variants of Cisco AutoQoS are described, along with configuration guidance.

Appendix A, "Answers Appendix," lists the answers to the end-of-chapter review questions.

Chapter 4

Introducing Dial Plans

After reading this chapter, you should be able to perform the following tasks:

- Describe the characteristics and requirements of a numbering plan.
- Explain the components of a dial plan and their functions.

Dial plans are essential for any Cisco Unified Communications deployment. Whether you are implementing single-site or multisite deployments, having a thorough understanding of dial plans and the knowledge of how to implement them on Cisco IOS gateways is essential for any engineer who designs and implements a Cisco Unified Communications network. This chapter describes the characteristics of a dial plan and associated components (for example, a numbering plan).

Numbering Plan Fundamentals

To integrate VoIP networks into existing voice networks, you should have the skills and knowledge to implement call routing and design an appropriate numbering plan. A scalable numbering plan establishes the baseline for a comprehensive, scalable, and logical dial plan.

This section describes call-routing principles, discusses attributes of numbering plans for voice networks, addresses the challenges of designing these plans, and identifies methods of implementing numbering plans.

Introducing Numbering Plans

A *numbering plan* is a numbering scheme used in telecommunications to allocate telephone number ranges to countries, regions, areas, and exchanges, and to nonfixed telephone networks such as mobile phone networks. A numbering plan defines rules for assigning numbers to a device.

Types of numbering plans include the following:

- **Private numbering plans:** Private numbering plans are used to address endpoints and applications within private networks. Private numbering plans are not required to adhere to any specific format and can be created to accommodate the needs of a network. Because most private telephone networks connect to the PSTN at some point in a design, it is good practice to plan a private numbering plan to coincide with publicly assigned number ranges. Number translation might be required when connecting private voice networks to the PSTN.
- **Public or PSTN numbering plans:** PSTN or public numbering plans are unique to the country in which they are implemented. The most common PSTN numbering plans are explained in this section.

Different regions of the globe have different numbering plans. However, all of these national numbering plans must adhere to the international E.164 standard. As an example, the E.164 standard stipulates than no phone number can be longer than 15 digits.

North American Numbering Plan

The North American Numbering Plan (NANP) is an integrated telephone numbering plan that serves 19 North American countries that share its resources. Developed in 1947 and first implemented in 1951 by AT&T, the NANP simplifies and facilitates the direct dialing of long-distance calls. The countries that use the NANP include the United States and its territories, Canada, Bermuda, Anguilla, Antigua and Barbuda, the Bahamas, Barbados, the British Virgin Islands, the Cayman Islands, Dominica, the Dominican Republic, Grenada, Jamaica, Montserrat, St. Kitts and Nevis, St. Lucia, St. Vincent and the Grenadines, Trinidad and Tobago, and Turks and Caicos Islands.

NANP numbers are ten-digit numbers, usually formatted as NXX-NXX-XXXX, in which N is any digit from 2 through 9 and X is any digit from 0 through 9. This structure is depicted in Figure 4-1.

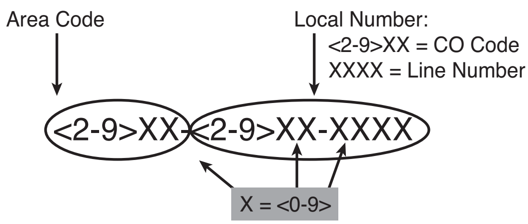


Figure 4-1 *North American Numbering Plan*

The first three digits of an NANP number (NXX) are called the Numbering Plan Area (NPA) code, often called the area code. The second three digits (NXX) are called the central office (CO) code, switched code, or prefix. The final four digits (XXXX) are called the line number or station number. The North American Numbering Plan Administration (NANPA) administers the NANP.

NANP Numbering Assignments

An area served by the NANP is divided into smaller areas, each identified by a three-digit NPA code, or area code. There are 800 possible combinations of area codes with the NXX format. However, some of these combinations are not available or have been reserved for special purposes, as shown in Table 4-1.

Table 4-1 *NANP Numbering Codes*

Reserved Code	Purpose
Easily Recognizable Codes (ERC)	When the second and third digits of an area code are the same, that code is called an ERC. These codes designate special use, such as toll-free service (for example, 800, 866, 877, or 888).
Automatic Number Identification (ANI) II digits	ANI II digits are two-digit pairs sent with an originating telephone number as part of the signaling that takes place during the setup phase of a call. These digits identify the type of originating station.
Carrier Identification Codes (CIC)	CICs are used to route and bill calls in the PSTN. CICs are four-digit codes in the format XXXX, where X is any digit from 0 through 9. There are separate CIC pools for different feature groups, such as line-side and trunk-side access.
International dialing	You dial 011 before the country code and the specific destination number to signal that you are placing an international call.
Long distance	The first 1 dialed defines a toll call within the NANP.
In-state long-distance or local call	A ten-digit number might be either a toll call within a common region or, in many larger markets, a local call if the area code is the same as the source.
Seven-digit number (<2–9>XX-XXXX)	A seven-digit number defines a local call. Some larger areas use ten-digit numbers instead of seven-digit numbers to define local calls. Notice that the first digit is in the range 2 through 9, while the remaining digits (as represented by X) can be any number in the range of 0 through 9.

Eight N11 codes, called *service codes*, are not used as area codes. These are three-digit codes in the N11 format, as shown in Table 4-2.

Table 4-2 *N11 Code Assignments*

N11 Code	Purpose
211	Community information and referral services (United States)
311	Nonemergency police and other governmental services (United States)
411	Local directory assistance
511	Traffic and transportation information (United States); reserved (Canada)
611	Repair service
711	Telecommunications relay services (TRS)
811	Business office
911	Emergency

In some U.S. states, N11 codes that are not assigned nationally can be assigned locally, if the local assignments can be withdrawn promptly if a national assignment is made. There are no industry guidelines for the assignment of N11 codes.

Additional NANP reserved area codes include the following:

- **456-<2–9>XX-XXXX numbers:** These codes identify carrier-specific services by providing carrier identification within the dialed digits. The prefix following 456, <2–9>XX, identifies the carrier. Use of these numbers enables the proper routing of inbound international calls, destined for these services into, and between, NANP area countries.
- **555-01XX line numbers:** These numbers are fictitious telephone numbers that can be used, for example, in the film industry, for educational purposes, and for various types of demonstrations. If anyone dials one of these numbers, it does not cause a nuisance to any actual person.
- **800-XXXX through 855-XXXX line numbers:** These numbers are in the format 800-855- XXXX and provide access to PSTN services for deaf, hard-of-hearing, or speech-impaired persons. Such services include Telecommunications Relay Service (TRS) and message relay service.
- **900-<2–9>XX-XXXX numbers:** These codes are for premium services, with the cost of each 900 call billed to the calling party. 900-<2–9>XX codes, each subsuming a block of 10,000 numbers, are assigned to service providers who provide and typically bill for premium services. These service providers, in turn, assign individual numbers to their customers.

European Telephony Numbering Space

The European Telephony Numbering Space (ETNS) is a European numbering space that is parallel to the existing national numbering spaces and is used to provision pan-European services. A pan-European service is an international service that can be invoked from at least two European countries.

The European Telecommunications Office (ETO) Administrative Council supervises the telecommunications work of the European Radiocommunications Office (ERO). This supervision includes the establishment, detailing, and change of ETNS conventions and the designation of European Service Identification (ESI) for new ETNS services.

The main objective of ETNS is to allow effective numbering for European international services for which national numbers might not be adequate and global numbers might not be available. The designation of a new European country code, 388, allows European international companies, services, and individuals to obtain a single European number for accessing their services.

Four ETNS services are now available: Public Service Application, Customer Service Application, Corporate Networks, and Personal Numbering. An ESI code is designated for each ETNS service. The one-digit code follows the European Country Code 388 and European Service Code 3 (3883), as shown in Table 4-3.

Figure 4-2 shows the structure of a standard international number. The initial part that is known as the ESI consists of the country code and group identification code that identifies the ETNS (3883), followed by a European Service Code that identifies a particular ETNS service. The European Subscriber Number is the number that is assigned to a customer in the context of the specific service. The maximum length of a European Subscriber Number is 15 digits; for example, 3883 X XXXXXXXXXX.

Table 4-3 *ETNS Service and ESI Codes*

ETNS Service	ESI
Public Service Application	3883 1
Customer Service Application	3883 3
Corporate Networks	3883 5
Personal Numbering	3883 7

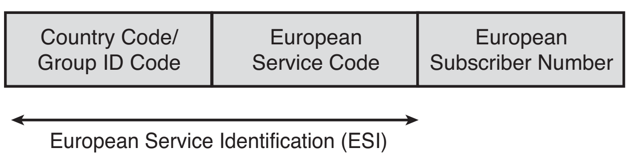


Figure 4-2 *European Numbering Structure*

Fixed and Variable-Length Numbering Plan Comparison

A fixed numbering plan, such as found in North America, features fixed-length area codes and local numbers. An open numbering plan, as found in countries that have not yet standardized on numbering plans, features variance in the length of the area code or the local number, or both.

A numbering plan can specify parameters such as the following:

- **Country code:** A country code is used to reach the particular telephone system for each country or special service.
- **Area code:** An area code is typically used to route calls to a particular city, region, or special service. Depending on the region, it might also be referred to as a Numbering Plan Area, subscriber trunk dialing code, national destination code, or routing code.
- **Subscriber number:** A subscriber number represents the specific telephone number to be dialed, but does not include the country code, area code (if applicable), international prefix, or trunk prefix.
- **Trunk prefix:** A trunk prefix refers to the initial digits to be dialed in a domestic call, prior to the area code and the subscriber number.
- **International prefix:** An international prefix is the code dialed prior to an international number (the country code, the area code if any, and then the subscriber number).

Table 4-4 contrasts the NANP and a variable-length numbering plan (Germany's numbering plan in this example).

Table 4-4 *Fixed and Variable-Length Numbering Plan Comparison*

Components	Fixed Numbering Plan	Variable-Length Numbering Plan
Example	NANP	Germany
Country code	1	49
Area code	Three digits	Two to four digits
Subscriber number	Three-digit exchange code + four-digit station code	Five to eight digits
Access code	9 (commonly used but not required)	0
International prefix	011	00 or +

E.164 Addressing

E.164, as illustrated in Figure 4-3, is an international numbering plan for public telephone systems in which each assigned number contains a one-, two-, or three-digit country code (CC) that is followed by a national destination code (NDC) and then by a subscriber number (SN). An E.164 number can have as many as 15 digits. The ITU originally developed the E.164 plan.

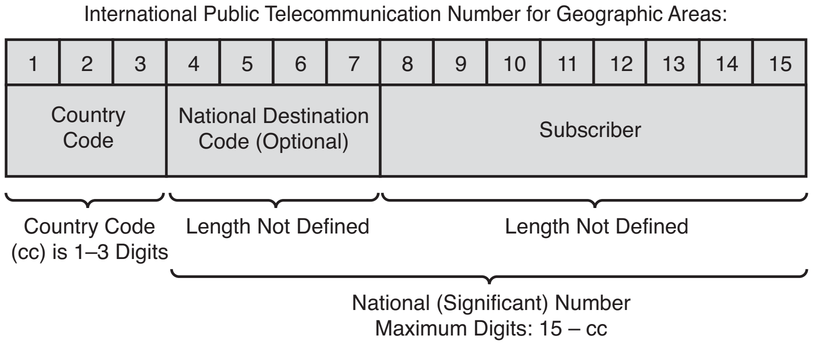


Figure 4-3 *E.164 Address Structure*

In the E.164 plan, each address is unique worldwide. With as many as 15 digits possible in a number, there are 100 trillion possible E.164 phone numbers. This makes it possible, in theory, to direct-dial from any conventional phone set to any other conventional phone set in the world by dialing no more than 15 single digits.

Most telephone numbers belong to the E.164 numbering plan, although this does not include internal private automatic branch exchange (PABX) extensions.

The E.164 numbering plan for telephone numbers includes the following plans:

- Country calling codes
- Regional numbering plans, such as the following:
  - ETNS
  - NANP
- Various national numbering plans, such as the U.K. National Numbering Scheme

Scalable Numbering Plans

Scalable telephony networks require well-designed, hierarchical telephone numbering plans. A hierarchical design has these five advantages:

- **Simplified provisioning:** Ability to easily add new numbers and modify existing numbers
- **Simplified routing:** Keeps local calls local and uses a specialized number key, such as an area code, for long-distance calls
- **Summarization:** Allows the grouping of numbers in number ranges
- **Scalability:** Leaves space for future growth
- **Management:** Control from a single management point

When designing a numbering plan, consider these four attributes to allow smooth implementation:

- Minimal impact on existing systems
- Minimal impact on users of the system
- Minimal translation configuration
- Consideration of anticipated growth

Although a non-overlapping numbering plan is usually preferable to an overlapping numbering plan, both plans can be configured to be scalable.

Non-Overlapping Numbering Plan

A dial plan can be designed so that all extensions within the system are reached in a uniform way. That is, a fixed quantity of digits is used to reach a given extension from any on-net origination point. Uniform dialing is desirable because of its simplicity. A user does not have to remember different ways to dial a number when calling from various on-net locations.

Figure 4-4 shows an example of a four-digit uniform dial plan, described here:

- The 0xxx and 9xxx number ranges are excluded due to off-net access code use and operator dialing. In such a system, where 9 and 0 are reserved codes, no other extensions can start with 0 or 9.
- Site A has been assigned the range 1xxx, allowing for as many as 1000 extensions.

- Site B has been assigned the range 2xxx, allowing for as many as 1000 extensions.
- Sites C and D were each assigned 500 numbers from the 4xxx range.
- The ranges 6xxx, 7xxx, and 8xxx are reserved for future use.

After a given quantity of digits has been selected, and the requisite ranges have been excluded (for example, ranges beginning with 9 or 0), the remaining dialing space has to be divided between all sites. Most systems require that two ranges be excluded, thus leaving eight different possibilities for the leading digit of the dial range. The table in Figure 4-4 is an example of the distribution of dialing space for a typical four-digit uniform dial plan.

Location	Range	Description
	0xxx, 9xxx	Reserved
Site A	1xxx	Site A Extensions
Site B	2xxx	Site B Extensions
Site C	4[0–4]xx	Site C Extensions
Site D	4[5–9]xx	Site D Extensions
	[6–8]xxx	Available for Future Needs

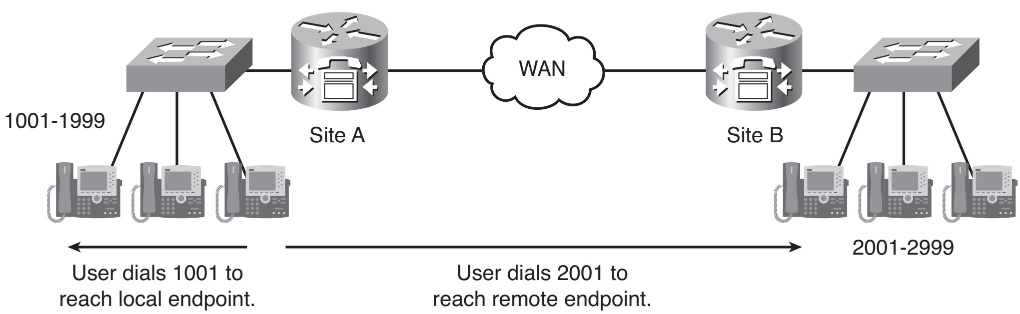


Figure 4-4 *Non-Overlapping Numbering Plan*

Scalable Non-Overlapping Numbering Plan Considerations

In a non-overlapping numbering plan, all extensions can be addressed using the same number of digits, making the call routing simple and making the network easily manageable. The same number length is used to route the call to an internal user and a remote user.

The disadvantage of the non-overlapping numbering plan is that it is often impractical in real life. It requires a centralized numbering approach and a careful design from the very beginning.

Overlapping Numbering Plans

In Figure 4-5, Site A endpoints use directory numbers 1001 through 1099, 3000 through 3157, and 3365 through 3985. At Site B, 1001 through 1099 and 3158 through 3364 are implemented. Site C uses ranges 1001 through 1099 and 3986 through 3999. There are two issues with these directory numbers: 1001 through 1099 overlap. These directory numbers exist at all three sites, so they are not unique throughout the complete deployment. In addition, the poor structure of splitting the range 3000 through 3999 requires many entries in call-routing tables, because the ranges cannot be summarized by one or a few entries.

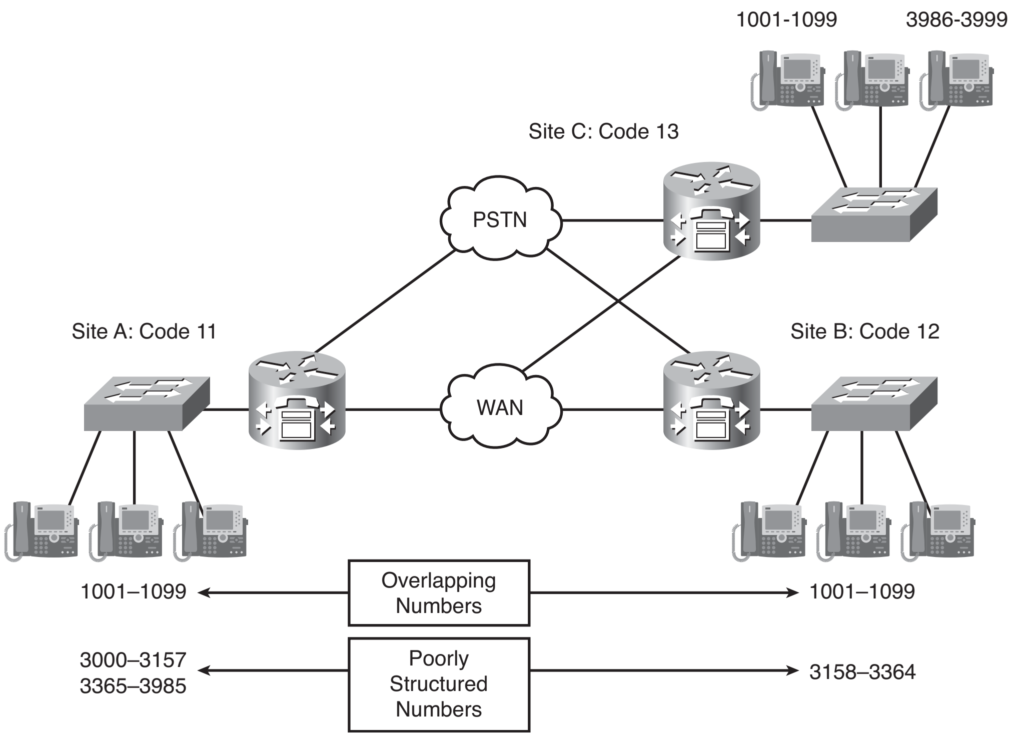


Figure 4-5 *Overlapping and Poorly Structured Numbering Plan*

A sampling of ways to solve overlapping and poorly structured directory number problems includes the following:

- Redesign the directory number ranges to ensure non-overlapping, well-structured directory numbers.
- Use an intersite access code and a site code that will be prepended to a directory number to create unique dialable numbers. For example, you could use an intersite code of 8, assigning Site A the site code 81, Site B the site code 82, and Site C the site code 83.
- Do not assign direct inward dialing (DID) numbers. Instead, publish a single number, and use a receptionist or auto-attendant.

Overlapping Numbering Plan Example

Figure 4-6 illustrates the most common solution to the overlap problem in numbering plans.

Location	Range	Site Code	Intersite Prefix
Site A	1xxx	11	8
Site B	1xxx	12	8
Site C	1xxx	13	8
Site D	2xxx	14	8

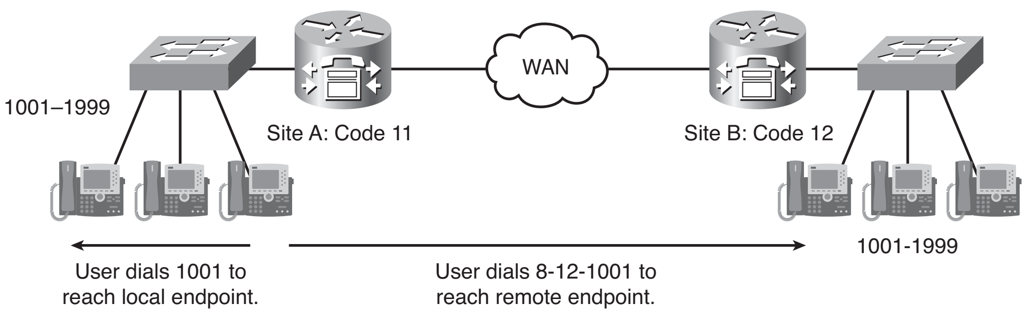


Figure 4-6 *Overlapping Numbering Plan Example*

The principle of site-code dialing introduces an intersite prefix (8, in this example) and a site code (1x, in this example) that must be prepended when dialing an internal extension in another site. With this solution, a Site A user dials a four-digit number starting with 1 to reach a local extension, and enters a seven-digit number starting with 8 to reach an endpoint in a remote site. The intersite prefix and the site code that are used in this scenario show sample values and can be set differently according to enterprise requirements. For example, the intersite prefix is commonly set to 8 and the access code to 9 in an NANP region, while the intersite prefix is typically 9 and the access code 0 in Europe.

Scalable Overlapping Numbering Plan Considerations

The site-code dialing solution of the overlap issue in numbering plans is useful in real life, as it allows a decentralized approach to the numbering effort. Even various departments within an organization can manage their own addressing space, and the site codes can interconnect them into a manageable unified communications network. Site code dialing does not require a careful design from the beginning and can be implemented as the enterprise grows.

Internal extensions should not start with the intersite prefix (for example, 8), because such entries could cause ambiguity in the dial plan. The intersite prefix notifies the call-routing device that the call is destined for a remote location and therefore should not overlap with any internal number.

Private and Public Numbering Plan Integration

Figure 4-7 illustrates an enterprise with four locations in the NANP region.

Location	Range	Site Code	Intersite Prefix	PSTN DID Range	Access Code
Site A	1xxx	11	8	200-555-1xxx	9
Site B	1xxx	12	8	300-555-3xxx	9
Site C	1xxx	13	8	400-555-1234	9
Site D	2xxx	14	8	500-555-22xx	9

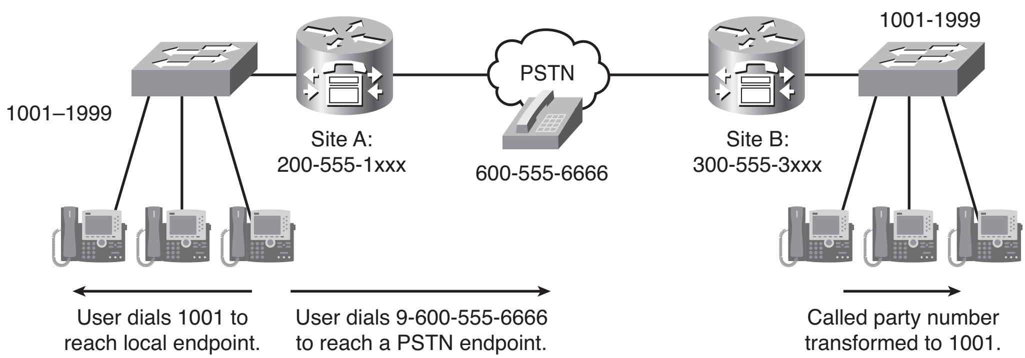


Figure 4-7 *Private and Public Numbering Plan*

Site-code dialing has been designed to allow calls between the enterprise locations. Each site has a trunk connection to the PSTN, with the PSTN DID range provided by the telephone company (telco) operator. Sites A and B have DID ranges that allow public addressing of each internal extension. Site C has a single DID number with an interactive voice response (IVR) solution that prompts the callers for the number of the internal extension for forwarding inbound calls to the intended callee. The DID range of Site D covers some internal extensions and must be combined with an IVR to provide inbound connectivity to others.

Access code 9 identifies a call that is destined to an external PSTN recipient. In this example, internal users dial 9-600-555-6666 to reach the PSTN endpoint.

The following are a few challenges that you might face with numbering plan integration:

- **Varying number lengths:** Within the IP network, consideration is given to varying number lengths that exist outside the IP network. Local, long-distance, and international dialing from within the IP network might require digit manipulation.
- **Necessity of prefixes or area codes:** It can be necessary to strip or add area codes, or prepend or replace prefixes. Rerouting calls from the IP network to the PSTN for failure recovery can require extra digits.

Private and Public Numbering Plan Integration Functions

The three basic features, as illustrated in Figure 4-8, that are provided by the integrated private and public numbering plans include the following.

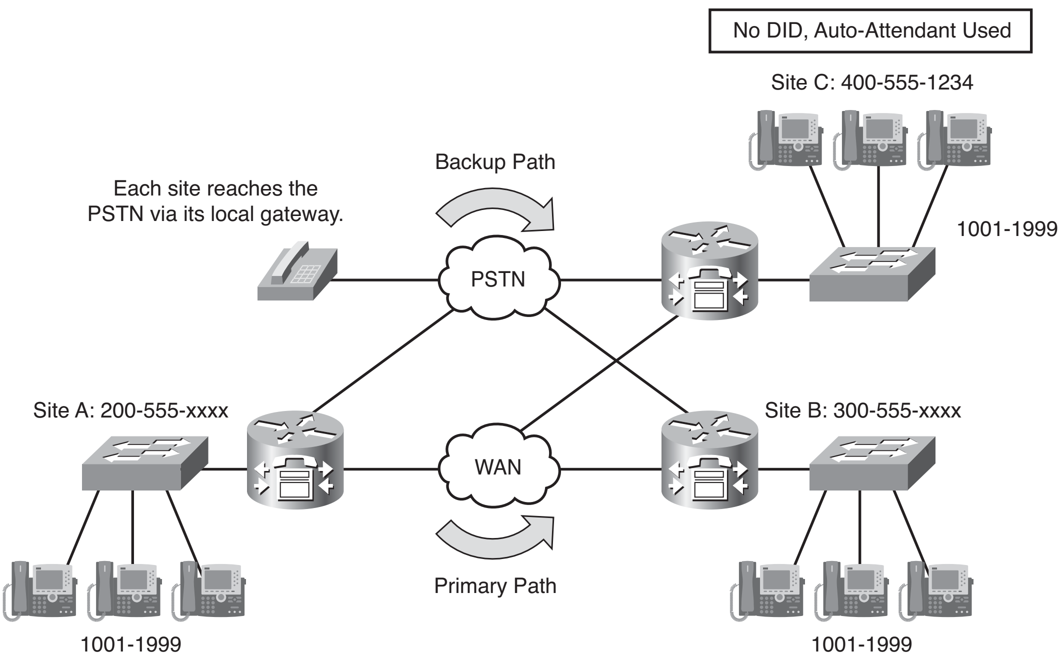


Figure 4-8 *Private and Public Numbering Plan Integration Functions*

- **Reachability to external PSTN destinations:** Internal users get access to external destinations over a gateway, which acts like a junction between the private and public addressing scheme.
- **Auto-attendant:** An IVR system is required to provide connectivity to internal extensions when a sufficient DID range is not available.
- **PSTN acts a backup path in case the IP WAN fails or becomes congested:** In such cases, the gateways redirect the intersite calls over the PSTN to provide uninterrupted service.

Private and Public Numbering Plan Integration Considerations

When integrating private and public numbering plans, give special consideration to these aspects:

- **No ambiguity with the internal and intersite dialing:** The prepended access code should uniquely identify all calls that should break out to the PSTN.
- **Path selection transparent to the user:** Users dial site codes to reach remote locations, and the intersite calls select the IP network as the primary path. If the IP WAN is unavailable, the call should be redirected over the PSTN. The user does not need to take any action for the secondary path to be chosen.
- **Auto-attendant for non-DID numbers:** When the DID range does not cover all internal extensions, an auto-attendant is needed to allow inbound calls.
- **Number adjustment:** The voice gateway needs to adjust the calling and called numbers when a call is set up between the sites or via the PSTN. One manipulation requirement arises when an intersite call is rerouted over the PSTN. The intersite prefix and site code (for example, 8-12) must then be replaced with a public number identifying the location (for example, 300-555). Another type of manipulation is needed to map the internal ranges to DID scopes, for example, 1xxx through 0-555-3xxx.

Number Plan Implementation Overview

The implementation of the private numbering plan takes into account the number of users per site and the number of sites. The length of the internal numbers and the site codes must match the size of the environment and at the same time allow space for future growth. Figure 4-9 illustrates that the internal extensions can consist of two, three, or four digits, and the site codes can consist of one, two, or three digits. Note that extension length should be consistent for each site to avoid interdigit timeout or reachability issues.

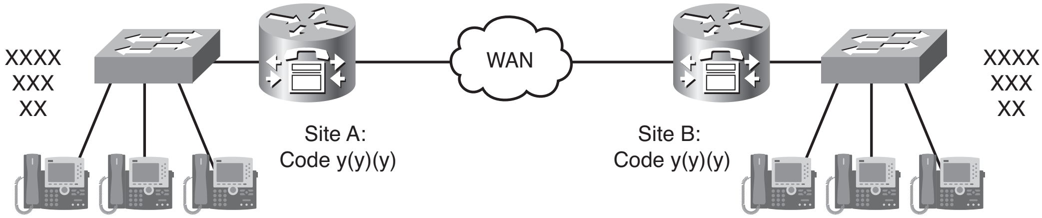


Figure 4-9 *Private Number Plan Implementation*

Call routing to local endpoints is achieved automatically, because the registering endpoints have virtual dial peers that are associated with them. The dial peers ensure that calls are routed to the registered phones based on their directory numbers.

Call routing to remote locations is enabled by VoIP dial peers that describe the primary path over an IP WAN.

Private Number Plan Implementation Example

Figure 4-10 shows the enterprise has one large site (Site A) with 7000 users and several smaller sites with less than 700 users each. The codes for all sites are two-digit numbers (21 through 40). The internal extensions in the large site are four digits long (1001–7999), while the extensions in the smaller sites are three digits long (101–799). To implement the dial plan, VoIP dial peers are configured with destination patterns that match seven-digit numbers in the large site and six-digit numbers in the remaining sites, starting with the intersite prefix 8.

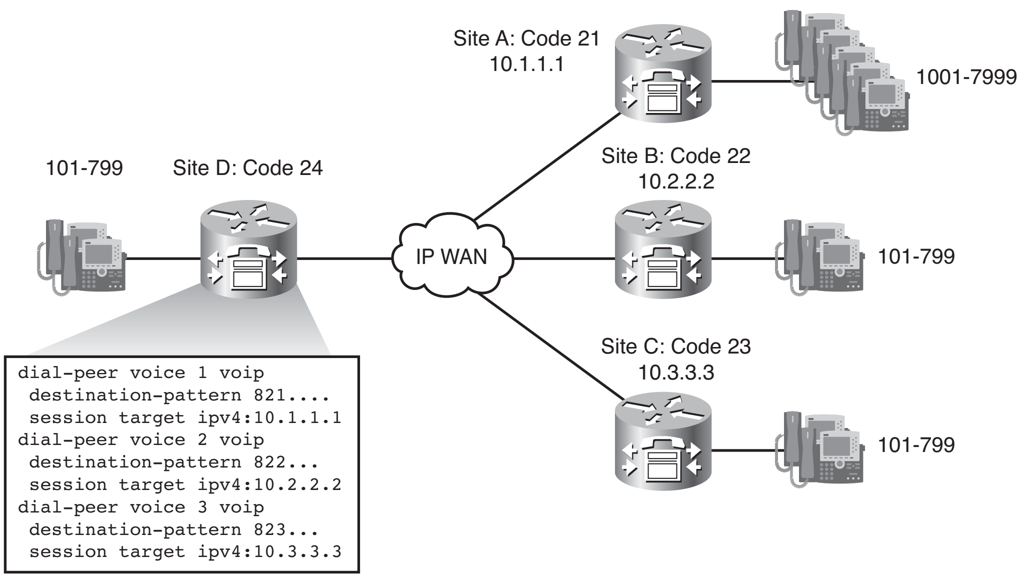


Figure 4-10 *Private Number Plan Implementation Example*

Public Number Plan Implementation

The enterprise does not design its public numbering plan. It is imposed by the telco operator. The enterprise might influence the size of the DID range, which is often related to a financial decision.

Gateways provide a mapping between the DID and the internal number ranges. For example, the PSTN DID range 200-555-3xxx can be easily converted to 1xxx and back when calls traverse the gateway. Complex mapping formulas (for example, mapping of 200-555-3xxx to 1xxx + 50) are too complex to implement and should be avoided.

Call Routing Overview

The most relevant properties of call routing can be compared to the characteristics of IP packet routing, as shown in Table 4-5.

Table 4-5 *Call Routing Refresher*

IP Routing	Call Routing
Static or dynamic	Only static.
IP routing table	Dial plan.
IP route	Dial peer.
Hop-by-hop routing, where each router makes an independent decision	Inbound and outbound call legs. The gateway negotiates VoIP parameters with preceding and next gateways before a call is forwarded.
Destination-based routing	Called number, matched by **destination-pattern**, is one of many selection criteria.
Most explicit match rule	The most explicit match rule for **destination-pattern** exists, but other criteria are also considered.
Equal paths	Preference can be applied to equal dial peers. If all criteria are the same, a random selection is made.
Default route	Possible. Often points at external gateway or gatekeeper.

The entries that define where to forward calls are the dial peers. All dial peers together build the dial plan, which is equivalent to the IP routing table. The dial peers are static in nature.

Hop-by-hop call routing builds on the principle of call legs. Before a call-routing decision is made, the gateway must identify the inbound dial peer and process its parameters. This process might involve VoIP parameter negotiation.

A call-routing decision is the selection of the outbound dial peer. This selection is commonly based on the called number, when the **destination-pattern** command is used. The selection might be based on other information, and that other criteria might have higher precedence than the called number. When the called number is matched to find the outbound dial peer, the longest match rule applies.

If more than one dial peer equally matches the dial string, all of the matching dial peers are used to form a so-called rotary group. The router attempts to place the outbound call leg using all of the dial peers in the rotary group until one is successful. The selection order within the group can be influenced by configuring a preference value.

A default call route can be configured using special characters when matching the number.

Call Routing Example

The voice gateways in this example are faced with the task of selecting the best path for a given destination number. Such a requirement arises when the preferred path goes through an IP WAN. A backup PSTN path should be chosen when an IP WAN is either unavailable or lacks the needed bandwidth resources.

Figure 4-11 illustrates a scenario with two locations that are connected to an IP WAN and PSTN. When the call goes through the PSTN, its numbers (both calling and called) have to be manipulated so that they are reachable within the PSTN. Otherwise, the PSTN switches will not recognize the called number, and the call will fail.

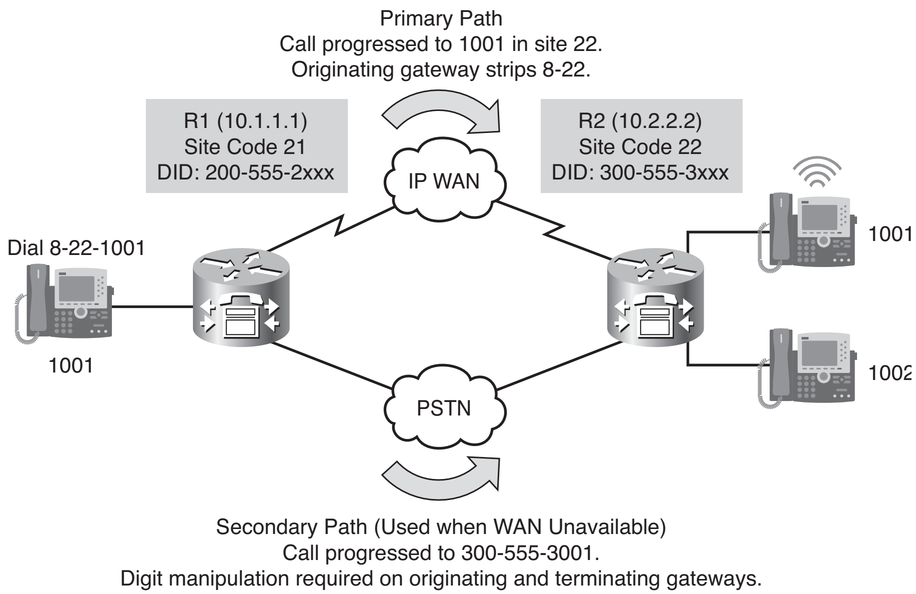


Figure 4-11 *Call Routing Example*

Dial Plan Components

A dial plan is the central part of any telephony solution and defines how calls are routed and interconnected. A dial plan consists of various components, which can be used in various combinations. This section describes the components of a dial plan and how they are used on Cisco IOS gateways.

Defining Dial Plans

Although most people are not acquainted with dial plans by name, they use them daily. A dial plan describes the process of determining how many and which digits are necessary for call routing. If dialed digits match a defined pattern of numbers, the call can processed and forwarded.

Designing dial plans requires knowledge of the network topology, dialing patterns, and traffic routing requirements. There are no dynamic routing protocols for E.164 telephony addresses. VoIP dial plans are statically configured on gateway and gatekeeper platforms.

A dial plan consists of these components:

- **Endpoint addressing (numbering plan):** Assigning directory numbers to all endpoints and applications (such as voice-mail systems, auto attendants, and conferencing systems) enables you to access internal and external destinations.
- **Call routing and path selection:** Multiple paths can lead to the same destination. A secondary path can be selected when a primary path is not available. For example, a call can be transparently rerouted over the PSTN during an IP WAN failure.
- **Digit manipulation:** Manipulation of numbers before routing a call might be required (for example, when a call is rerouted over the PSTN). This can occur before or after the routing decision.
- **Calling privileges:** Different privileges can be assigned to various devices, granting or denying access to certain destinations. For example, lobby phones might reach only internal destinations, while executive phones could have unrestricted PSTN access.
- **Call coverage:** You can create special groups of devices to manage incoming calls for a certain service according to different rules (top-down, circular hunt, longest idle, or broadcast). This also ensures that calls are not dropped without being answered.

While these dial plan components can be implemented using a Cisco Unified Communications Manager server, the focus in this book is on implementing these dial plan components on a Cisco IOS router acting as a call agent.

Dial Plan Implementation

Cisco IOS gateways, including Cisco Unified Communications Manager Express and Cisco Unified Survivable Remote Site Telephony (SRST), support all dial plan components. Table 4-6 provides an overview of the methods that Cisco IOS gateways use to implement dial plans.

Table 4-6 *Dial Plan Implementation*

Dial Plan Component	Cisco IOS Gateway
Endpoint addressing	POTS dial peers for FXS ports, ephone-dn, and voice register directory number
Call routing and path selection	Dial peers
Digit manipulation	**voice translation profile**, **prefix**, **digit-strip**, **forward-digits**, and **num-exp**
Calling privileges	Class of Restriction (COR) names and lists
Call coverage	Call hunt, hunt groups, call pickup, call waiting, call forwarding, overlaid directory numbers

Dial Plan Requirements

Figure 4-12 shows a typical dial plan scenario. Calls can be routed via either an IP WAN link or a PSTN link, and routing should work for inbound and outbound PSTN calls, intrasite calls, and intersite calls.

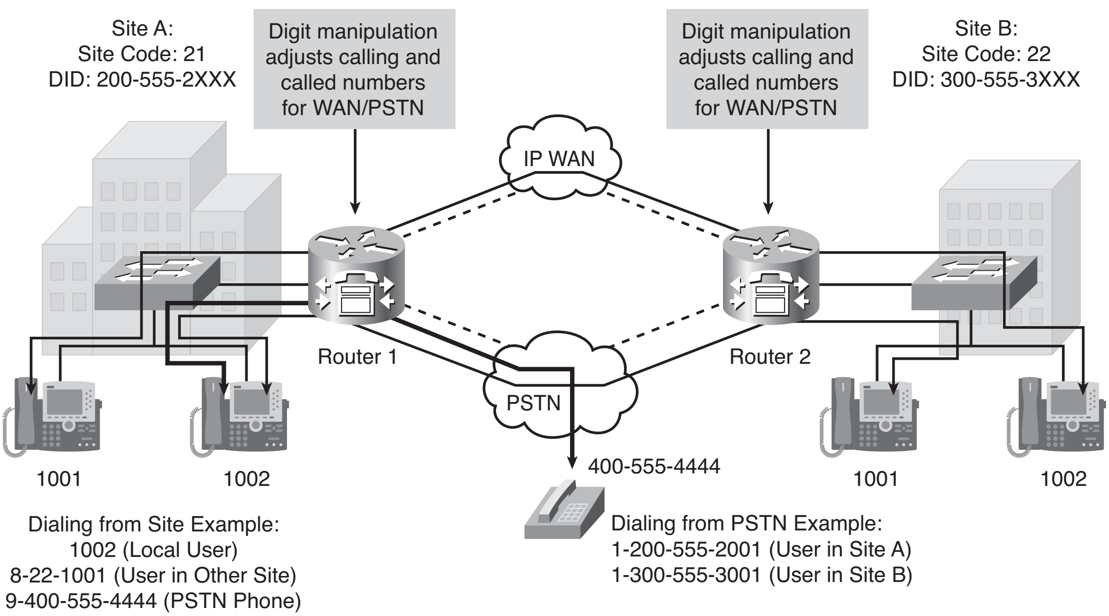


Figure 4-12 *Dial Plan Requirements*

The dial plan defines the rules that govern how a user reaches any destination. Definitions include the following:

- **Extension dialing:** Determines how many digits must be dialed to reach an extension
- **Extension addressing:** Determines how many digits are used to identify extensions
- **Dialing privileges:** Allows or disallows certain types of calls
- **Path selection:** Selects one path from several parallel paths
- **Automated selection of alternate paths in case of network congestion:** For example, using a local carrier for international calls if the preferred international carrier is unavailable
- **Blocking of certain numbers:** Prevents unwarranted high-cost calls
- **Transformation of the called-party number: Allows appropriate digits (that is, DNIS digits) to be presented to the PSTN or a call agent**
- **Transformation of the calling-party number:** Allows appropriate caller-ID information (that is, ANI information) to be presented to a called party

A dial plan suitable for an IP telephony system is not fundamentally different from a dial plan that is designed for a traditional telephony system. However, an IP-based system presents additional possibilities. In an IP environment, telephony users in separate sites can be included in one unified IP-based system. These additional possibilities presented by IP-based systems require you to think about dial plans in new ways.

Endpoint Addressing Considerations

Reachability of internal destinations is provided by assigning directory numbers to all endpoints (such as IP phones, fax machines, and analog phones) and applications (such as voice-mail systems, auto-attendants, and conferencing systems). An example of number assignment is provided in Figure 4-13.

The number of dialable extensions determines the quantity of digits needed to dial extensions. For example, a four-digit abbreviated dial plan cannot accommodate more than 10,000 extensions (from 0000 through 9999). If 0 and 9 are reserved as operator code and external access code, respectively, the number range is further reduced to 8000 (1000 through 8999). If direct inward dialing (DID) is enabled for PSTN calls, the DID numbers are mapped to internal directory numbers.

The most common issue with endpoint addressing is related to the mapping of internal extensions to available DID ranges assigned by the PSTN. When the DID range does not cover the entire internal address scope, an auto-attendant can be used to route calls between the PSTN and the internal network.

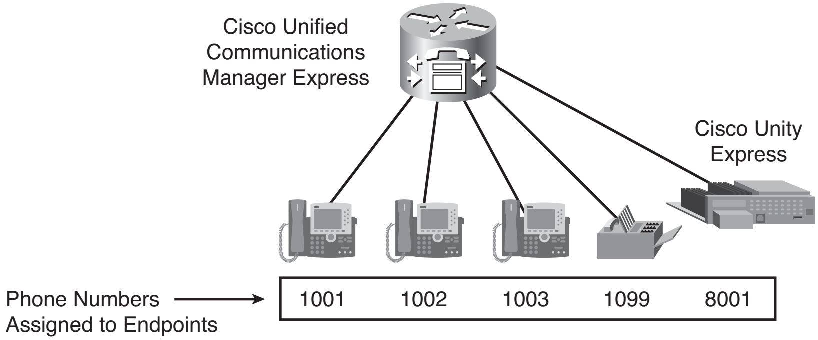


Figure 4-13 *Endpoint Addressing*

One of the biggest challenges when creating an endpoint addressing scheme for a multisite installation is to design a flexible and scalable dial plan that has no impact on the end user. The existing overlapping directory numbers present a typical issue that must be addressed.

Endpoint addressing is primarily managed by a call agent, such as Cisco Unified Communications Manager or Cisco Unified Communications Manager Express.

Call Routing and Path Selection

Call routing and path selection are the dial plan components that define where and how calls should be routed or interconnected. Call routing usually depends on the called number (that is, destination-based call routing is usually performed). This is similar to IP routing, which also relies on destination-based routing. Multiple paths to the same destination might exist, especially in multisite environments (for example, a path using an IP connection or a path using a PSTN connection). Path selection helps you decide which of the available paths should be used.

A voice gateway might be involved with call routing and path selection, depending on the protocol and design that is used. For example, an H.323 gateway will at least route the call between the call leg that points to the call handler and the call leg that points to the PSTN. When a Cisco IOS gateway performs call routing and path selection, the key components that are used are dial peers.

In Figure 4-14, if a user dials an extension number in another location (8-22-2001), the call should be sent over the IP WAN. If the WAN path is unavailable (due to network failure, insufficient bandwidth, or no response), the call should use the local PSTN gateway as a backup and send the call through the PSTN.

For PSTN-routed calls, digit manipulation should be configured on the gateway to transform the internal numbers to E.164 numbers that can be used in the PSTN.

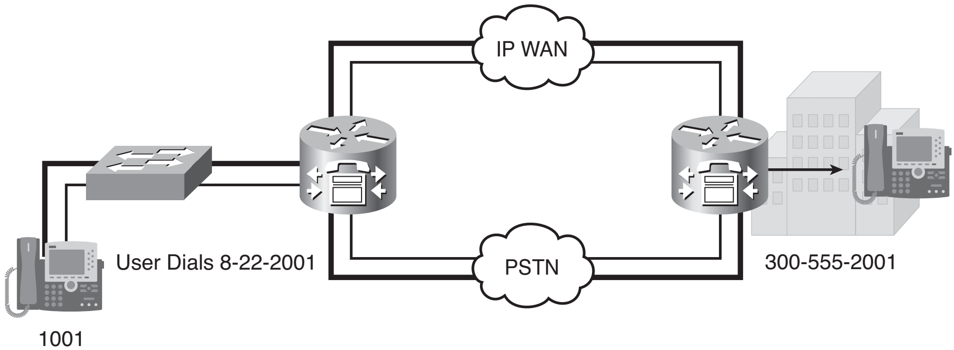


Figure 4-14 *Path Selection Example*

PSTN Dial Plan Requirements

A PSTN dial plan has three key requirements:

- **Inbound call routing:** Incoming calls from the PSTN must be routed correctly to their final destination, which might be a directly attached phone or endpoints that are managed by Cisco Unified Communications Manager or Cisco Unified Communications Manager Express. This inbound call routing also includes digit manipulation to ensure that an incoming called number matches the pattern expected by the final destination.
- **Outbound call routing:** Outgoing calls to the PSTN must be routed to the voice interfaces of the gateway (for example, a T1/E1 or a Foreign Exchange Office [FXO] connection). As with inbound calls, outbound calls might also require digit manipulation to modify a called number according to PSTN requirements. This outbound call routing usually includes stripping of any PSTN access code that might be included in the original called number.
- **Correct PSTN calling-party number presentation:** An often-neglected aspect is the correct calling number presentation for both inbound and outbound PSTN calls. The calling number for inbound PSTN calls is often left untouched, which might have a negative impact on the end-user experience. The calling number that is presented to the end user should include the PSTN access code and any other identifiers that are required by the PSTN to successfully place a call using that calling number (for example, using the missed calls directory).

Inbound PSTN Calls

Figure 4-15 shows how gateways manage inbound PSTN calls.

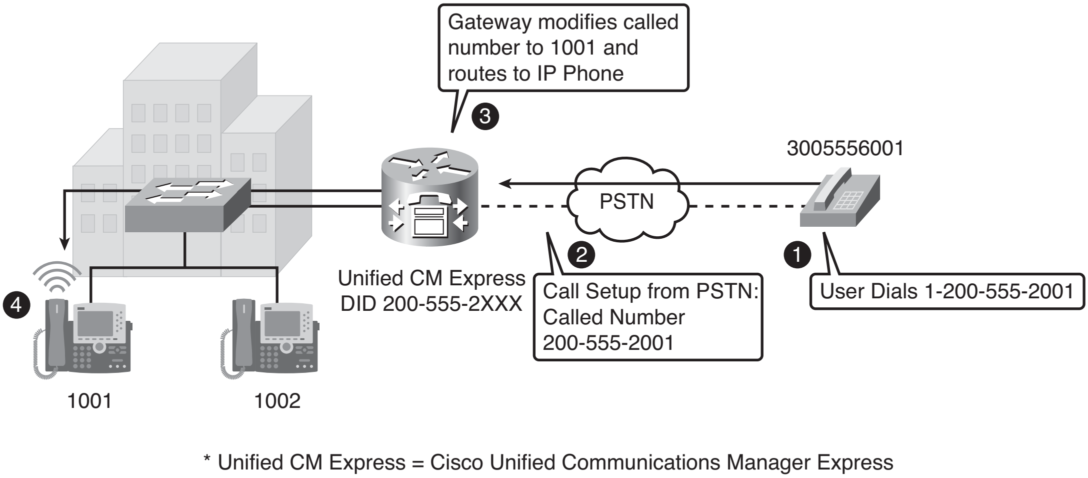


Figure 4-15 *Inbound PSTN Calls*

The site consists of a Cisco Unified Communications Manager Express system with endpoints registered to it, connected to the PSTN over a digital trunk. The DID range of the PSTN trunk is 2005552XXX, and phones use the extension range 1XXX. The processing of an inbound PSTN call occurs in these steps:

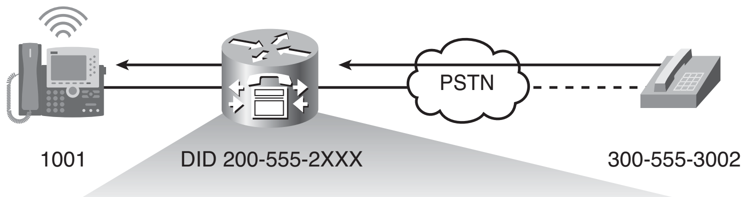
1. A PSTN user places a call to 1-200-555-2001 (that is, an endpoint with internal extension 1001).
2. The call setup message is received by the gateway with a called number of 200-555-2001.
3. The gateway modifies the called number to 1001 and routes the call to the voice port that was created when a Cisco Unified IP Phone registered with Cisco Unified Communications Manager Express.
4. The phone rings.

Figure 4-16 provides a description of the required number manipulation when a gateway receives an inbound PSTN call.

Both the called and calling numbers must be transformed:

- The called number can be converted from the public E.164 format to the internal number used for internal dialing. This transformation ensures that the call matches the outbound dial peer that is automatically created at endpoint registration. Directory numbers are commonly configured with their internal extensions.
- The calling number must be presented to the callee in a way that allows callback. Because access codes are commonly used to reach external destinations, a calling number forwarded to the destination should include an access code. Optionally, some

region-specific prefixes might have to be added, such as the long-distance prefix in the NANP region, 1.



	Incoming	Outgoing
Called Party Number	200-555-2001	1001
Calling Party Number	300-555-3002	9-1-300-555-3002

Figure 4-16 *Numbers in Inbound PSTN Calls*

Outbound PSTN Calls

Figure 4-17 shows the call flow for an outbound call.

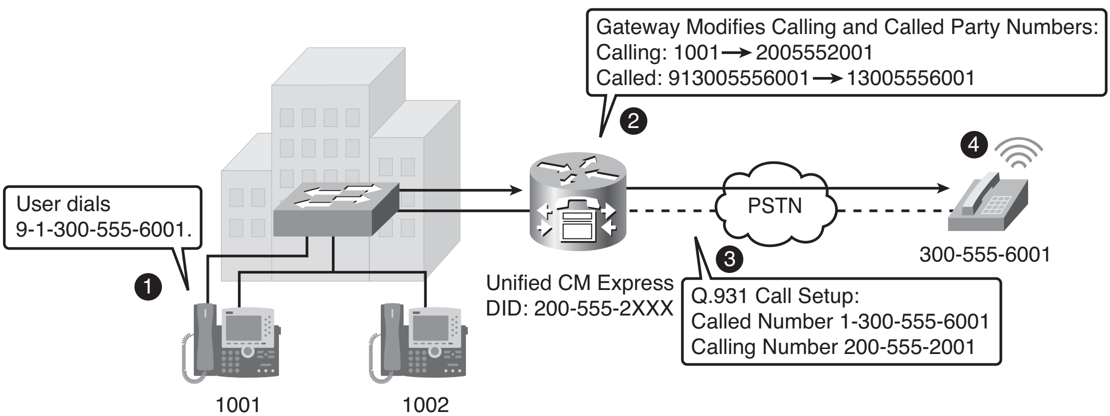


Figure 4-17 *Outbound PSTN Calls*

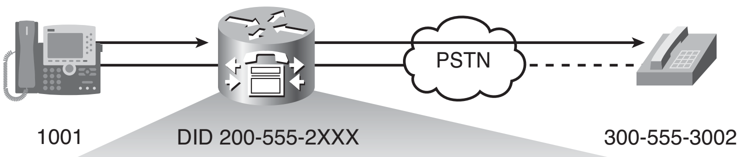
The site consists of a Cisco Unified Communications Manager Express system with endpoints registered to it, connected to the PSTN over a digital trunk. The access code is 9. The processing of an outbound PSTN call occurs in these steps:

1. A user places a call to 9-1-300-555-6001 from the phone with extension 1001.
2. The gateway accepts the call and modifies the called number to 1-300-555-6001, stripping off the PSTN access code 9. The gateway also modifies the calling number to 200-555-2001 by prefixing the area code and local code and mapping the four-digit extension to the DID range.

3. The gateway sends out a call setup message with the called number set to 1-300-555-6001 and the calling number set to 200-555-2001.

4. The PSTN subscriber telephone at 300-555-6001 rings.

Figure 4-18 summarizes the requirements for number manipulation when a gateway processes an outbound PSTN call.



	Incoming	Outgoing
Called Party Number	9-1-300-555-3002	1-300-555-3002
Calling Party Number	1001	200-555-2001

Figure 4-18 *Numbers in Outbound PSTN Calls*

Both the called and calling numbers must be transformed as follows:

- The called number processing involves the stripping of the access code. Optionally, some region-specific prefixes might have to be added, such as the long-distance prefix in the NANP region, 1.
- The calling number must be converted from the internal extension to the public E.164 format. If the outgoing calling number is not configured on the gateway, the telco operator sets the value to the subscriber number, but this setting might be inaccurate if a DID range is available. For example, the calling number for a call originating from 1002 would be set to 222-555-2000. Setting the calling number is considered a good practice and ensures proper callback functionality.

ISDN Dial Plan Requirements

The type of number (TON) or nature of address indicator (NAI) parameter indicates the scope of the address value, such as whether it is an international number (including the country code) a "national," or domestic number (without country code), and other formats such as "local" format (without an area code). It is relevant for E.164 (regular telephone) numbers.

The TON is carried in ISDN-based environments. Voice gateways must consider the TON when transforming the called and calling numbers for ISDN calls.

ISDN networks impose new number manipulation needs, in addition to the typical requirements for PSTN calls:

- **Correct PSTN inbound ANI presentation, depending on TON:** Some ISDN networks present the inbound ANI as the shortest dialable number combined with the TON. This treatment of the ANI can be a potential problem, because simply prefixing the PSTN access code might not result in an ANI that can be called back. A potential problem can be solved by proper digit manipulation on gateways.
- **Correct PSTN outbound ANI presentation, depending on TON:** Some ISDN networks and PBXs might expect a certain numbering plan and TON for both DNIS and ANI. Using incorrect flags might result in incomplete calls or an incorrect DNIS and ANI presentation. Digit manipulation can be used to solve these issues.

Note The calling-party number in ISDN is called *Automatic Number Identification* (ANI). The called-party number in ISDN is referred to as *Dialed Number Identification Service* (DNIS).

In Figure 4-19, three different calls are received at the voice gateway. The first call is received from the local area with a subscriber TON and a seven-digit number. This number only needs to be prefixed with access code 9. The second call, received with a national TON and ten digits, is modified by adding access code 9 and the long-distance number 1, all of which are required for placing calls back to the source of the call. The third call is received from oversees with an international TON. For this call, the access code 9 and 011 must be added to the received number, as a prefix to the country code.

Digit Manipulation

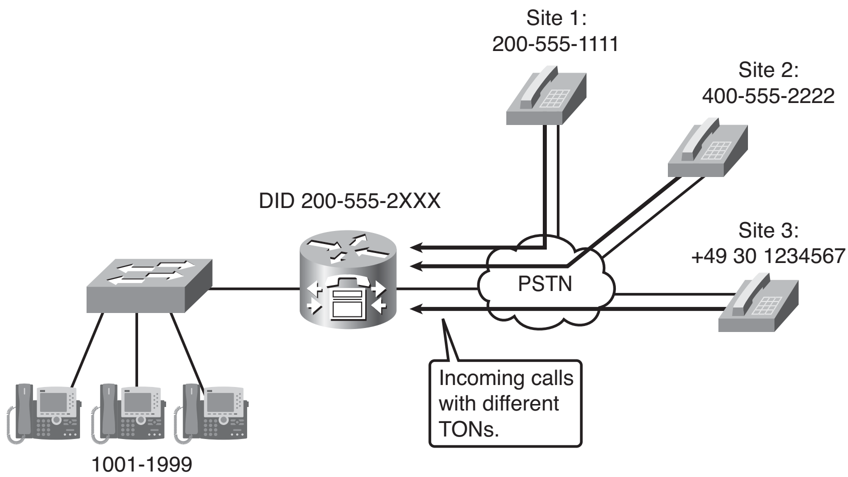
Digit manipulation is closely related to call routing and path selection. Digit manipulation is performed for inbound calls to achieve these goals:

- Adjust the called-party number to match internally used patterns
- Present the calling-party number as a dialable number

Digit manipulation is implemented for outbound calls to ensure the following:

- Called number satisfies the internal or PSTN requirements
- Calling number is dialable and provides callback if sufficient PSTN DID is available

Digit manipulation is covered in Chapter 5, "Implementing Dial Plans."



Site	TON	ANI	Required ANI Transformation
1	Subscriber	555-1111	9-555-1111
2	National	400-555-2222	9-1-400-555-2222
3	International	49-30-1234567	9-011-49-30-1234567

Figure 4-19 *Inbound ISDN Calls*

Calling Privileges

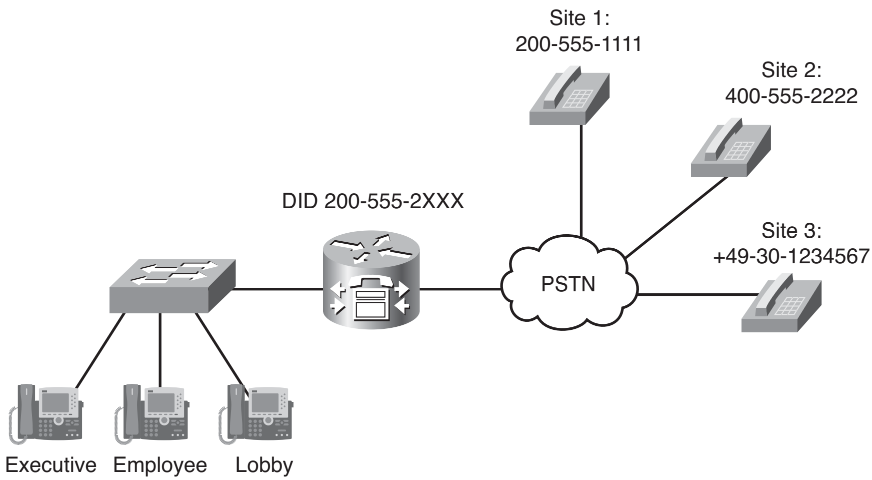
Calling privileges are equivalent to firewalls in networking. They define call permissions by specifying which users can dial given destinations. The two most common areas of deploying calling privileges are as follows:

- Policy-defined rules to reach special endpoints. For example, manager extensions cannot be reached from a lobby phone.
- Billing-related rules that are deployed to control telephony charges. Common examples include the blocking of costly service numbers and restricting international calls.

Calling privileges are referred to as a "Class of Service," but should not be confused with the Layer 2 class of service (CoS) that describes quality of service (QoS) treatment of traffic on Layer 2 switches.

Figure 4-20 illustrates the typical deployment of calling privileges. The internal endpoints are classified into three roles: executive, employee, and lobby. Each role has a set of dialable PSTN destinations that is associated with it. The executives can dial any PSTN number, the employees are allowed to dial any external numbers except international destinations, and the lobby phones permit the dialing of local numbers only.

The deployment of calling privileges is covered in Chapter 5.



User	Call Permission
Executive	Site 1 (Local), Site 2 (Long Distance), Site 3 (International)
Employee	Site 1 (Local), Site 2 (Long Distance)
Lobby	Site 1 (Local)

Figure 4-20 *Calling Privileges Example*

Call Coverage

Call coverage features are used to ensure that all incoming calls to Cisco Unified Communications Manager Express are answered by someone, regardless of whether the called number is busy or does not answer.

Call coverage can be deployed for two different scopes:

- **Individual users:** Features such as call waiting and call forwarding increase the chance of a call being answered by giving it another chance for a connection if the dialed user cannot manage the call.
- **User groups:** Features such as call pickup, call hunt, hunt groups, and overlaid directory numbers provide different ways to distribute the incoming calls to multiple numbers and have them answered by available endpoints.

Call Coverage Features

Cisco voice gateways provide various call coverage features:

- **Call forwarding:** Calls are automatically diverted to a designated number on busy, no answer, all calls, or only during night-service hours.

- **Call hunt:** The system automatically searches for an available directory number from a matching group of directory numbers until the call is answered or the hunt is stopped.
- **Call pickup:** Calls to unstaffed phones can be answered by other phone users using a softkey or by dialing a short code.
- **Call waiting:** Calls to busy numbers are presented to phone users, giving them the option to answer or let them be forwarded.
- **Basic automatic call distribution (B-ACD):** Calls to a pilot number are automatically answered by an interactive application that presents callers with a menu of choices before sending them to a queue for a hunt group.
- **Hunt groups:** Calls are forwarded through a pool of agents until answered or sent to a final number.
- **Overlaid ephone-dn:** Calls to several numbers can be answered by a single agent or multiple agents.

Summary

The main topics covered in this chapter are the following:

- Public and private numbering plans were contrasted, along with the characteristics and requirements of each.
- You were introduced to the components of dial plans and their functions. These components include endpoint addressing, call routing and path selection, digit manipulation, calling privileges, and call coverage.

Chapter Review Questions

The answers to these review questions are in the appendix.

1. Which dial plan component is responsible for choosing the appropriate path for a call?
   a. Endpoint addressing
   b. Call routing and path selection
   c. Call coverage and path selection
   d. Calling privileges
2. What is the dial plan component called endpoint addressing responsible for assigning to the endpoints?
   a. IP addresses
   b. E.164 addresses
   c. Gateways
   d. Directory numbers

3. Which option implements call routing and path selection on Cisco IOS gateways?

   a. Call-routing tables

   b. Dialer maps

   c. Dial peers

   d. Route patterns

4. What is one way to implement call coverage?

   a. COR

   b. Pilot numbers

   c. Digit manipulation

   d. Endpoint addressing

5. Which of the following are characteristics of a scalable dial plan? (Choose three.)

   a. Backup paths

   b. Full digit manipulation

   c. Hierarchical numbering plan

   d. Dial plan logic distribution

   e. Granularity

   f. High availability

6. Which of the following options are key requirements for a PSTN dial plan? (Choose three.)

   a. Internal call routing

   b. Inbound call routing

   c. Outbound call routing

   d. Correct PSTN ANI presentation

   e. Internet call routing

7. What might some ISDN networks and PBXs expect along with a certain numbering plan for both DNIS and ANI?

- **a.** ToS
- **b.** TON
- **c.** QoS
- **d.** CoS

8. Which command should be used to display information for all voice dial peers?

- **a.** **show dial-peer voice summary**
- **b.** **show dial-peer voice all**
- **c.** **show dial-peer summary**
- **d.** **show dial-peer all**

9. Which function best describes a numbering plan?

- **a.** Determines routes between source and destination
- **b.** Defines a telephone number of a voice endpoint or application
- **c.** Performs digit manipulation when sending calls to the PSTN
- **d.** Performs least-cost routing for VoIP calls

10. Which worldwide prefix scheme was developed by the ITU to standardize numbering plans?

- **a.** E.164
- **b.** G.114
- **c.** G.164
- **d.** E.114

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