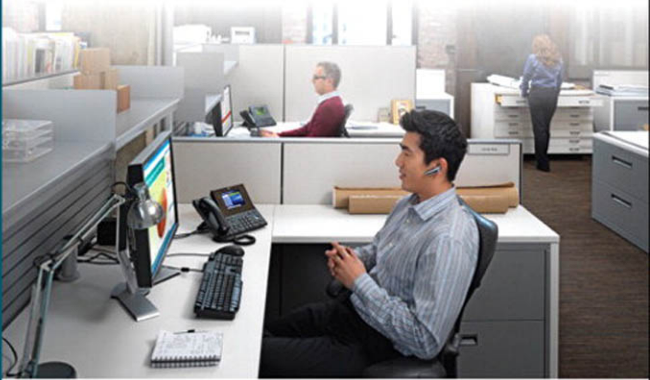
Implementing Cisco Unified Communications Manager, Part 1 (CIPT1)

Foundation Learning Guide Second Edition



Josh Finke, CCIE® No. 25707
Dennis Hartmann, CCIE® No. 15651

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About the Authors

Josh Finke, CCIE No. 25707, is the practice director for collaboration and networking at Iron Bow Technologies, a Cisco Gold and Master Unified Communications Partner. Josh was previously a lead instructor and director of operations for Internetwork Expert, a leading CCIE training company. Josh has multiple certifications, including the Cisco CCIE Voice, CCNP, CCDP, CCNA, CCDA, and Cisco Meeting Place Specialist. Josh specializes in Cisco UC, routing and switching, and enterprise network design. Josh started working with Cisco networking technologies in 2000 and later became one of the youngest Voice CCIEs in the world. He lives with his wife in Seattle, Washington.

Dennis J. Hartmann, CCIE No. 15651, is a Unified Communications consultant. Dennis is also a lead instructor at Global Knowledge. Dennis was first exposed to CallManager during the CallManager 2.0 time frame when Cisco acquired Selsius. Dennis has various certifications, including the Cisco CCVP, CCSI, CCNP, CCIP, and the Microsoft MCSE. Dennis has worked for various Fortune 500 companies, including AT&T, Sprint, Merrill Lynch, KPMG, and Cabletron Systems. Dennis lives with his wife and children in Hopewell Junction, New York.

About the Technical Reviewer

Manny Richardson, CCIE No. 6056, is a Voice and Routing and Switching CCIE. He is a design and implementation engineer consultant with MARTA and the City of Atlanta in Atlanta, Georgia. He is also an instructor with more than five years of worldwide teaching experience. He has worked in the field of networking for 12 years, with the last three years primarily focused on Cisco Voice.

Dedication

I dedicate this book to the love and support in my life, Alissa.

Acknowledgments

Thank you to my wife, my family, and all of those who have supported and believed in me.

Thank you to Brett Bartow, Chris Cleveland, and the entire Cisco Press team, who are excellent at what they do and made this book possible.

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).
- *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

Professional certifications have been an important part of the computing industry for many years and will continue to become more important. Many reasons exist for these certifications, but the most popularly cited reason is that of credibility. All other considerations held equal, a certified employee/consultant/job candidate is considered more valuable than one who is not.

Goals and Methods

The most important goal of this book is to provide you with knowledge and skills in Unified Communications, deploying the Cisco Unified Communications Manager product. Another goal of this book is to help you with the Cisco IP Telephony (CIPT) Part 1 exam, which is part of the Cisco Certified Network Professional Voice (CCNP) certification. The methods used in this book are designed to be helpful in both your job and the CCNP Voice Cisco IP Telephony exam. This book provides questions at the end of each chapter to reinforce the chapter content. Additional test-preparation software from companies such as www.selftestsoftware.com gives you additional test-preparation questions to arm you for exam success.

The organization of this book helps you discover the exam topics that you need to review in more depth, helps you fully understand and remember those details, and helps you test the knowledge you have retained on those topics. This book does not try to help you pass by memorization, but helps you truly learn and understand the topics. The Cisco IP Telephony Part 1 exam is one of the foundation topics in the CCNP Voice certification. The knowledge contained in this book is vitally important for you to consider yourself a truly skilled Unified Communications (UC) engineer. The book helps you pass the Cisco IP Telephony 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
- Providing practice exercises on the topics and the testing process through test questions at the end of each chapter

Who Should Read This Book

This book is designed to be both a general Cisco Unified Communications Manager book and a certification preparation book. This book provides you with the knowledge required to pass the CCNP Voice Cisco IP Telephony exam for CIPT Part 1.

Why should you want to pass the CCNP Voice Cisco IP Telephony exam? The first CIPT test is one of the milestones toward getting the CCNP Voice certification. The CCNP Voice could mean a raise, promotion, new job, challenge, success, or recognition, but ultimately you determine what it means to you. Certifications demonstrate that you are serious about continuing the learning process and professional development. In technology, it is impossible to stay at the same level when the technology all around you is advancing. Engineers must continually retrain themselves, or they find themselves with out-of-date, commodity-based skill sets.

Strategies for Exam Preparation

The strategy you use for exam preparation might be different than strategies used by others. It will be based on skills, knowledge, experience, and finding the recipe that works best for you. If you have attended the CIPT course, you might take a different approach than someone who learned Cisco Unified Communications Manager on the job. Regardless of the strategy you use or your background, this book is designed to help you get to the point where you can pass the exam. Cisco exams are quite thorough, so don't skip any chapters.

How This Book Is Organized

The book covers the following topics:

- **Chapter 1, "Cisco Unified Communications Manager Architecture,"** discusses the architecture and all the components involved. CUCM hardware requirements, operating system, database, signaling, licensing, and database replication are discussed.
- **Chapter 2, "Deployment Models,"** covers the deployment models in which CUCM can be used. This chapter introduces the technologies required for the different UC models. The advantages and disadvantages of each deployment model are considered.
- **Chapter 3, "Cisco Unified Communications Manager Services and Initial Configuration Settings,"** examines the network configuration, Network Time Protocol (NTP), and DHCP configuration options of CUCM. The chapter also covers frequently adjusted CUCM enterprise and service parameters.
- **Chapter 4, "Managing User Accounts in Cisco Unified Communications Manager,"** examines user account configuration in CUCM administration, the Bulk Administration Tool (BAT), and the Lightweight Directory Access Protocol (LDAP).
- **Chapter 5, "Cisco Unified Communications Manager Endpoints,"** covers the various Cisco Unified IP Phones and the features that they support. Third-party Session Initiation Protocol (SIP) endpoint support is covered, in addition to the Cisco IP Phone boot cycle and registration process.
- **Chapter 6, "Cisco Catalyst Switches,"** covers the power and voice VLAN requirements of the Cisco IP Phone. The Catalyst switch configurations are examined for both Native IOS and CatOS switches. The Cisco and IEEE power specifications are also covered.

- **Chapter 7, "Implementing and Hardening IP Phones,"** covers the methods for endpoint (phone) registration within CUCM, including manual registration and autoregistration, and the tools available for each process.
- **Chapter 8, "Implementing PSTN Gateways in Cisco Unified Communications Manager,"** covers the implementation of the gateways used in conjunction with CUCM. MGCP, H.323, and SIP gateways are each explored.
- **Chapter 9, "Call-Routing Components,"** covers the fundamentals of call routing and a public switched telephone network (PSTN) dial plan. Digit analysis and path selection are achieved through the use of the router pattern, route list, and route group CUCM configuration elements.
- **Chapter 10, "Calling Privileges,"** covers the process of class of service through the use of partitions and calling search spaces. The chapter also covers time-of-day routing through the use of time periods and time schedules.
- **Chapter 11, "Digit Manipulation,"** covers the process of digit manipulation through calling and called party transformation masks, translation patterns, prefixing digits, and digit discard instructions (DDI).
- **Chapter 12, "Call Coverage,"** covers the topic of call-coverage paths through the use of a hunt pilot, hunt list, and line groups. Call-hunting flow is discussed through the various distribution algorithms supported in CUCM.
- **Chapter 13, "Media Resources,"** discusses the media resources supported in and through CUCM. The media resource topics include music on hold (MoH), conference bridges, annunciators, transcoders, and media termination points. Media resource allocation is discussed through the application of CUCM Media Resource Manager (MRM), media resource group list, and media resource groups.
- **Chapter 14, "Phone Services,"** explores the concept of phone services and their use within CUCM, including configuration, subscriptions, and considerations.
- **Chapter 15, "Presence-Enabled Speed Dials and Lists,"** covers presence theory and configuration through the use of presence groups, presence speed dials, and presence calling search spaces.
- **Chapter 16, "Implementing Cisco Unified Mobility,"** covers the concept and configuration of mobility for CUCM end users using constructs such as single-number reach and mobile voice access.
- **Appendix A, "Answers to Review Questions,"** lists the answers to the chapter review questions.

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Chapter 11

Digit Manipulation

Upon completing this chapter, you will be able to use digit manipulation techniques to change calling party (caller ID) and called party (dialed digits) information, and be able to meet the following objectives:

- Describe when to use digit manipulation in CUCM
- Describe CUCM digit manipulation operation
- Identify CUCM digit manipulation configuration options
- Describe how to use external phone number masks
- Describe how to use translation patterns
- Describe how to use transformation masks in CUCM
- Describe how to use digit stripping and digit prefixes in CUCM
- Describe how to use significant digits in CUCM
- Describe how to use global transformations in CUCM
- Describe how to use incoming number prefixes in CUCM

Users of a phone system need to communicate with a variety of destinations. Destinations might be located within the same site, different sites within the same company, and other companies located within the same country or different countries. Completing various types of calls often requires dialing access codes or prefix numbers. It is often prudent to restrict users from dialing certain destinations that could incur high costs, such as 1-900 pay service phone numbers and international dialing.

Users should be provided with a dial plan with the lowest amount of complexity. Cisco Unified Communications Manager (CUCM) has the capability to provide digit manipulation, which achieves the goal of adding or subtracting digits to comply with a private or public numbering plan. Toll bypass calls that are routed over the data network should be transparently rerouted across the public switched telephone network (PSTN) when WAN resources are not available or are fully utilized.

This chapter describes digit manipulation tools that allow a CUCM administrator to implement flexibility and transparency in the dial plan of the company. The chapter covers external phone number masks, digit prefixing, digit stripping, transformation masks, translation patterns, and significant digits.

CUCM Digit Manipulation

Digit manipulation is often used to change calling party numbers for caller ID purposes on outgoing PSTN calls. Digit manipulation is also used to strip PSTN access codes before CUCM routes calls to the gateway (PSTN). Digit manipulation is required for abbreviated dialing and to properly route inbound calls from the PSTN where an abbreviated internal dial plan exists. Inbound calls from the PSTN can be received with a ten-digit called party length, but the internal dial plan might use only a subset of those numbers (four or five digits). These inbound calls would need to have the called party number transformed to the digit length used in the internal dial plan. PSTN access codes do not adhere to public standards, so they need to be stripped from the called party number before routing the call to the PSTN. Most organizations use the number 0, 8, or 9 as the access code for PSTN dialing. The calling party number also needs to be changed from the abbreviated internal extension number to a full E.164 PSTN number to allow easier redial.

Mechanics of CUCM Digit Manipulation

An IP phone with extension 1002 in Figure 11-1 calls a phone on the PSTN with a called party number of 408 555-111. The user at extension 1002 must first dial a PSTN access code of 9 to route a call to the PSTN. The PSTN Class 5 switch will not be able to route the call unless the access code is dialed before the PSTN number. The calling party number is transformed into a ten-digit pattern so that the PSTN is presented with a routable caller ID of 706 555-1002, not the extension of 1002. Four-digit dialing is not possible in the North American Numbering Plan (NANP).

Note In some countries, the calling party number must be set to the correct PSTN number of the used PSTN subscriber line or trunk.

Table 11-1 displays some often-used digit manipulation requirements and the methods in which they are handled in CUCM.

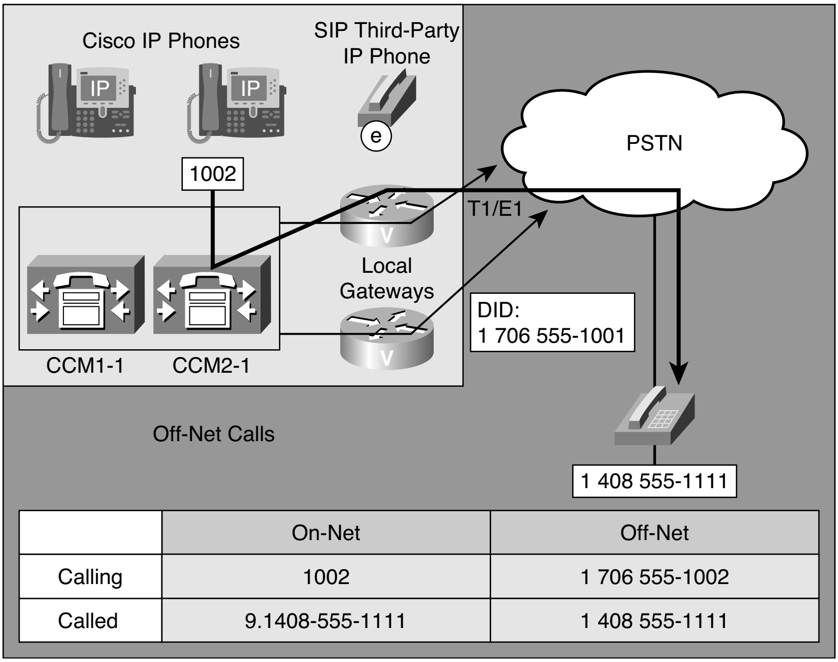

Figure 11-1 *Digit Manipulation Overview*

Table 11-1 *Digit Manipulation Methods*

Requirement	**Call Type**
Expand calling party directory number to full E.164 PSTN number	Internal to PSTN
Strip PSTN access code	Internal to PSTN
Expand abbreviated number	Internal to internal
Convert E.164 PSTN called party directory number to internal number	PSTN to internal
Expand endpoint directory numbers to accommodate overlapping dial plan	Internal to internal PSTN to internal

Figure 11-2 illustrates an internal caller at extension 1005 dialing a PSTN number using a PSTN access code of 9 followed by the 11-digit PSTN number. The process of digit manipulation occurs as follows:

1. Extension 1005 dials 9-1-303-555-6007.

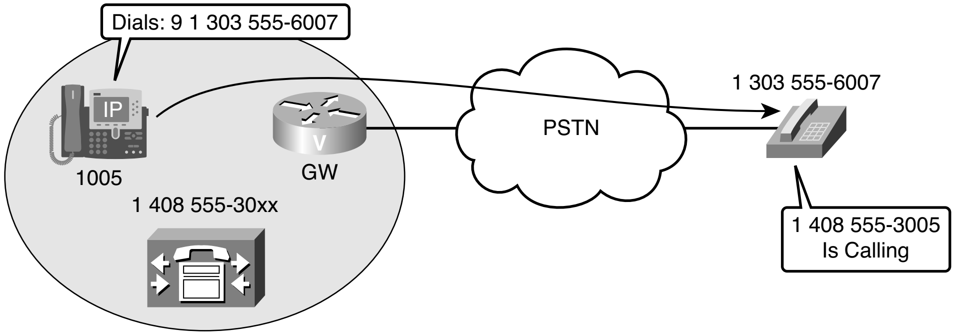

Figure 11-2 *Outgoing Call to the PSTN*

2. The dialed number (called party) matches the 9.! route pattern, where digit manipulation is taking place. For the sake of simplicity, let's imagine that there is only one gateway with this very simple dial plan. The route pattern is pointed directly to the gateway where the following is configured:
   - Called party transformations > Discard digits: PreDot
   - Calling party transformations: 40855530XX
   - Route the call to the gateway
3. CUCM provides digit stripping of the access code from the called party and sends 11 digits (1-303-555-6007) to the PSTN through the gateway. The calling party number is modified from 1005 to 408 555-3005.
4. The PSTN phone at (303) 555-6007 rings and sees 4085553005 as the calling number.

Calling and called party transformations are configured at the route pattern level in the example, but these digit manipulation techniques are normally preferred at the route list detail level of the route list (per route group). The calling party transformation is often performed first at the external phone number mask configuration level. The external phone number mask is a directory number (DN) configuration parameter that will display a phone's ten-digit PSTN phone number to the end user at the phone. External phone number masks are also used when Automated Alternate Routing (AAR) reroutes a call over a call admission control (CAC) call rejection in a centralized call processing model. AAR is covered in detail in the Cisco Press book *Implementing Cisco Unified Communications Manager, Part 2 (CIPT2) Foundation Learning Guide*.

Figure 11-3 illustrates a call coming from the PSTN to an internal phone. The call-routing process from the gateway is as follows:

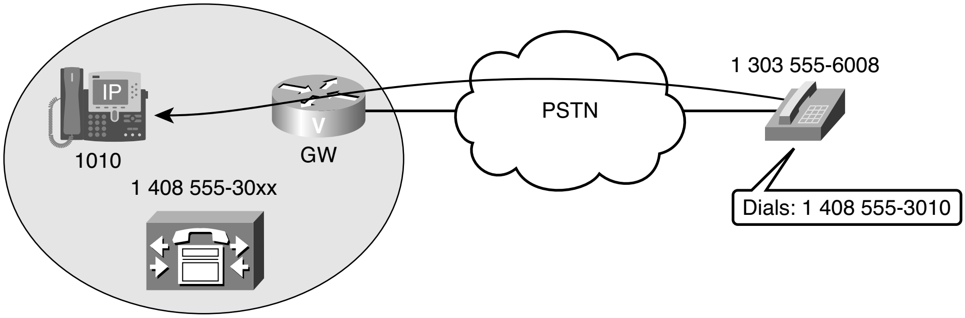

Figure 11-3 *Incoming Call from the PSTN*

1. The PSTN phone calls the full E.164 number of the destination. The call is received at the PSTN gateway with a called party number ten digits in length. Digit manipulation is performed to convert the inbound ten-digit called number to a four-digit number matching the internal dial plan. Digit manipulation might occur in the translation configuration of the gateway if the gateway is an H.323 or Session Initiation Protocol (SIP) gateway. Media Gateway Control Protocol (MGCP) gateways can perform digit manipulation on an individual endpoint basis using called party transformation patterns. Digit manipulation can be configured in CUCM if the gateways are H.323 or SIP using the same called party transformation patterns beginning with CUCM version 7.0.

2. The called party number received from the PSTN can also be manipulated to align to the internal dial plan using a translation pattern that matches the called party number digits received from the provider. The translation pattern then applies any calling and called party digit manipulations in a manner very similar to the digit manipulation performed at the route list detail level of the route list. Translation patterns are unique in the respect that they do not forward calls to a trunk or gateway device. Translations are leveraged only to perform digit manipulation.

   Translation patterns are normally not necessary to change the incoming called party E.164 number to an internal directory number unless the digits received from the carrier don't map directly to the internal dial plan. The calling party transformation mask of the translation pattern can be used to insert 91 into the calling party number, enabling callback functionality from the Cisco IP Phone's call history (missed and received calls). Calling party digit manipulation can be more granular if the call is coming in over ISDN Q.931 signaling or H.323 Q.931 signaling. At the time of this writing, SIP trunks do not support the passing of numbering plan type (subscriber, national, international, or unknown). Q.931 signaling used in ISDN and H.323 supports the passing of numbering plan type, allowing the calling party number to be transformed as follows:

   - Calling number (prefix 9) for seven- or ten-digit dialing indicated by the "subscriber" numbering plan type.
   - Calling number (prefix 91) for 11-digit dialing indicated by the "national" number plan type.
   - Calling number (prefix 9011) for international dialing indicated by the "international" numbering plan type.

- Calling number (prefix 91) to the "unknown" numbering plan type. If most calls are received from international locations, or local seven- or ten-digit callers, change the unknown field to match the highest percentage of inbound call sources.

This step is optional because the Cisco IP Phone user can use the Edit softkey and edit the phone number from a call history list and manually dial the required codes to properly route the call.

3. The Cisco IP Phone receives the call or the call is forwarded as a result of the application of the call-forwarding configuration.

External Phone Number Mask

The external phone number mask is a directory number (DN) configuration attribute. The external phone number mask is leveraged in call routing to manipulate the internal directory number to digits that can be routed over the PSTN. The external phone number mask is configured on the Directory Number configuration page in CUCM Administration. The use of the external phone number mask is enabled in the route list detail calling party number digit manipulations. The external phone number mask can also be leveraged at the route pattern, translation pattern, calling party transformation pattern, and hunt pilot configurations. Automated alternate routing (AAR) uses the external phone number mask to change the internal dial plan into a PSTN-routable dial plan when rerouting intersite calls from the WAN to the PSTN. The external phone number mask of the first DN of the phone is also used for the following functions:

- To change the display of the main phone number at the top of the LCD screen. A DN of 15001 with an external phone number mask of 21255XXXXX would result in a displayed phone number of 2125515001. Any user on the phone can instantly identify his PSTN direct inward dialing (DID) number by viewing the LCD of the phone.
- AAR technology uses the external phone number mask to manipulate digits for PSTN outbound dialing when bandwidth is not available for a guaranteed-quality call over the WAN (CAC). The AAR call will be rerouted out the PSTN using the full PSTN phone number of the destination as determined by the application of the external phone number mask.
- To change the display of the caller ID for all calls in which the call classification is Off-Net. The calling party number (caller ID) is changed to the full ten-digit DID phone number of the calling party.

Figure 11-4 displays the configuration of the external phone number mask at the Directory Number Configuration page. This page is accessed by navigating to the following in CUCM Administration:

Step 1. Choose **Device** > **Phone**.

Step 2. Insert the search criteria and click the **Find** button.

Step 3. Click the phone that has the required directory number (DN).

Step 4. Click the directory number.

Figure 11-5 displays the configuration option that is normally used at the route list detail level. The Calling Party Transformations section includes a check box to use the calling party's external phone number mask for the calling party presentation on the PSTN. This same option can be seen in various call-routing configuration elements.

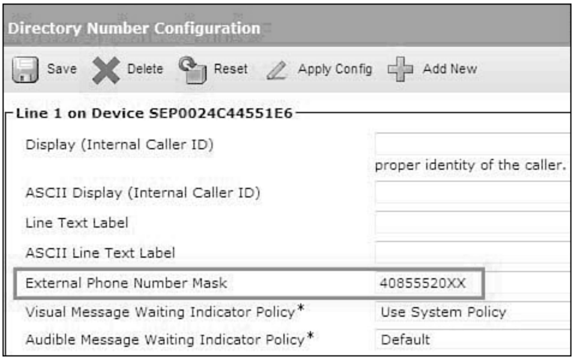

Figure 11-4 *Directory Number Configuration: External Phone Number Mask*

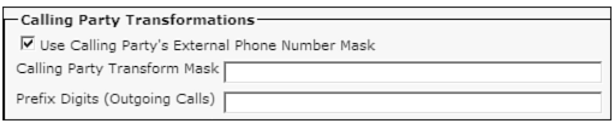

Figure 11-5 *Route Pattern Configuration: External Phone Number Mask*

Translation Patterns

CUCM uses translation patterns to manipulate digits before forwarding a call. A translation pattern normally requires another digit analysis attempt. Translation patterns and route patterns can be used to block patterns, but the default action is to attempt call routing.

Digit manipulation and translation patterns are used frequently in geographically distributed systems where office codes might not be the same for all locations. A uniform dialing plan can be created and translation patterns applied to accommodate the unique office codes at each location. Here are some additional examples where translation patterns can be leveraged:

- Security and operator desks (abbreviated dialing to PSTN locations enabling more productivity)
- Hotlines with a need for private line automatic ringdown (PLAR) functionality (security phones in elevators, phones used to access lab facilities, college campuses, financial trading markets, and so on)
- Extension mapping from a public to a private network

Translation patterns use route pattern style matching and transformation mask–based digit manipulation. The pattern resulting after the translation pattern is applied is then rerouted by the system, causing a second round of digit analysis. The new pattern can match another translation pattern where digit transformation can occur once again. Eventually, the call is routed to a device or blocked by CUCM. CUCM passes digits through translation patterns for only ten iterations to prevent call-routing loops. There are various call-routing loop-deterrent mechanisms that are in the system by default.

Figure 11-6 illustrates the operation of a translation pattern. A translation pattern matches the called party number in a similar manner to the matching of a route pattern. The primary difference between route patterns and translation patterns is that translation patterns do not have a final path selection destination (route list, gateway, or trunk). Translation patterns exist only to manipulate digits; they do not perform call routing.

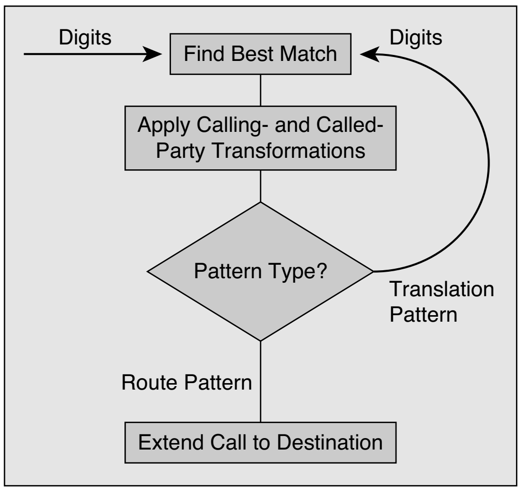

Figure 11-6 *Translation Patterns*

To configure a translation pattern, navigate to **Call Routing** > **Translation Pattern** in CUCM Administration.

Figure 11-7 is a screen capture of a translation pattern configuration. The translation pattern identifies the dialed digit string to match and the calling or called party transformation settings that should be applied.

If the Block This Pattern radio button is selected, a cause code must be selected. Choose a value from the drop-down menu:

- No Error
- Unallocated Number
- Call Rejected
- Number Changed

- Invalid Number Format
- Precedence Level Exceeded

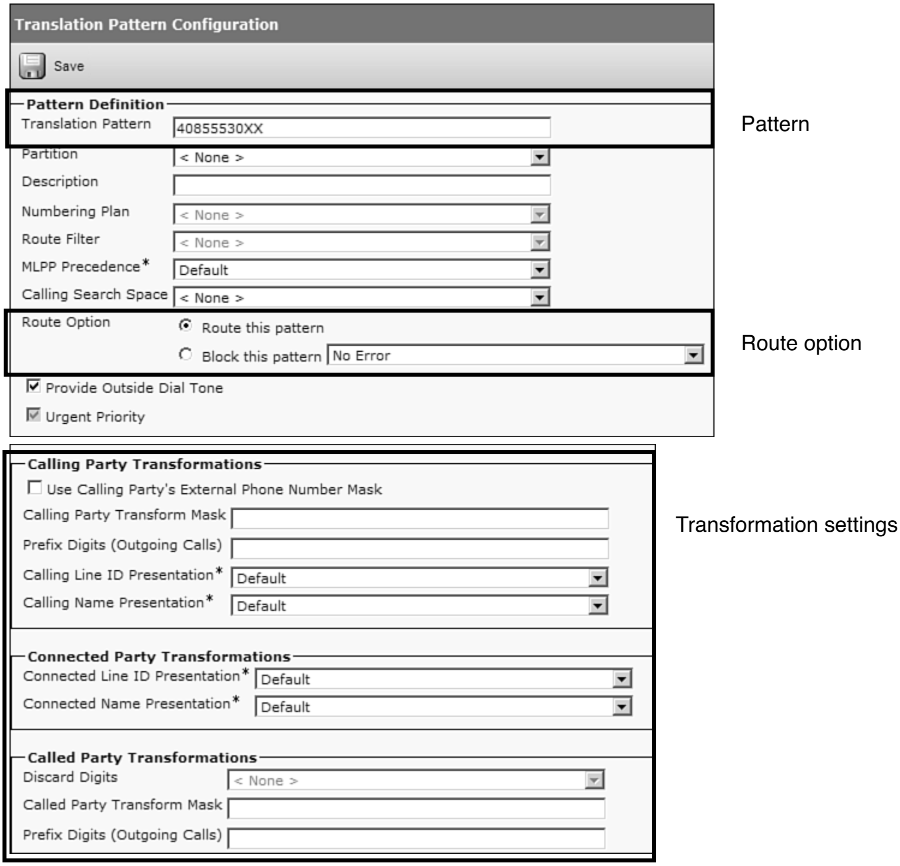

Figure 11-7 *Translation Pattern Configuration*

The transformation settings are not applicable if the Block This Pattern radio button is selected.

If the translation pattern contains an @ sign, a numbering plan and route filter can be selected to match certain number patterns of the selected numbering plan.

Translation patterns are processed as urgent priority by default. The Urgent Priority check box can be disabled beginning with CUCM 7.0. Prior versions of the product did not allow the urgent priority option to be disabled at the translation pattern configuration. An overlapping dial plan involving a translation pattern could result in call-routing issues. Translation patterns are ignored when performing analysis of the dial plan with the Dialed Number Analyzer (DNA) tool that is integrated into the Cisco Unified Serviceability web pages.

When the direct inward dialing (DID) range from the provider does not match the internal DN range, a translation pattern can be used to map the PSTN number to the internal DNs.

Figure 11-8 illustrates a scenario in which a company has a PSTN DID range of 408 555-1XXX, but the internal four-digit extensions use the four-digit range of 4XXX. The company uses a translation pattern that matches the assigned PSTN DID range of 408 555-1XXX. The called party transformation mask of 4XXX is applied to the translation pattern, resulting in a 4XXX called party number. CUCM applies the transformations and reanalyzes the resulting pattern. Eventually the call is routed to a device or explicitly rejected.

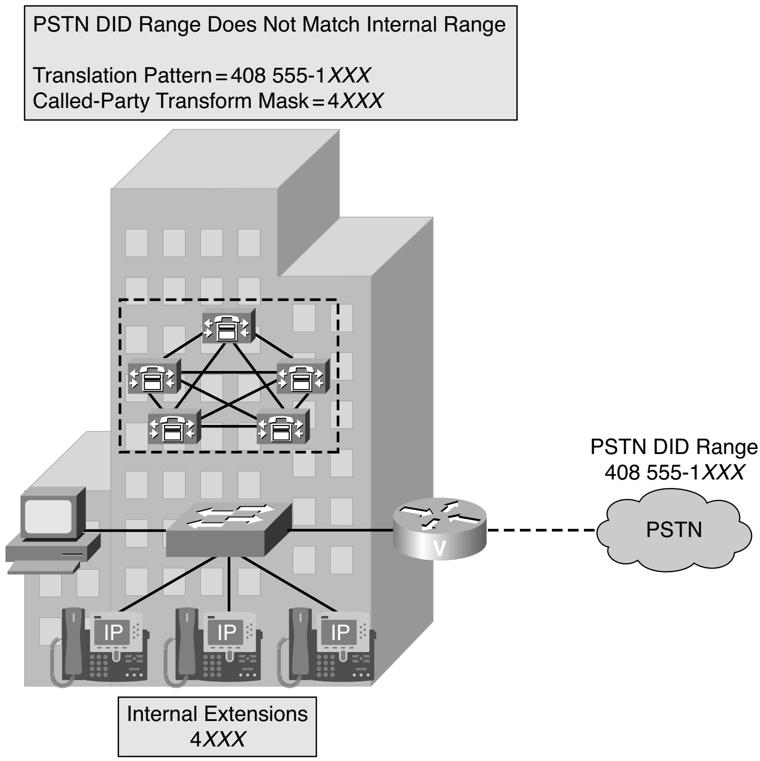

Figure 11-8 *Translation Pattern Example*

An additional translation pattern of XXXX with a called party transformation mask of 4111 can be used to route calls of unidentified internal extensions to the company operator. Many companies own large blocks of DID numbers that they are not currently using. Assume that the DN of 4333 no longer exists in the system because the person that had the phone number won the lottery and decided that he was not going to work anymore. Because of cost-cutting measures implemented, a replacement is not hired and the Cisco IP Phone is reused with a unique configuration for a different department. When a customer calls that user, the customer will receive a reorder tone unless a call forward unregistered (CFUR) number has been configured for the DN that receives the call. If a called party number of 408 555-1333 is received from the PSTN, the call will be routed to DN 4333. If a DN of 4333 no longer exists in CUCM, the generic XXXX translation pattern will be matched and the call is routed to the company operator at extension 4111. The company operator instructs the outside caller that the employee no longer works for the company and tries to assist the caller in resolving his issue.

Transformation Masks

Dialing transformations allow the call-routing component to modify either the calling (initiator) or called (destination) digits of a call. Transformations that modify the calling number (automatic number identification [ANI]) are calling party transformations; transformations that modify the called party (dialed digits) are called party transformations. Dialed Number Identification System (DNIS) is a public standard implemented in the PSTN for modifying called party numbers.

Digit translation is possible in CUCM mainly through the Transformation Mask feature that can be found in various configuration options in CUCM (for example, route list details and translation pattern). CUCM overlays the calling or called party number with the transformation mask so that the last character of the mask aligns with the last digit of the calling or called party number. CUCM uses the original calling or called party digit of the source number anytime the mask contains an X. The X acts as a binary OR function. If the number is longer than the mask, the mask will add extra digits to the original calling or called party pattern.

Figure 11-9 illustrates an approach typically used to change the calling party (ANI) of internal directory numbers when he or she makes calls that are routed to the PSTN. The five-digit extension of 45000 in Figure 11-9 is transformed into a ten-digit pattern for the purposes of caller ID (ANI) on the PSTN. There is a distinction between ANI and caller ID that I would like to point out. Caller ID (CLID) refers to the presentation of the calling party name and number, whereas automatic number identification (ANI) refers only to the calling number. The mask of 8086236XXX has been applied to 45000 in Figure 11-9, resulting in 45 being replaced with 36, while the first five digits of 80862 are prefixed before the number so that users connected to the PSTN can return phone calls to the presented calling party number.

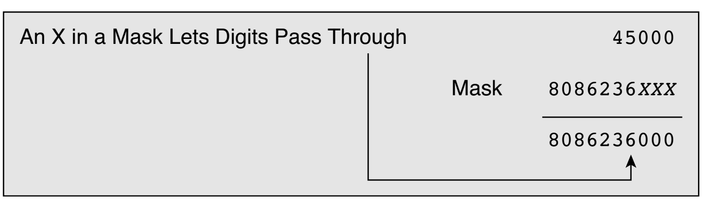

Figure 11-9 *Transformation Mask Operation*

Figure 11-10 illustrates the process in which a ten-digit number from the PSTN could be converted to a five-digit number using transformation masks. This process would be useful if the incoming called party from the PSTN gateway to CUCM was ten digits long, but incoming calls had to be converted to an abbreviated five-digit internal dial plan. Masks are always processed from right to left in CUCM. Transformation masks can be used to manipulate either the calling or called party number. A ten-digit pattern with a five-digit mask applied to it will result in a five-digit number. Figure 11-10 illustrates a ten-digit pattern with a five-digit pattern of 45XXX, which indicates that the last three digits will not change but the leading two digits will be set to 45, regardless of the incoming pattern.

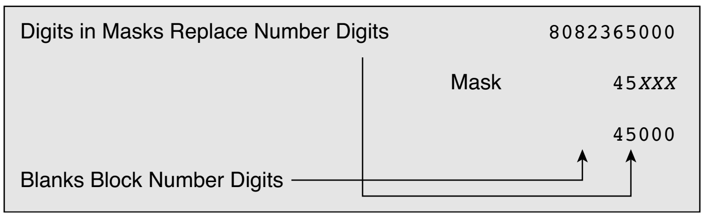

Figure 11-10 *Transformation Mask Operation*

Transformation masks are configurable at various CUCM configuration levels including route patterns, translation patterns, and route lists (per route group).

The calling and called party transformation settings are assigned to route groups in the route list details of the route list that the route pattern is pointed to. Route pattern transformations apply only when a route pattern is pointed directly to a gateway. Route patterns are normally pointed to a route list. Multiple route patterns can point to the same route list, but multiple route patterns cannot point directly to the same gateway. Inserting gateways into route groups allows the gateways to be used for many different route patterns.

Most intersite calls in private companies are routed over WAN links as Voice over IP (VoIP) calls, but routed over PSTN links if the WAN is down or congested. Distributed Multi-Cluster Call Processing architectures require call routing to be configured for all intersite calls that cross cluster boundaries. Intercluster calls are routed over trunks in CUCM. H.225 trunks, SIP trunks, nongatekeeper-controlled intercluster trunks, and gatekeeper-controlled intercluster trunks are covered in more detail in *Implementing Cisco Unified Communications Manager, Part 2 (CIPT2) Foundation Learning Guide*.

The call routing between sites that belong to different CUCM clusters is normally configured to have a PSTN route group and an IP WAN route group. The IP WAN route group includes one or more intercluster trunks (ICT) or SIP trunks, while the PSTN route group contains one or more gateway endpoints (MGCP) or gateway devices (H.323/SIP) that connect the cluster to the PSTN. CUCM will forward the internal abbreviated dialing extension number if proper digit manipulation has not been configured. CUCM routes calls to a gateway in the PSTN route group. Proper digit manipulation requires that the calling pattern reflect a phone number that can be called back on the PSTN and that the dialed digits are properly routed.

CUCM Digit Prefix and Stripping

The Digit Prefix feature prepends digits to the beginning of a dialed number. Any digits that can be entered from a standard phone (0 through 9, *, and #) can be prepended to the calling or called party numbers. Digit prefixing is available for either the calling or called party number and can be configured at the route pattern, route list, or translation pattern configuration levels.

Figure 11-11 displays the calling and called party prefix configuration available at the route pattern, route list, and translation pattern configuration levels.

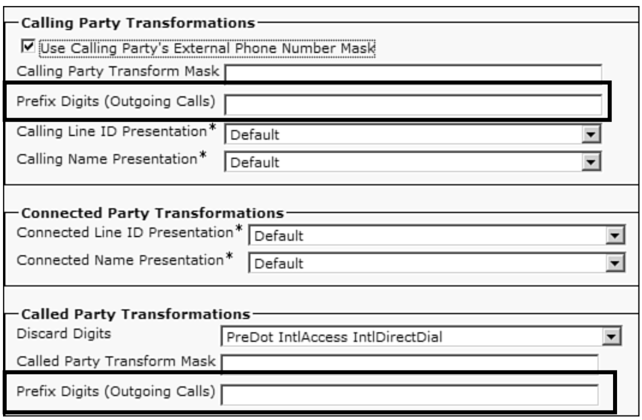

Figure 11-11 *Digit Manipulation: Prefix Digits*

Digit discard instructions (DDI) remove parts of the dialed digit string before passing the number on to the adjacent system. A DDI removes a certain portion of the dialed string (called party). Access codes are typically used to make a phone call that will be routed to the PSTN. The PSTN switch does not expect the access code, so the access code must be stripped out of the called party number before sending the call to the carrier.

Digit stripping is configured in the Called Party Transformations section by selecting a Discard Digits setting from the drop-down menu. Discard digits can be configured at the route pattern and at the route group details level of the route list.

The entire range of discard digits are supported if the @ wildcard pattern is used in the route pattern. If the @ wildcard is not used in the route pattern, only the <None>, NoDigits, PreDot, PreDot Trailing #, and Trailing # discard digits can be used.

Table 11-2 displays different digit discard instructions and their effects on dialed digits leveraging a route pattern of 9.5@. 9.5@ would not be used in most deployments, but it is a good example that can use various DDIs that are not available without the @ wildcard character. The digits that would be discarded appear in bold in Table 11-2.

The PreAt, 11D/10D@7D, 11D@10D, IntlTollBypass, and 10-10-Dialing complex DDIs are not available without the @ symbol in the route pattern.

Table 11-2 *Digit Discard Instructions 9.5@*

Instructions	Discarded Digits	Used For
PreDot	95 1 214 555 1212	Removes access code
PreAt	**95** 1 214 555 1212	Removes all digits that are in front of a valid numbering plan pattern
11D/10D@7D	**95 1 214** 555 1212	Removes PreDot/PreAt digits and local or long-distance area code
11D@10D	**95 1** 214 555 1212	Removes long-distance identifier
IntlTollBypass	**95 011 33** 1234 #	Removes international access (011) and country code
10-10-Dialing	**95 1010321** 1 214 555 1212	Removes carrier access (1010) and following carrier ID code
Trailing #	95 1010321 011 33 1234 #	Removes the # sign for PSTN compatibility

Note Trailing # is automatically removed by default in CUCM. You can turn off this behavior by changing the Strip # Sign from Called Party Number CUCM service parameter to False.

Figure 11-12 illustrates a call in which CUCM applies the PreDot DDI to the 9.8XXX route pattern, resulting in the access code (9) being stripped out of the dialed digits. The resulting four digits of 8123 are routed to the traditional PBX across a gateway or trunk device. The PBX analyzes the called party number and forwards the call to the necessary device. If the 8123 pattern did not match on a device in the PBX, it is very probable that the PBX would route the call back to CUCM, causing a call-routing loop. The PBX can have a route pattern–like configuration that routes all calls four digits in length beginning with an 8 (8XXX) to CUCM to accommodate phones that have been migrated to CUCM. CUCM probably has a route pattern of 8XXX to accommodate phones that have not been migrated from the PBX yet. If neither system has line 8123 configured on a device, a call-routing loop will normally occur. CUCM has service provider call-loop protection mechanisms that will only process each call reference value a certain number of times within a time interval. Supplementary service actions (call forward, conference, park, and so on) result in a new call reference value.

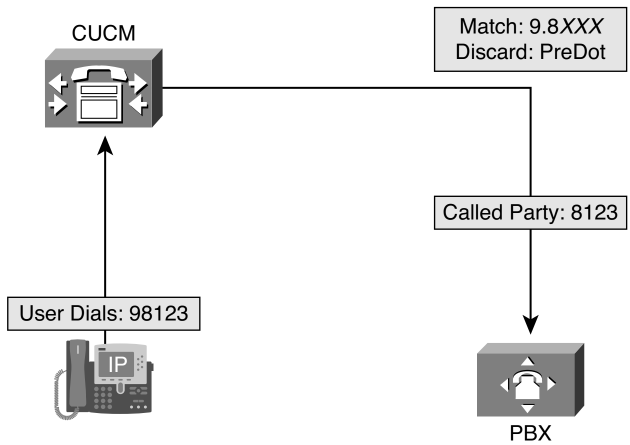

Figure 11-12 *PreDot Digit Discard Instructions*

Figure 11-13 illustrates the PreDot 10-10-Dialing DDI applied to the 9.@ route pattern. The PreDot 10-10-Dialing compound DDI strips the access code (9), the carrier selection code (1010), and the carrier identification code (288) from the called party number. The resulting 11-digit long-distance called party number of 1 214 555-1212 is then routed to the gateway device. Removing the 10-10 dialing parameters guarantees that long-distance calls will be billed by the preferred carrier. Most organizations contract a minimum number of long-distance minutes per month with the long-distance carrier. Although end users might believe that they are saving the company money by routing the call across an advertised carrier, they might be incurring additional costs to the organization. This compound DDI works only if the @ symbol is part of the route pattern. Translation patterns could perform similar functionality without introducing a route pattern with the @ symbol into the dial plan.

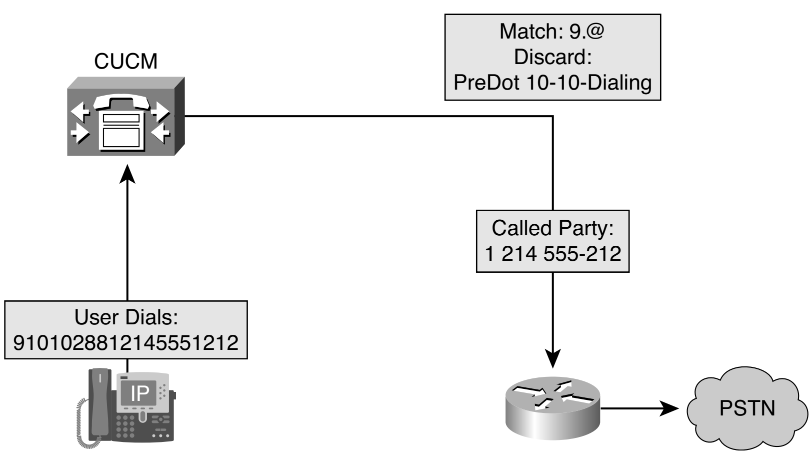

Figure 11-13 *Compound Digit Discard Instructions*

Significant Digits

The Significant Digits feature instructs CUCM to analyze the configured number of digits (from right to left) of the called number for incoming calls received by a gateway or trunk. Setting the significant digits to 5 on a PSTN gateway instructs CUCM to ignore all but the last five digits of the called party number for routing incoming gateway or trunk calls. The Significant Digits feature is the easiest approach to convert incoming PSTN called numbers to an internal extension, but the setting affects all calls received from the gateway. The Significant Digits setting also assumes that the internal dial plan is using the last five digits (or other number specified) of the DID block as the internal extension (directory number). The Significant Digits setting also cannot accommodate variable-length extension numbers on the internal network. Variable-length internal extensions could also lead to a variety of overlapping dial plan challenges.

The PSTN gateway illustrated in Figure 11-14 is using the Significant Digits setting in CUCM to instruct CUCM to only analyze the last four digits of the incoming call with a called party number of (408) 555-1010 received from the gateway. The significant digits configuration is available in the gateway or trunk CUCM Administration configuration pages under the Incoming Calls section (toward the bottom of the gateway/trunk web page).

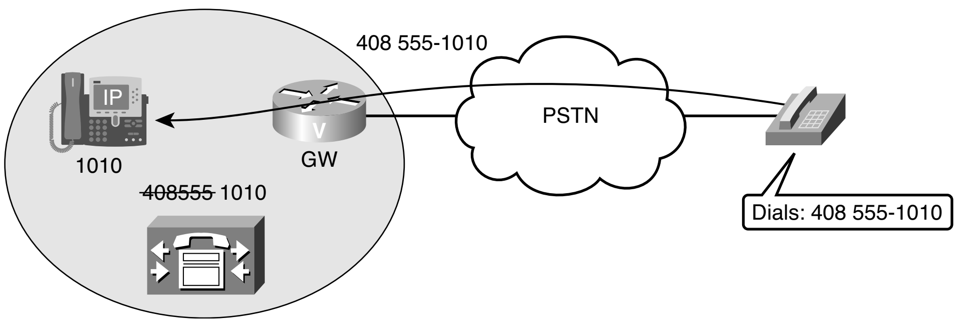

Figure 11-14 *Significant Digits Example*

> **Note** In contrast to using translation patterns to map E.164 numbers to internal DNs on incoming PSTN calls, this solution performs only one call-routing table lookup. The Significant Digits feature is a more processor-friendly alternative than translation patterns, but this approach will not allow the same flexibility as translation patterns.

Cisco Unified Communications Manager Global Transformations

CUCM version 7.0 introduced number normalization and number globalization support for E.164-based call routing. Calling and called party transformation patterns extend the power of CUCM's digit manipulation. Calling and called party transformation patterns have the following characteristics:

- Transformations are implemented in the global CUCM configuration.

- Calling and called party transformation patterns are put into partitions.
- Identical transformation patterns with different transformation settings can exit if they are put into different partitions. Partitions separate dial plan elements so that each pattern will only be evaluated if that partition is in the calling party's Calling Search Space (CSS).
- Gateways and trunks can be configured with calling and called party transformation CSSs. Calling party transformations are supported at the Cisco IP Phone, but called party transformations are not supported on the Cisco IP Phone.
- The transformation CSS determines which transformation patterns are visible to the device.

Calling and called party transformation patterns are applicable only to calls from CUCM to gateways, trunks, and phones. A call to a phone is usually not considered to be an outgoing call from a user's perspective. Think of a phone as the outgoing call leg of an internal call from another phone or incoming call.

Instead of configuring an individual calling and called party transformation CSS at each device, you can configure the devices to use calling and called party transformation CSSs configured at the device pool level. No transformation is performed if the device and associated device pool are not configured with a transformation CSS.

Calling and called party transformations are not applicable to calls that CUCM receives from devices (incoming call legs). Figure 11-15 illustrates called party transformations for four different phone numbers.

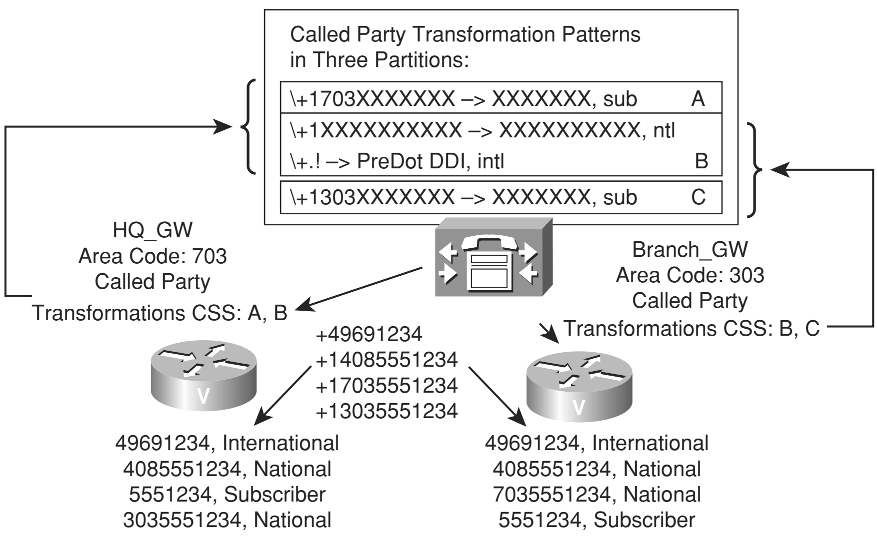

Figure 11-15 *Called Party Transformation Patterns*

Calling and called party globalized call routing has been configured in Figure 11-15, as indicated by the leading + character shown in the following four called party number strings:

+49691234

+14085551234

+17035551234

+13035551234

Transformations patterns only apply to outgoing call legs. Figure 11-15 is an example of globalized outbound call routing. Only the localization of the called number at the selected outgoing gateway is considered in this example.

Figure 11-15 is an example with four called party transformation patterns in three partitions at headquarters (HQ_GW) and branch (Branch_GW) sites. Partition A is specific to HQ (local area code 703), while partition B includes generic transformation patterns used by both HQ and Branch. Partition C is specific to the Branch site (local area code 303). The HQ gateway is configured with a called party transformation CSS that includes partitions A and B. The Branch gateway is configured with a called party transformation CSS that includes partitions B and C.

The transformation pattern in partition A modifies all 11 called party number information into a seven-digit called party number. The pattern also configures the numbering plan type to subscriber. Ten- and 11-digit dialing is normally categorized with a numbering plan type of national. Some providers require the numbering plan type to be set to the proper numbering plan type or they will reject the call. The transformation pattern in partition C provides the same function for called party numbers that are within the Branch area code of 303. Partition B is a partition that is shared between both the HQ and Branch transformation CSSs. Partition B includes two transformation patterns:

\+1XXXXXXXXXX

\+.!

The first pattern matches on all 11-digit patterns beginning with the E.164 + character used to route international calls followed by a 1 and any ten digits. This pattern represents all U.S. area codes within a globalized route plan. The second pattern represents all other possible numbers that begin with the + character followed by two digits or more.

Calls to the following four called party numbers are transformed differently depending on the gateway to which they are routed:

- +49691234 is matched and transformed on both gateways to 49691234 with a numbering plan type set to international. If the ISDN provider does not support number types, a prefix of 011 must be used to indicate the fact that this is an international call.
- +14085551234 is matched and transformed on both gateways to 4085551234, with type national. If the ISDN provider does not support number types, a prefix of 011 must be used.

- +17035551234 is matched and transformed on the both gateways, but the outbound calls match on different transformation patterns because of the different CSSs used at the respective gateways. The +17035551234 called party number is routed out the HQ gateway as 5551234 with a numbering plan type of subscriber. The Branch gateway matches the \+1XXXXXXXXXX with a number plan type of national. If the ISDN provider does not support numbering plan types for international call routing, a prefix of 011 must be used to route an international call.
- +13035551234 is matched and transformed on the Branch gateway with the \+1303XXXXXXX transformation pattern. The called party number is sent out the HQ gateway with a called party number of 5551234 and a numbering plan type of subscriber. The called party number is sent out the Branch gateway as 303 5551234 and a number plan type of national. If the ISDN provider does not support number types, a prefix of 011 must be used.

Figure 11-16 shows an example of calling party number transformation using calling party transformation patterns in different partitions. The HQ and Branch gateways and phones are configured with different calling party transformation CSSs to change the calling number differently depending on which gateway processes the call. Only the localization of the calling party number at the HQ outgoing gateway is considered in this example.

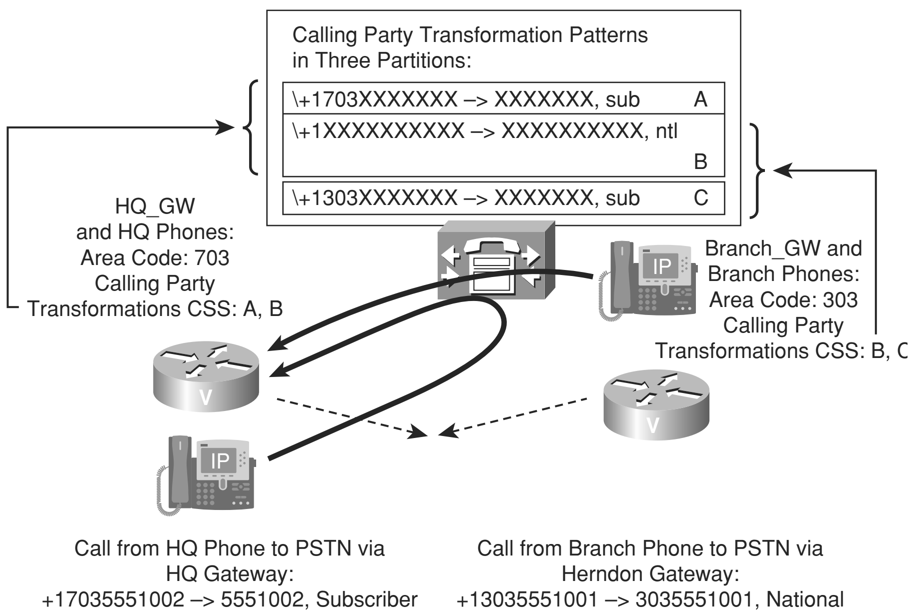

Figure 11-16 *Calling Party Transformation Patterns in Partitions*

There are three calling party transformation patterns in three different partitions. Partition A is specific to HQ (local area code 703), while partition B includes a generic transformation pattern for all 11 digit numbers in the North American Numbering Plan (NANP). Partition C is specific to the Branch (local area code 303).

The HQ gateway phones are configured with a calling party transformation CSS that includes partitions A and B, while the Branch gateway and phones have a calling party transformation CSS that includes partitions B and C. The transformation pattern in partition A modifies all HQ globalized numbers to a seven-digit number with a numbering plan type of subscriber. The transformation pattern in partition C provides the same functionality for local calls at the Branch site. Partition B is used by both the HQ and Branch transformation CSSs. Partition B includes the transformation pattern of \+1XXXXXXXXXX and represents all area codes in the NANP.

The calling party numbers will be transformed as follows:

- A +17035551002 call from an HQ phone to the PSTN through the HQ gateway is transformed to 5551002 with a numbering plan type of subscriber.
- A +13035551001 call from a Branch phone to the PSTN through the HQ gateway is transformed to 3035551001 with a numbering plan type of national.

Calling Party Transformation Pattern Configuration

Calling party transformation patterns are configured in CUCM Administration. Choose **Call Routing > Transformation > Transformation Pattern > Calling Party Transformation Pattern**. Click the **Add New** button to create a new calling party transformation pattern.

In the pattern configuration, define a matching pattern and assign a partition to this pattern. Specify calling party transformations in the same way as the route pattern, route list, and translation pattern configurations covered earlier in this chapter. Figure 11-17 is a screen capture of the Calling Party Transformation Pattern Configuration page in CUCM Administration.

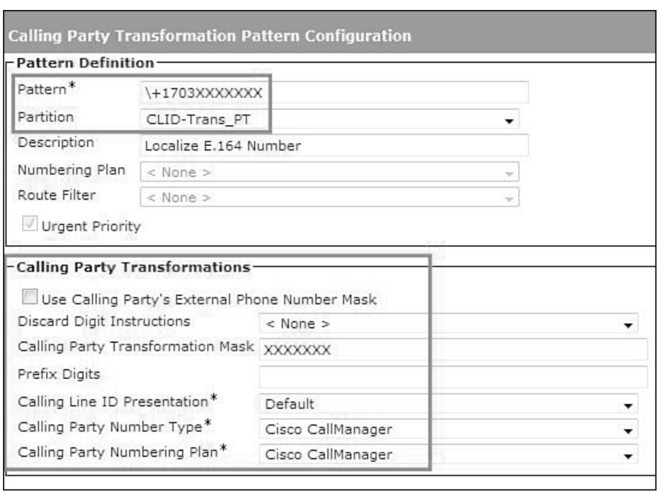

Figure 11-17 *Calling Party Transformation Pattern Configuration*

Called Party Transformation Pattern Configuration

Called party transformation patterns are configured in CUCM Administration. Choose **Call Routing > Transformation > Transformation Pattern > Called Party Transformation Pattern**. Click **Add New** to create a new called party transformation pattern. Figure 11-18 is a screen capture of a Called Party Transformation Pattern Configuration page.

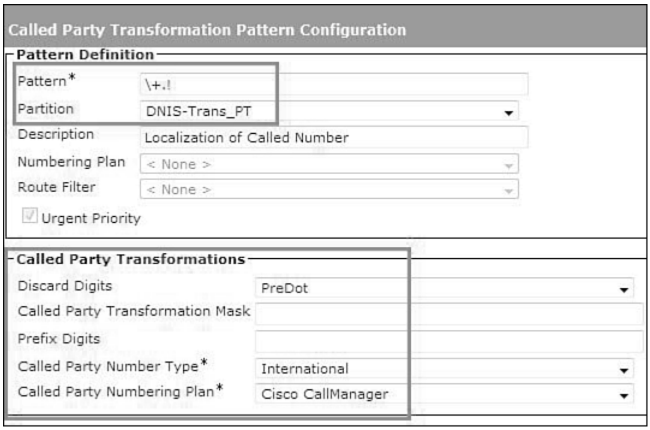

Figure 11-18 *Called Party Transformation Pattern Configuration*

Transformation Calling Search Space

The transformation Calling Search Space (CSS) configuration is identical to the CSS configuration used to configure class of service (CoS) restrictions that was covered in the last chapter. The CSS is applied differently to restrict the patterns that are matched for the purpose of digit transformation. During digit analysis, CUCM treats transformation patterns similar to any other pattern in the call-routing database. Independent CSSs are normally created for the purpose of performing calling and called party digit transformation using transformation patterns. Calling and called party transformation CSSs can be applied in the phone, gateway, and device pool configuration locations of CUCM Administration.

Figure 11-19 is a screen capture of a CSS configuration that will be used as a transformation CSS. Transformation CSSs normally only have one partition.

Figure 11-20 illustrates the application of the CSS created in Figure 11-19 as a calling party transformation CSS on a Phone Configuration page in CUCM Administration.

Incoming Number Settings

Incoming transformation settings have the following characteristics:

- They allow the configuration of digit prefixes, digit stripping, and transformations to be applied to calling and called party numbers for calls inbound to the CUCM cluster. Different settings can be configured per number plan type (unknown, subscriber, national, and international) if this information is in the call signaling.

- Incoming calling and called party settings can be configured at the device, device pool, and/or global service parameter configuration levels in CUCM Administration.
- Incoming calling and called party setting apply to calls received from gateways and trunks. Incoming calling and called party settings are not applicable to calls that are received from phones. The external phone number mask of directory numbers is used to globalize the calling party number from Cisco IP Phones.

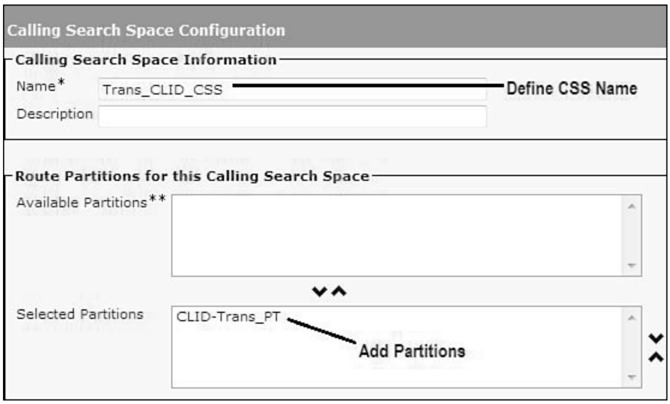

Figure 11-19 *Transformation CSS*

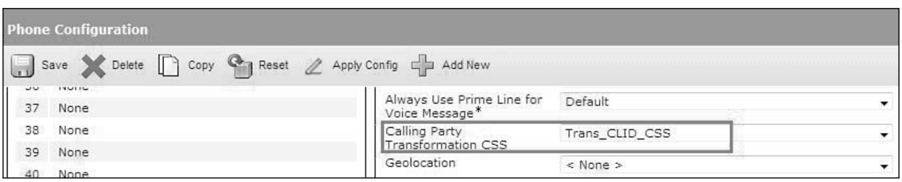

Figure 11-20 *Transformation CSS Application*

H.225 trunks and H.323 gateways support incoming calling and called party settings based on numbering plan type, but Media Gateway Control Protocol (MGCP) gateways support only incoming calling party settings based on numbering plan type. Session Initiation Protocol (SIP) does not support numbering plan types.

Incoming Calling Party Prefix Example: Globalization of Calling Number

Figure 11-21 shows an example of incoming calling party digit transformation for calling party number globalization using the E.164 + international operator pattern. Figure 11-22 is performing digit transformation based on the numbering plan type provided in the incoming call signaling from the provider in Hamburg, Germany.

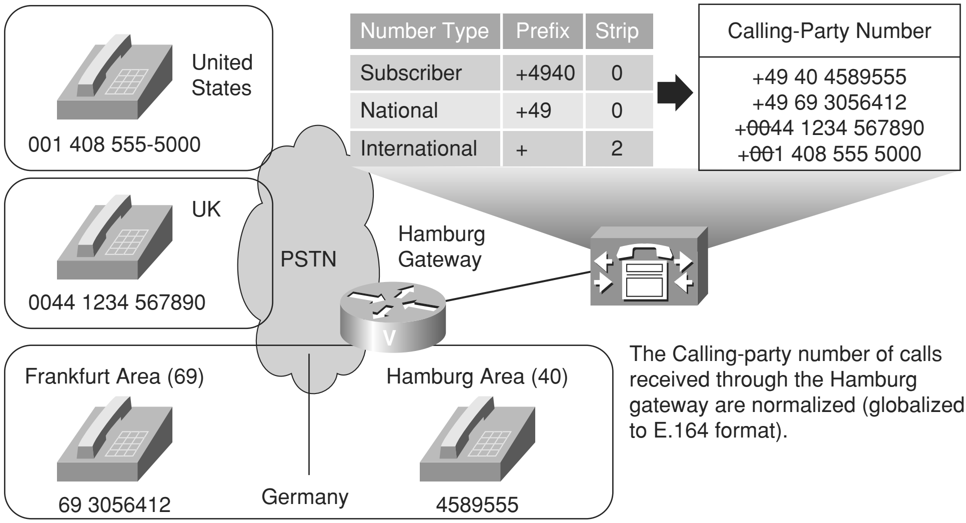

Figure 11-21 *Globalization of Calling Number*

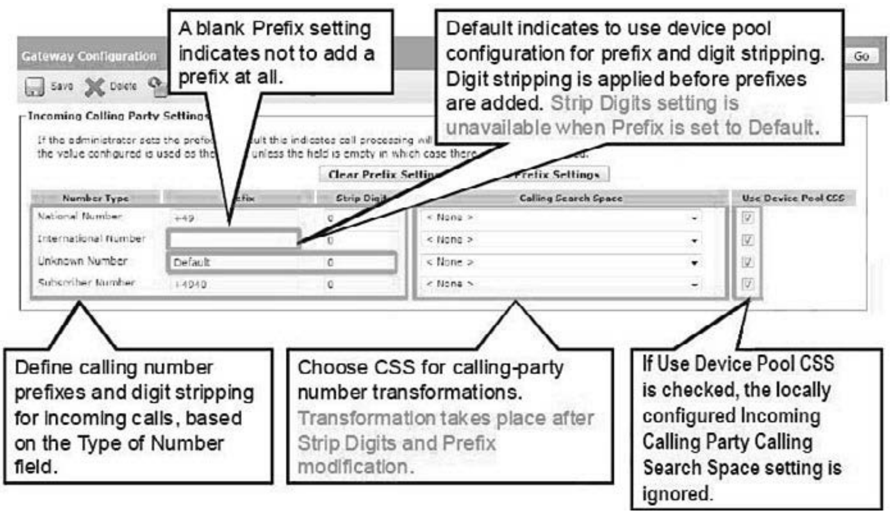

Figure 11-22 *Gateway Calling Party Settings*

Gateway Incoming Calling Party Settings Configuration

The gateway is configured with the following incoming calling party number digit manipulation:

- Prefix +4940 for calls that are received with a numbering plan type of subscriber.
- Prefix +49 for calls that are received with a numbering plan type of national.

- Prefix a + and strip the leading two digits of the calling party number for calls that are received with a numbering plan type of international.

Incoming calling party settings can be configured at the bottom of the gateway or trunk configuration level of CUCM Administration. Figure 11-23 is a screen capture of the configuration required to perform the digit transformation illustrated in Figure 11-22.

Device Pool Incoming Calling and Called Party Transformation Calling Search Space

Selecting the Use Device Pool CSS check box causes CUCM to ignore any transformation CSS configured at the gateway or trunk level. The transformation CSS defined at the device pool that is associated to the gateway or trunk is applied instead.

The configuration of incoming calling and called party settings in the device pool is nearly identical to the configuration of these settings on gateways or trunks.

The only differences are the following:

- The device pool does not include a Use Device Pool CSS check box.
- If the Default keyword is used in any Prefix field, the corresponding incoming calling or called party settings set at the Cisco CallManager service parameter configuration level are applied.

Transformation Examples

Multiple transformations can take place when placing a phone call. External phone number masks instructs the call routing of CUCM to apply the external phone number mask to the calling party directory number (DN) to pass caller ID information when calls are routed across a gateway to the PSTN. The external phone number mask is applied on an individual line basis through the DN configuration.

Figure 11-23 illustrates the multiple levels of calling party manipulations that are possible if the company wants to change the calling party number information so that a call appears to be coming from a main support number instead of an end user's extension (DN). The DN of 35062 will appear as 214 713-5062 when calls are routed through a gateway if only the external phone number mask is applied to the DN. The X character in the external phone number mask will pass through the original digits, while any digit specified in the mask will override the original number. If a mask applies more digits than the original number, a larger number will result. If the mask applies less digits than the original pattern, a smaller pattern will result. A calling party transformation mask has been applied at the route list detail level that changes the calling party number.

Figure 11-24 is an example of called party modifications where the user dials the pattern 10-10-321 before her phone number in an effort to save the company money on the phone call. The route pattern of 9.@ was matched by the dialed digits of 9 10-10-321 1 808 555-1221. The called party digit discard instruction (DDI) was configured to remove the 10-10 dialing. The resulting number is applied to the called party transformation mask, which consists of ten X wildcard characters. The access code of 9 and long-distance code of 1 have also been removed from the dialed digits. An 8 is prefixed as a new access code because the call can be routed to another system like a traditional PBX where an 8 is required as an access code to route the call to the PSTN.

Directory Number	35062
External Phone Number Mask	21471XXXXX
	2147135062
Calling-Party Transformation Mask	40885XX000
Caller ID	4088535000

Figure 11-23 *Calling Party Transformation Mask Example*

Dialed Number	9 10-10-321 1 808 555-1221
Discard Digits	10-10-Dialing
	9 1 808 555-1221
Called-Party Transformation Mask	XXXXXXXXXX
	808 555-1221
Prefix Digits	8
Called Number	8 808 555-1221

Figure 11-24 *Called Party Digit Manipulation*

Figure 11-25 is an example where the Cisco Unified Communications (UC) TAC support group in Richardson, Texas, is placing calls to Cisco TAC in San Jose, California. The corporate policy is to not allow direct calls to members of either support team. The calling and called party numbers will be manipulated to reflect the main hunt pilot used to distribute calls (call coverage) to support group members at each site:

1. User A at extension 5062 dials 91234.
2. The route pattern of 9.1XXX is matched against the dialed digits (called party).
3. A DDI of PreDot is applied to the called party. The resulting called party number is 1234.
4. A calling party transformation mask of X000 is applied to caller 5062.
5. The caller ID at the destination will now appear as if the call were placed from the hunt pilot of 5000 in Richardson, Texas.
6. A called party transformation mask of X000 is applied to the dialed digits. 1234 is applied to the mask, and the resulting number is 1000.
7. San Jose receives a call destined for extension 1000 with a caller ID of extension 5000.

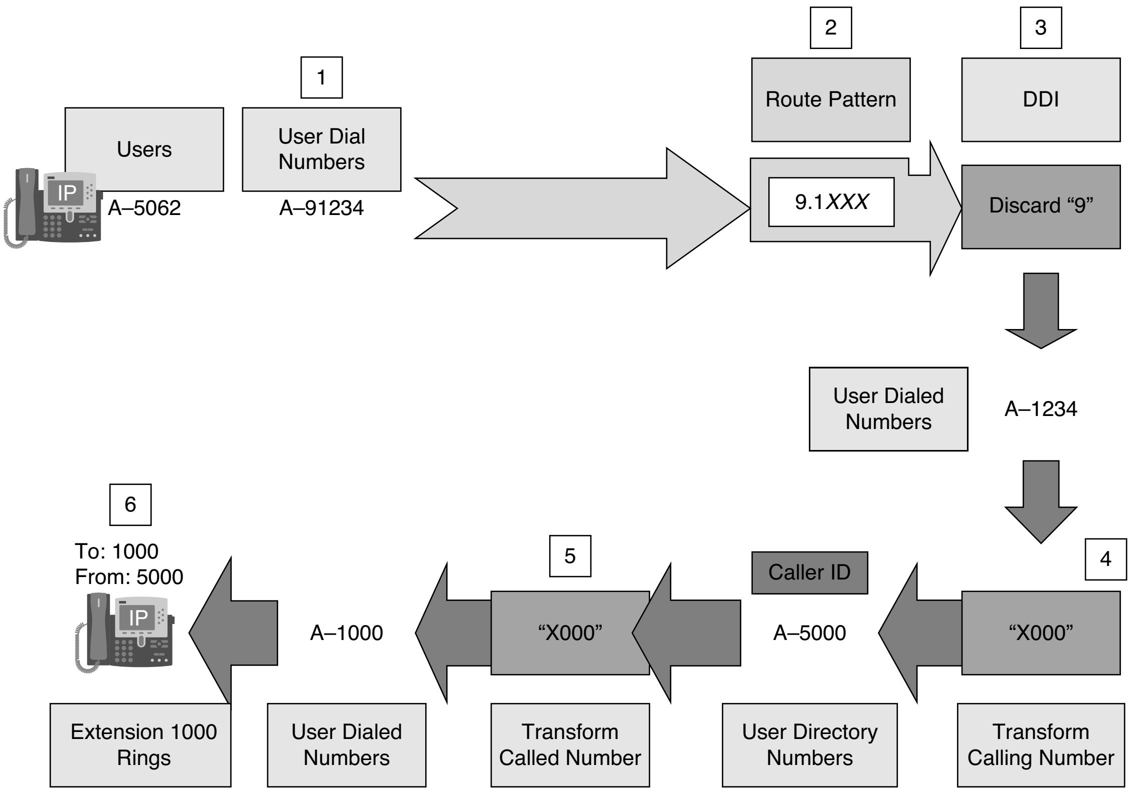

Figure 11-25 *Complex Digit Manipulation*

Three levels of digit-manipulation options are available for outbound calls:

- Digit manipulation that is configured on the route pattern (not used if the route pattern is routed to the route list)

- Digit manipulation that is configured at the route list detail level
- Digit manipulation that is configured by using a transformation CSS on the gateway/trunk or device pool

The three levels of digit manipulation are not cumulative. Only one level of digit manipulation will be applied. The hierarchy for these digit manipulations are as follows:

1. Digit manipulation settings on the route pattern take effect only when the route list details do not have any defined digit manipulations. A transformation CSS applied at the gateway/trunk or device pool will also cause the digit manipulations applied at the route pattern level to be skipped.
2. If the transformation CSS at the gateway or trunk matches, but the route list details have configured digit manipulations, the manipulations configured at the route list details are used. Route pattern digit manipulations are ignored.
3. If any manipulation matches through a gateway or trunk transformation CSS, all other digit manipulations are ignored.

Chapter Summary

The following list summarizes the key points that were discussed in this chapter:

- Digit manipulation is an essential dial plan function. It is mandatory to provide the correct called number to the PSTN and present appropriate calling party numbers on IP phones.
- Depending on the call flow, different methods and configuration elements can be used to manipulate calling and called party numbers.
- CUCM provides a variety of digit manipulation configuration elements, such as transformation masks, translation patterns, incoming calling party prefixes, and so on.
- CUCM external phone number masks can be used to display the full DID number on Cisco IP Phones. The external phone number masks also provide calling party modification for calls sent out to gateways or trunks.
- CUCM translation patterns provide powerful functionality to manipulate dialed digits and calling party numbers for any type of call.
- CUCM transformation masks are an integral part of digit manipulation at route patterns, translation patterns, and so on.
- CUCM digit stripping provides an easy way to apply DDI to route patterns or translation patterns.
- CUCM significant digits functionality allows simple called party number length normalization on incoming calls from gateways or trunks.
- CUCM global transformations provide a flexible and scalable way to implement globalization and normalization for functions such as globalized call routing.

- CUCM incoming number prefixes are used to modify incoming called and calling party numbers, based on their Type of Number setting.

Review Questions

Use the questions here to review what you learned in this chapter. The correct answers are found in Appendix A, "Answers to Review Questions."

1. The external phone number mask modifies which of the following for calls routed to the PSTN?
   a. ANI
   b. DNIS
   c. Caller ID name
   d. Route pattern
2. What dial plan element is used to manipulate digits when a route pattern can be routed to multiple devices?
   a. Route pattern
   b. Route list
   c. Route group
   d. Gateway
   e. Trunk
3. Which of the following items do external phone number mask configurations not have an effect upon?
   a. Automatic number identification
   b. Automatic alternate routing
   c. Extension mobility
4. Calling party modifications change which portion of a phone number?
   a. ANI
   b. DNIS
5. Called party modifications change which portion of a phone number?
   a. ANI
   b. DNIS
   c. RDNIS
   d. Original calling party

6. Which of the following items is processed as urgent priority by default?
   a. Directory numbers
   b. 911
   c. Route patterns
   d. Translation patterns
7. Which of the following patterns does the 10-10-Dialing digit discard instruction apply to?
   a. 9.!
   b. 9.[2–9]XXXXXX
   c. 9.@
   d. 9.1[2–9]XX[2–9]XXXXXX
8. Which of the following digit discard instructions can be applied to a route pattern of 9.1[2–9]XX[2–9]XXXXXX?
   a. 10-10-Dialing
   b. 11D@10D
   c. PreDot
   d. PreDot 11D@10D
9. A directory number of 11001 with an external phone number mask of 212551XXXX would result in what phone number?
   a. 11001
   b. 212 551-1001
   c. 212 551-100X
   d. 212 555-1001
10. A number of 212 555-1212 with a called party transformation mask of 646XXX3456 would result in which of the following numbers?
   a. 212 555-1212
   b. 646 555-1212
   c. 646 555-3456
   d. 212 646-1212

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