CVOICE

Implementing Cisco Voice Communications and QoS

Version 8.0

Lab Guide

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Lab Guide

Overview

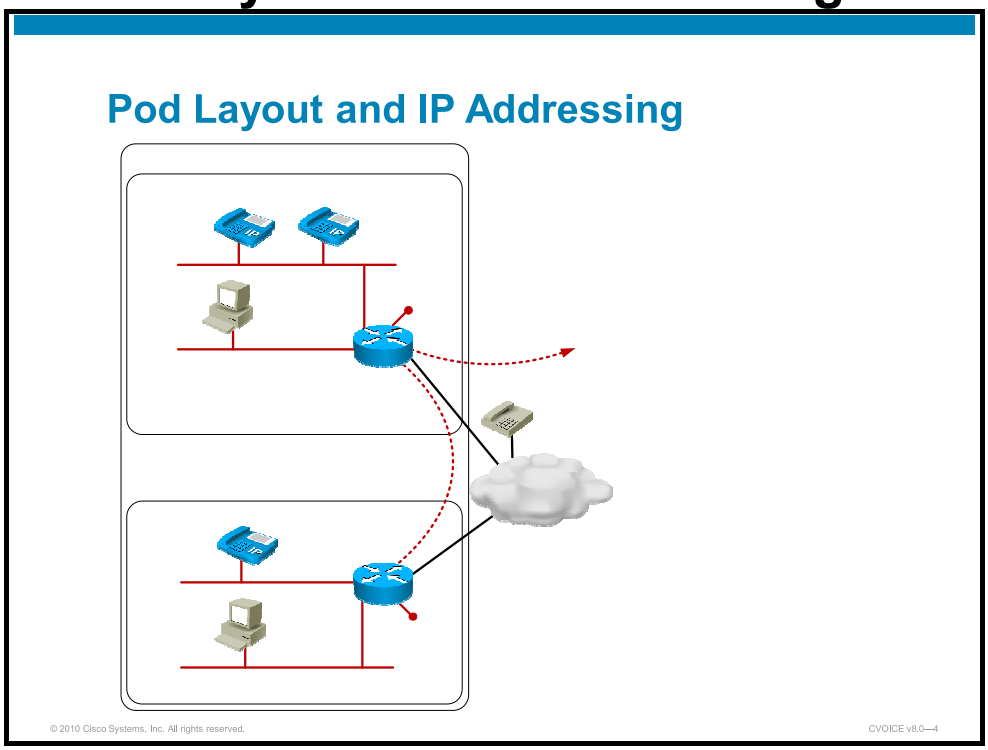
This guide presents the instructions and other information concerning the lab activities for this course. You can find the solutions in the lab activity Answer Key.

Outline

This guide includes these activities:

- Pullout Pod Layout and IP Addressing
- Description of Simulated PSTN
- Lab 1-1: Configuring Voice Ports
- Lab 1-2: Configuring DSPs
- Lab 2-1: Configuring VoIP Call Legs
- Lab 3-1: Configuring Cisco Unified Communications Manager Express to Support Endpoints
- Lab 4-1: Implementing Digit Manipulation
- Lab 4-2: Implementing Path Selection
- Lab 4-3: Implementing Calling Privileges
- Lab 5-1: Implementing Gatekeepers
- Lab 5-2: Implementing Cisco Unified Border Element
- Lab 6-1: Implementing QoS Using Cisco AutoQoS and Manual Configuration
- Answer Key

Pullout Pod Layout and IP Addressing



Description of Simulated PSTN

This section describes the functionality of the simulated PSTN.

The lab layout consists of two student pods: pod 1 and pod 2. Each pod consists of an HQ-x site and a BR-x site. These sites are virtually placed in the following different geographical regions:

- The HQ-x site is located in a virtual European country.
- The BR-x site is located in a virtual North American country.

This placement determines the dialing rules that are to be followed when dialing to the simulated PSTN. The HQ-x site follows European dialing rules. The BR-x site follows North American dialing rules (that is, the NANP).

Note There is no common European numbering plan in effect within the European Union at the moment. Therefore, for simplicity, the simulated PSTN supports dialing rules like the NANP. This will be explained in more detail in a later section.

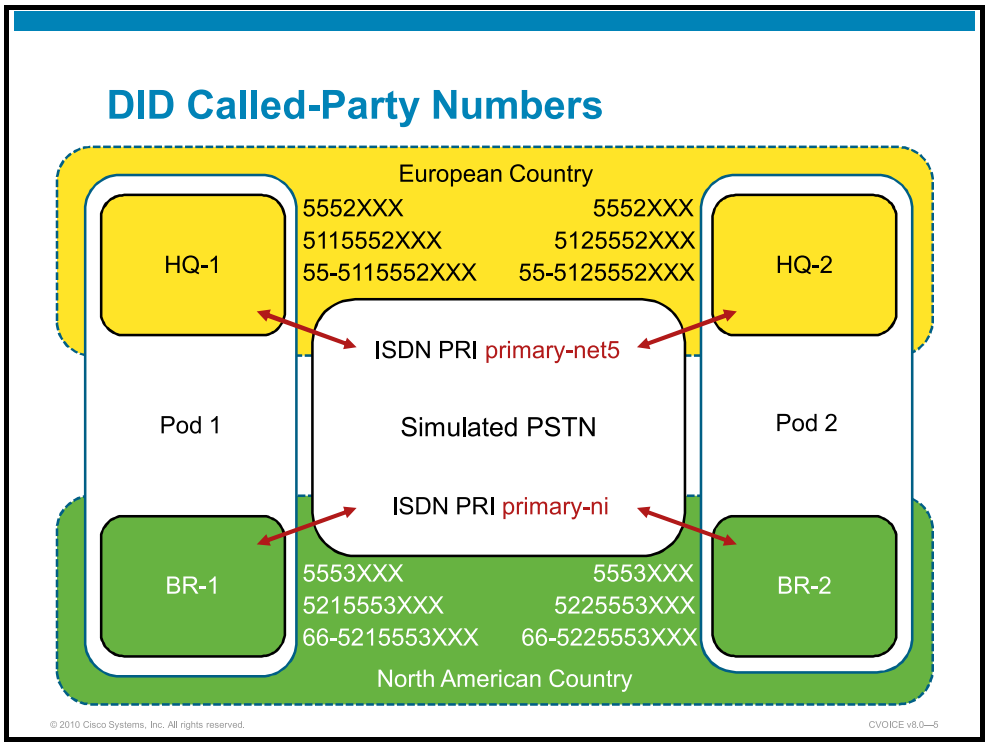
Student pods are connected to the simulated PSTN using ISDN PRI interfaces. ISDN PRI interfaces use one of two switch types toward the simulated PSTN.

- HQ-1 and HQ-2 use **switch-type primary-net5**.
- BR-1 and BR-2 use **switch-type primary-ni**.

Be aware that using **switch-type primary-ni** automatically modifies the type of number (TON) for outbound ISDN calls if they have one of the following called-party number formats:

- A seven-digit called-party number gets TON **subscriber.**
- A 10-digit called-party number gets TON **national.**
- A 12-digit called-party number (includes 2-digit country code) prefixed with 011 gets TON **international.**

This figure shows DID called-party number ranges that are used to dial into individual sites. These calls can be placed either from a simulated PSTN phone to a site or from one site to another via the simulated PSTN. Pods use overlapping directory numbers at the HQ-x and BR-x sites.



The individual sites use the DID number ranges that are listed in this table. Area codes 511, 512, 521, and 522 are assigned to individual sites. Country codes 55 and 66 represent the virtual European (EU) and the virtual North American (NA) countries.

Site DID Number Ranges

	HQ-m Site (EU)	BR-m Site (NA)
Site internal directory numbers	2XXX	3XXX
Local PSTN DID range	555-2XXX	555-3XXX
National PSTN DID range	51**m**-555-2XXX	52**m**-555-3XXX
International PSTN DID range	55-51**m**-555-2XXX	66-52**m**-555-3XXX

Note **m** is your pod number.

Simulated PSTN Phone

Each of the student pods has been using its own simulated PSTN phone: PSTN phone 1 (for pod 1) and PSTN phone 2 (for pod 2).

The simulated PSTN phone is used to initiate simulated PSTN calls to sites of the pod to which the simulated PSTN phone has been associated, or the simulated PSTN phone is used to terminate simulated PSTN calls that originated from a student pod. For this purpose, the simulated PSTN phone has been using six lines that represent different types of calls.

This table outlines which line buttons are available at the simulated PSTN phone as well as their functions.

Lines at the Simulated PSTN Phone

Button Position	Button Name*	Function
1	Local	Terminates valid local PSTN calls
2	National	Terminates valid national PSTN calls
3	Interntl	Terminates valid international PSTN calls
4	800	Terminates valid toll-free PSTN calls
5	Premium	Terminates valid premium (900) PSTN calls
6	Emergency	Terminates emergency PSTN calls

*Button names as they appear on the simulated PSTN phone display (button labels)

The definition of valid calls for each type of call is explained in later sections. In these later sections, it will also be explained how the simulated PSTN phone can be used to initiate calls of various types.

Caution Placing calls between simulated PSTN phones of two pods is not supported.

Calls Inbound to Simulated PSTN

This section describes which calls are recognized by simulated PSTN in the inbound direction when dialed from student pods and how these calls are routed.

Calls Inbound from HQ Sites

HQ sites are virtually located in Europe and, therefore, all calls from HQ sites into the simulated PSTN follow European dialing rules. For simplicity, however, variable-length numbers are implemented for international calls only and not for local and national calls, as is common in many European countries.

A call from an HQ site, depending on its called number, is either terminated at the simulated PSTN phone or routed to another site (HQ or BR) of pod 1 or pod 2. The call can also be returned (hairpinned) to the site that it came from.

This table lists all the types of valid inbound calls to the simulated PSTN from HQ sites.

Valid Inbound PSTN Calls from HQ Sites

Valid Called Number	Valid TON	Call Routed To	Calling Number Presentation Format
112	unknown	Emergency line*	preserved
[2-9]XX-XXXX (7 digits)	unknown	Local line*	preserved
[2-9]XX-XXXX (7 digits)	subscriber	Local line*	preserved
0-[2-9]XX-[2-9]XX-XXXX (11 digits)	unknown	National line*	preserved
[2-9]XX-[2-9]XX-XXXX (10 digits)	national	National line*	preserved
00-any number of digits	unknown	Interntl line*	preserved
Any number of digits	international	Interntl line*	preserved
0-800-[2-9]XX-XXXX (11 digits)	unknown	800 line*	preserved
800-[2-9]XX-XXXX (11 digits)	national	800 line*	preserved

Valid Called Number	Valid TON	Call Routed To	Calling Number Presentation Format
0-900-[2-9]XX-XXXX (11 digits)	unknown	Premium line*	preserved
900-[2-9]XX-XXXX (11 digits)	national	Premium line*	preserved
555-2XXX (7 digits)	unknown	HQ-**m**	subscriber
555-2XXX (7 digits)	subscriber	HQ-**m**	subscriber
0-511-555-2XXX (11 digits)	unknown	HQ-1	national
511-555-2XXX (10 digits)	national	HQ-1	national
00-55-511-555-2XXX (14 digits)	unknown	HQ-1	national
55-511-555-2XXX (12 digits)	international	HQ-1	national
0-512-555-2XXX (11 digits)	unknown	HQ-2	national
512-555-2XXX (10 digits)	national	HQ-2	national
00-55-512-555-2XXX (14 digits)	unknown	HQ-2	national
55-512-555-2XXX (12 digits)	international	HQ-2	national
00-66-521-555-3XXX (14 digits)	unknown	BR-1	international
66-521-555-3XXX (12 digits)	international	BR-1	international
00-66-522-555-3XXX (14 digits)	unknown	BR-2	international
66-522-555-3XXX (12 digits)	international	BR-2	international

*Lines at the simulated PSTN phone that are associated with the call originating pod.

Note Where **m** is the pod number that originated the call.

Note All calls being routed to pod sites present the called-party number in the national format, regardless of the call type.

Calls Inbound from BR Sites

BR sites are virtually located in North America. The simulated PSTN follows the NANP number formats.

A call from a BR site, depending on its called number, is either terminated at the simulated PSTN phone or routed to another site (HQ or BR) of pod 1 or pod 2. The call can also be returned (hairpinned) to the site from which it came.

This table lists all the types of valid inbound calls to the simulated PSTN from BR sites.

Valid Inbound PSTN Calls from BR Sites

Valid Called Number	Valid TON	Call Routed To	Calling Number Presentation Format
911	unknown	Emergency line*	preserved
[2-9]XX-XXXX (7 digits)	unknown	Local line*	preserved
[2-9]XX-XXXX (7 digits)	subscriber	Local line*	preserved
1-[2-9]XX-[2-9]XX-XXXX (11 digits)	unknown	National line*	preserved
[2-9]XX-[2-9]XX-XXXX (10 digits)	national	National line*	preserved
011-any number of digits	unknown	Interntl line*	preserved
Any number of digits	international	Interntl line*	preserved
1-800-[2-9]XX-XXXX (11 digits)	unknown	800 line*	preserved
800-[2-9]XX-XXXX (11 digits)	national	800 line*	preserved
1-900-[2-9]XX-XXXX (11 digits)	unknown	Premium line*	preserved
900-[2-9]XX-XXXX (11 digits)	national	Premium line*	preserved
555-3XXX (7 digits)	unknown	BR-**m**	subscriber
555-3XXX (7 digits)	subscriber	BR-**m**	subscriber
1-521-555-3XXX (11 digits)	unknown	BR-1	national
521-555-3XXX (10 digits)	national	BR-1	national
011-66-521-555-3XXX (15 digits)	unknown	BR-1	national
66-521-555-3XXX (12 digits)	international	BR-1	national
1-522-555-3XXX (11 digits)	unknown	BR-2	national
522-555-3XXX (10 digits)	national	BR-2	national
011-66-522-555-3XXX (15 digits)	unknown	BR-2	national
66-522-555-3XXX (12 digits)	international	BR-2	national
011-55-511-555-2XXX (15 digits)	unknown	HQ-1	international
55-511-555-2XXX (12 digits)	international	HQ-1	international
011-55-512-555-2XXX (15 digits)	unknown	HQ-2	international
55-512-555-2XXX (12 digits)	international	HQ-2	international

*Lines at the simulated PSTN phone that are associated with the call originating pod

Where **m** is the pod number that originated the call

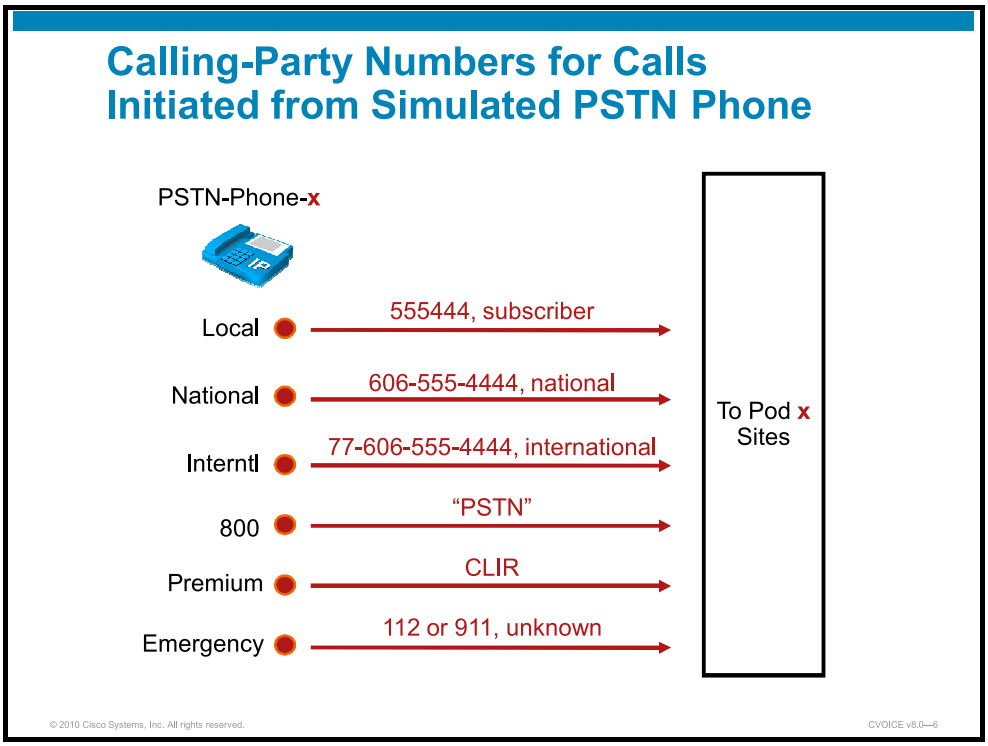
All calls being routed to pod sites present the called-party number in the national format, regardless of the call type.

Calls Outbound from Simulated PSTN

This section explains how outbound calls from the simulated PSTN toward student pods can be placed.

Predictable Calling-Party Numbers

For placing calls to student pods, each pod is using its own simulated PSTN phone. Calling-party numbers for the simulated PSTN phone-initiated calls are determined depending on which simulated PSTN phone line was selected to originate the call. This figure shows calling-party numbers and the associated TONs, based on a line selected. If a call is originated from the 800 (toll-free) line, instead of from the calling number, the calling-party name “PSTN” is presented (the calling-party number field is erased completely). If a call is originated from the Premium line, the presentation of the calling-party number is restricted.



Each line can place any of the valid supported call types. The line selection influences the calling-party presentation only. The call types and called-party number formats that are valid is explained in later sections.

If no line is selected before dialing at the simulated PSTN phone, and the called number is dialed immediately, then a call is placed as if it was dialed from the Local line.

Valid Outbound Calls from Simulated PSTN Phone

The calls that are placed from the simulated PSTN phone are routed to the HQ and BR sites of pod 1 or pod 2. Because HQ and BR sites are virtually located in different geographical regions, an outbound call must comply with the dialing rules in that particular region. The simulated PSTN phone covers both regions as if it was located in both Europe and North America at the same time.

This table lists all the call types and their accepted called-party numbers that are permitted to dial from the simulated PSTN phone.

Valid Outbound PSTN Calls

Accepted Called Number at Simulated PSTN Phone	Call Routed To	Called Number Presentation Format
555-2XXX (7 digits)	HQ-**m**	subscriber
555-3XXX (7 digits)	BR-**m**	subscriber
0-511-555-2XXX (11 digits)	HQ-1	national
0-512-555-2XXX (11 digits)	HQ-2	national
1-521-555-3XXX (11 digits)	BR-1	national
1-522-555-3XXX (11 digits)	BR-2	national
00-55-511-555-2XXX (14 digits)	HQ-1	international

Accepted Called Number at Simulated PSTN Phone	Call Routed To	Called Number Presentation Format
00-55-512-555-2XXX (14 digits)	HQ-2	international
00-66-521-555-3XXX (14 digits)	BR-1	international
00-66-522-555-3XXX (14 digits)	BR-2	international
011-55-511-555-2XXX (15 digits)	HQ-1	international
011-55-512-555-2XXX (15 digits)	HQ-2	international
011-66-521-555-3XXX (15 digits)	BR-1	international
011-66-522-555-3XXX (15 digits)	BR-2	International

Note Where **m** is the pod number that is associated with the simulated PSTN phone that is originating the call.

Lab 1-1: Configuring Voice Ports

Complete this lab activity to practice what you learned in the related module.

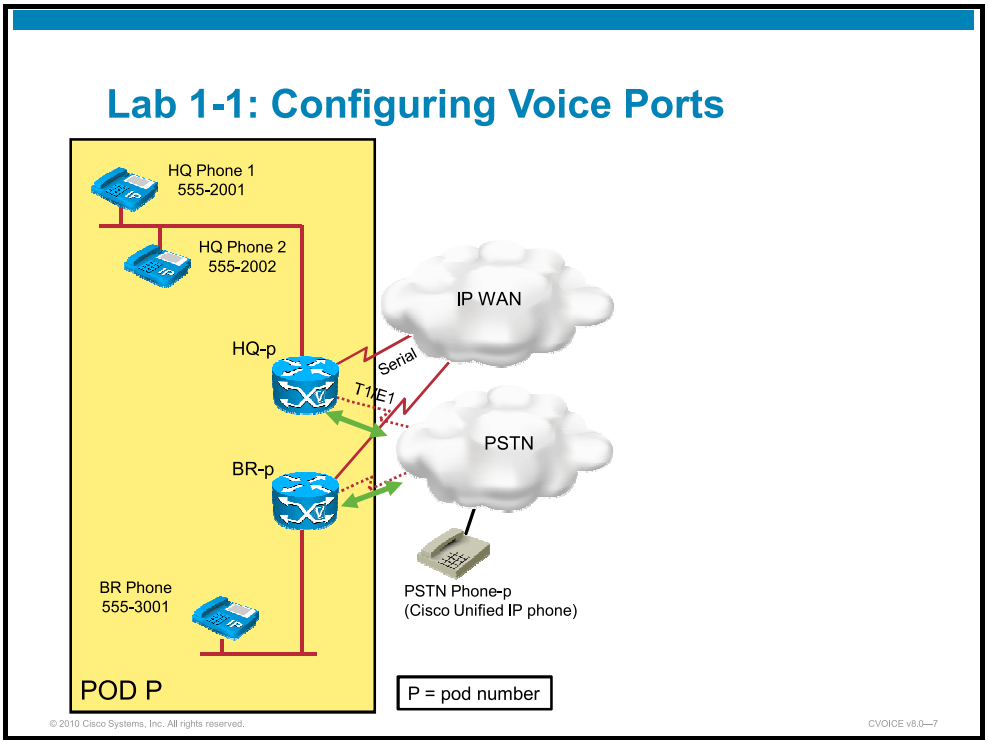
Activity Objective

In this activity, you will configure voice ports in order to connect a gateway to the PSTN using digital interfaces. After completing this activity, you will be able to meet these objectives:

- Configure a PRI interface for correct signaling, framing, and time slot association
- Configure the appropriate ports and dial peers to place calls to the PSTN
- Verify the call processing and dial peer matching

Visual Objective

The figure illustrates what you will accomplish in this activity.



Required Resources

These are the resources and equipment that are required to complete this activity:

- A PSTN phone
- At least one Cisco Unified IP phone in each site (HQ and BR)

Command List

The table describes the commands that are used in this activity.

Cisco IOS Commands

Command	Description
ip dhcp pool *pool-name*	Creates DHCP server pool and enters its configuration mode
network *address mask*	Configures network IP address and mask at DHCP server pool
default-router *IP-address*	Configures default gateway IP address at DHCP server pool
option 150 ip *IP-address*	Configures IP address of TFTP server at DHCP server pool
telephony-service	Activates Cisco Unified Communications Manager Express and enters its configuration mode
ip source-address *IP-address port*	Configures source IP address and port for Cisco Unified Communications Manager Express
max-dn *number*	Limits maximum number of directory numbers at Cisco Unified Communications Manager Express
max-ephones *number*	Limits maximum number of IP phones at Cisco Unified Communications Manager Express
auto assign *number* **to** *number*	Automatically assigns an already defined directory number to button 1 of autoregistering Cisco Unified IP phones
ephone-dn *dn-identifier*	Enters IP phone directory number configuration mode
number *telephone-extension*	Configures IP phone directory number
create cnf	Rebuilds configuration of IP phones at Cisco Unified Communications Manager Express
controller {t1 | e1} *slot/subslot/port*	Enters controller configuration mode
forward-digits {*num-digit* | **all** | **extra**}	Specifies which digits to forward for voice calls
framing {**sf** | **esf** | **crc4** | **no-crc4**}	Selects the frame type for the E1 or T1 controller. T1 options are sf and esf. E1 options are crc4 and no-crc4.
linecode {**ami** | **b8zs** | **hdb3**}	Selects the line code type for the T1 or E1 controller. T1 options are ami and b8zs. E1 options are ami and hdb3.
clear interface *slot/port*	Resets the specified ISDN interface
clock source {[**primary**] **line** | **internal** | **free-running**}	Sets the clocking for individual T1 or E1 links
isdn switch-type {*country-specific-switch-type*}	Defines the telephone company switch type
interface {**bri** | **pri**} *interface-number*	Enters interface configuration mode
isdn incoming-voice voice	Routes all incoming voice calls to the modem and determines how they will be treated
isdn protocol-emulate {**user** | **network**}	Defines Layer 2 and Layer 3 user- or network-side emulation

Command	Description
prefix *string*	Specifies the prefix of the dialed digits for a dial peer
pri-group timeslots *timeslot-range* [**nfas_d** {**backup** | **none** | **primary** {**nfas_int** *number* | **nfas_group** *number* | **rlm-group** *number*}} | **service**]	Specifies an ISDN PRI group on a channelized T1 or E1 controller
network-clock-participate {**aim** | **slot** | **wic**} *slot-number*	Specifies which clock source to use for DSP clocking
show call active voice	Displays parameters of active voice calls
show controllers e1 | t1	Displays information about the T1/E1 controllers
show isdn status	Displays the status of ISDN interfaces
show voice port [*slot/port:ds0-group* | **summary**]	Displays configuration information about a specific voice port or a summary of all voice ports
debug isdn q931	Examines the ISDN Layer 3 signaling messages
debug voice dialpeer	Examines the dial peer operations, including the matching of inbound and outbound dial peers

Job Aids

These job aids are available to help you complete the lab activity.

- Lab ISDN PRI requirements for both sites (HQ and BR):
  - Framing = cyclic redundancy check 4 (CRC4)
  - Line coding = high-density bipolar 3 (HDB3)
  - Clock source = line
  - PRI time slots 1 to 8
- Numbering plans:

Internal Numbering Plan

	Local HQ Site (EU)	Local BR Site (NA)
Internal numbering	555-2XXX	555-3XXX

Valid Numbers in Simulated PSTN

	Calls from HQ (EU) to PSTN	Calls from BR (NA) to PSTN
Local calls	**N**XX-XXXX (7 digits), TON: unknown **N**XX-XXXX (7 digits), TON: subscriber Example: **455-8000**	**N**XX-XXXX (7 digits), TON: unknown **N**XX-XXXX (7 digits), TON: subscriber Example: **455-8000**
National calls	0-**N**XX-**N**XX-XXXX, TON: unknown (0 + 3-digit area + 7 digits) **N**XX-**N**XX-XXXX, TON: national (3-digit area + 7 digits) Example: **0-455-455-8000**	1-**N**XX-**N**XX-XXXX, TON: unknown (1 + 3-digit area + 7 digits) **N**XX-**N**XX-XXXX, TON: national (3-digit area + 7 digits) Example: **1-455-455-8000**
International calls	00 + any number of digits, TON: unknown Any number of digits, TON: international Example: **00-23-455-455-8000**	011 + any number of digits, TON: unknown Any number of digits, TON: international Example: **011-23-455-455-8000**
Emergency calls	112, TON: unknown	911, TON: unknown

Note **N** represents a digit between 2 and 9.

Task 1: Configure DHCP Servers to Support Autoregistration

In this task, you will configure the DHCP server at the Cisco Unified Communications Manager Express that is running on your HQ and BR router to support the autoregistration of your IP phones.

Activity Procedure

Complete these steps:

Step 1 Configure the DHCP server on your HQ gateway to support HQ IP phones. Use the settings from the following table. Exclusion of IP addresses has been preconfigured.

DHCP Server Settings for IP Phones at HQ

Parameter	Value
DHCP pool name	HQ**p**-Phones
Network address	10.**p**.2.0
Network mask	255.255.255.0
Default router	10.**p**.2.101
Option 150	10.**p**.250.101

Note **p** is your pod number.

Step 2 Configure the DHCP server on your BR gateway to support BR IP phones. Use the settings from the following table. The exclusion of IP addresses has been preconfigured.

DHCP Server Settings for IP Phones at BR

Parameter	Value
DHCP pool name	BR**p**-Phones
Network address	10.**p**.4.0
Network mask	255.255.255.0
Default router	10.**p**.4.101
Option 150	10.**p**.250.102

Note **p** is your pod number.

Step 3 Connect your IP phones to the appropriate ports and observe the DHCP assigned IP settings.

Note You can verify the settings at the IP phone by pressing the **Settings** button and browsing through the menu to check the obtained IP parameters. The load file (found in the Model Information menu) should be based on SCCP for the phone to autoregister. If it uses a file that is based on Session Initiation Protocol (SIP), contact the instructor.

Activity Verification

You have completed this task when you attain this result:

- Your IP phones received the IP address from the DHCP server.

Task 2: Autoregister Cisco Unified IP Phones

In this task, you will autoregister Cisco Unified IP phones at the Cisco Unified Communications Manager Express that is running on your HQ and BR router.

Note This task involves only rudimentary settings that enable the verification of PSTN connectivity. The configuration of Cisco Unified Communications Manager Express is covered in a later exercise.

Activity Procedure

Complete these steps:

Step 1 On your HQ gateway, enable autoassignment of two directory numbers. Configure the Cisco Unified Communications Manager Express source address using the HQ loopback 0 interface. Allow five SCCP endpoints and 10 directory numbers.

Step 2 On your HQ gateway, configure two directory numbers with identifiers 1 and 2 and telephone extensions 5552001 and 5552002. The directory numbers should support two parallel calls.

Step 3 Rebuild the configuration files for IP phones at the HQ gateway.

Step 4 On your BR gateway, enable autoassignment of two directory numbers. Configure the Cisco Unified Communications Manager Express source address using the BR loopback 0 interface. Allow five SCCP endpoints and 10 directory numbers.

Step 5 On your BR gateway, configure one directory number with identifier 1 and telephone extension 5553001. The directory number should support two parallel calls.

Step 6 Rebuild the configuration files for IP phones at BR gateway.

Step 7 Ensure that the IP phones successfully register.

Activity Verification

You have completed this task when you attain these results:

- You verified that the IP phones registered and that they obtained a directory number.
- You verified that the two phones that are registered at HQ gateway could call each other.

Task 3: Configure PRI Interfaces

In this task, you will configure an ISDN PRI trunk for each site (HQ and BR) to the PSTN.

Activity Procedure

Complete these steps:

Step 1 On your HQ router, ensure that the digital signal processors (DSPs) are clocked correctly.

Step 2 On your HQ router, set the ISDN switch type to primary-net5 (the setting that is most common in Europe).

Step 3 Configure an ISDN PRI trunk on the E1 controller. Use time slots 1 through 8.

Step 4 Repeat this procedure to configure the ISDN PRI interface on the BR gateway, but set the ISDN switch type to primary-ni (the setting that is most common in North America).

Activity Verification

You have completed this task when you attain these results:

- The ISDN status on Layer 2 at the HQ and BR gateway was MULTIPLE_FRAME_ESTABLISHED.
- The operational state of the ISDN PRI voice port 0/0/0:15 was dormant (dorm), and the administrative state was up.

Task 4: Enable HQ to PSTN Calls

In this task, you will configure dial peers that enable outbound PSTN calls from the HQ site. The HQ is virtually located in Europe, thus it uses European dialing rules and PSTN access code 0.

Activity Procedure

Complete these steps on your HQ router:

Step 1 Create three POTS dial peers that all point to the voice port that corresponds to the ISDN PRI interface. Use these destination patterns:

- PSTN access code 0 followed by a seven-digit local number (NXXXXXX, where N is 2 to 9 and X is any digit). The called number that is sent to the PSTN must include seven digits without the PSTN access code. Use dial peer tag 7.

- PSTN access code 0 followed by the interarea code 0, the area code (three digits), and a seven-digit subscriber. The dial peer must match the 12-digit number (00NXX-XXXXXXX, where N is 2 to 9 and X is any digit). The called number that is sent to the PSTN must exclude the PSTN access code. Use dial peer tag 10.
- PSTN access code 0 followed by an international prefix 00 and a variable-length international number. The dial peer must match any number that starts with 000. The called number that is sent to the PSTN must exclude the PSTN access code. Use dial peer tag 9011.

Step 2 Create two POTS dial peers that point to the voice port corresponding to the ISDN PRI interface with these destination patterns:

- Emergency number 112 (dial peer tag 112)
- PSTN access code 0 followed by the emergency number 112 (dial peer tag 1120)

Step 3 From an HQ phone, call the numbers, using the configured dial peers. The goal is to make the PSTN phone ring at the appropriate line. Test the following dialing:

- Emergency 112
- Local call 4551000 (remember to add the PSTN access code 0)
- National call 4554551000 (remember to add the PSTN access code 0 and interarea code 0)
- International call 774554551000 (remember to add the PSTN access code 0 and international dialing prefix 00). You can avoid waiting for interdigit timeout by adding #.

Step 4 On your HQ router, debug ISDN Q.931 to inspect the called number that is sent to the PSTN in the previous step. If needed, correct the configuration of the dial peers. Verify that all calls go through.

Step 5 When placing calls, verify the dial peer matching using the **debug voice dialpeer** command, and examine active calls using the **show call active voice** command.

Step 6 With ISDN Q.931 debugging active, call an HQ phone from your PSTN phone, using the number 5552001. You should hear a second dial tone, and your HQ phone will not ring. Verify in the debugging output that the call arrived on the HQ gateway. In the space that is provided, write down the PSTN number from which the call originated (it will be needed in the next task). Why did the call not forward to the internal IP phone? You will fix the problem in the next task.

_____

Activity Verification

You have completed this task when you attain these results:

- You successfully placed outbound PSTN calls from your HQ phones.
- You monitored the call processing using the appropriate debug commands.
- You verified that inbound PSTN calls arrive on the HQ gateway but are not delivered to the destination (in other words, a phone does not ring).
- You examined the parameters of active calls using the **show call active voice** command.

Task 5: Enable PSTN to HQ Calls

In this task, you will enable inbound PSTN calls to your HQ site.

Activity Procedure

Complete these steps on your HQ gateway:

Step 1 Create a POTS dial peer for matching inbound PSTN calls, which has the following characteristics:

- It matches calls destined to any number.
- It uses direct inward dial.
- It has dial peer tag 1.

Step 2 From your PSTN phone, call an HQ phone (5552001 or 5552002), and verify that the call goes through and rings the appropriate phone.

Step 3 Debug the voice dial peer operations (**debug voice dialpeer** command), and make another call from your PSTN phone to an HQ phone. Which outbound dial peer is used for the call?

_____

Activity Verification

You have completed this task when you attain these results:

- You successfully placed inbound PSTN calls to your HQ phones.
- You monitored the call processing using the debug command, and you found out that the virtual dial peer that represents the IP phone directory number was used.

Task 6: Enable BR to PSTN Calls

In this task, you will configure dial peers that enable outbound PSTN calls from the BR site. The BR site is located in North America, thus it uses the NANP and PSTN access code 9.

Activity Procedure

Complete these steps on your BR gateway:

Step 1 Create three POTS dial peers, all pointing to the voice port that corresponds to the ISDN PRI interface. Use these destination patterns:

- PSTN access code 9 followed by a seven-digit local NANP number (NXXXXXX, where N is 2 to 9 and X is any digit). The called number that is sent to the PSTN must include seven digits without the PSTN access code. Use dial peer tag 7.
- PSTN access code 9 followed by a valid NANP long-distance number—the long distance prefix 1, the area code (three digits), and a seven-digit subscriber (1NXX-NXX-XXXX, where N is 2 to 9 and X is any digit). The called number that is sent to the PSTN must exclude the PSTN access code. Use dial peer tag 10.
- PSTN access code 9 followed by an international prefix 011 and a variable length international number. The called number that is sent to the PSTN must exclude the PSTN access code. Use dial peer tag 9011.

Note	Outbound calling from the BR site to the international PSTN number will not work until you reach the Lab 4-1 activity.

Step 2 Create two POTS dial peers that point to the voice port corresponding to the ISDN PRI interface with these destination patterns:

- Emergency number 911 (dial peer tag 911)
- PSTN access code 9 followed by the emergency number 911 (dial peer tag 9911)

Step 3 From the BR phone, place test calls to the numbers that are configured in the dial peers. The goal is to make the PSTN phone ring at the appropriate line. Remember to dial with the correct PSTN prefix.

Step 4 Use ISDN Q.931 debugging to inspect the called number that is sent to the PSTN. If needed, correct the configuration of dial peers. Verify that all calls go through.

Step 5 When placing calls, verify the dial peer matching using the **debug voice dialpeer** command, and examine active calls using the **show call active voice** command.

Activity Verification

You have completed this task when you attain these results:

- You successfully placed outbound PSTN calls from your BR phone.
- You monitored the call processing using the appropriate debug commands.
- You examined the parameters of active calls using the **show call active voice** command.

Task 7: Enable PSTN-to-BR Calls

In this task, you will enable inbound PSTN calls to your BR site.

Activity Procedure

Complete these steps on your BR gateway:

Step 1 Create a POTS dial peer for matching inbound PSTN calls, which has the following characteristics:

- It matches calls destined to any number.
- It uses direct inward dial.
- It has dial peer tag 1.

Step 2 From your PSTN phone, call the BR phone (5553001), and verify that the call succeeds.

Step 3 Verify call routing on the BR gateway using the appropriate debug commands.

Activity Verification

You have completed this task when you attain these results:

- You successfully placed inbound PSTN calls to your BR phone.
- You monitored the call processing using the appropriate debug commands.

Lab 1-2: Configuring DSPs

Complete this lab activity to practice what you learned in the related module.

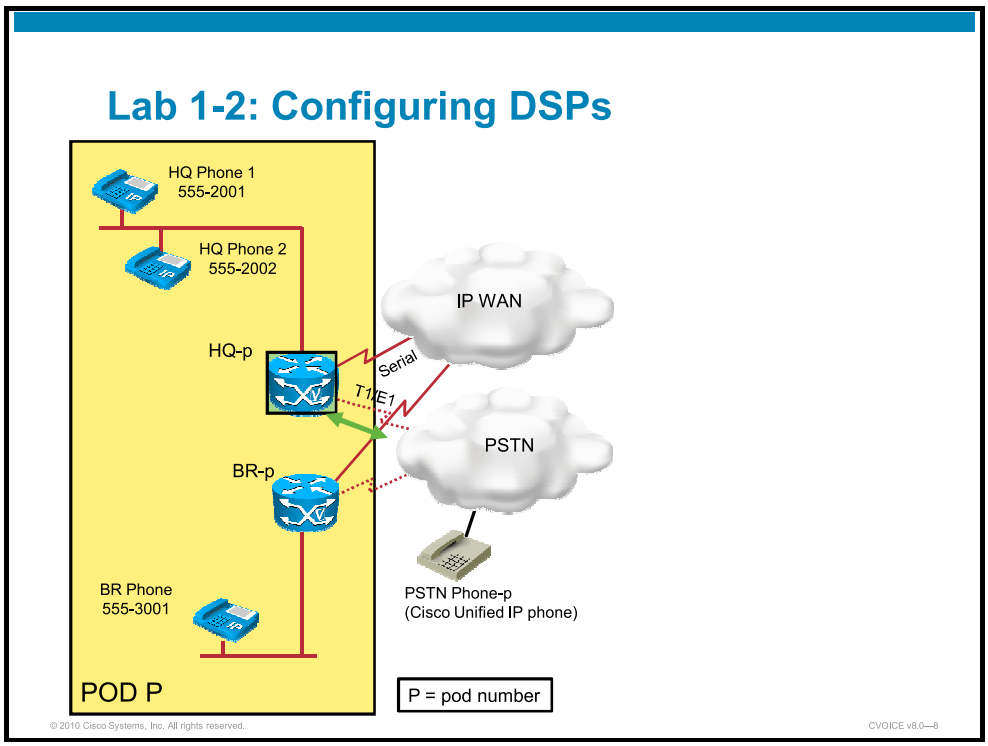
Activity Objective

In this activity, you will configure DSPs for voice termination and verify their operations. After completing this activity, you will be able to meet these objectives:

- Configure DSP resources to terminate voice calls
- Configure DSPs to support appropriate codec complexity
- Verify the use of DSP resources

Visual Objective

The figure illustrates what you will accomplish in this activity.



Required Resources

These are the resources and equipment that are required to complete this activity:

- A PSTN phone
- Two Cisco Unified IP phones in your HQ site

Command List

The table describes the commands that are used in this activity.

Cisco IOS Commands

Command	Description
voice-card *voice interface slot*	Enters voice card configuration mode
codec complexity flex | high | medium | secure	Specifies the maximum codec complexity that is supported by the voice card DSPs
codec *codec-name*	Specifies the codec that is used on the IP phone
show voice dsp	Displays the DSP resource usage and call parameters
show voice call status	Provides a compact view of active call legs and their parameters
show call active voice	Provides detailed information about active voice calls

Job Aids

These job aids are available to help you complete the lab activity.

Internal Numbering Plan

	Local HQ Site (EU)	Local BR Site (NA)
Internal numbering	555-2XXX	555-3XXX

Valid Numbers in Simulated PSTN

	Calls from HQ (EU) to PSTN	Calls from BR (NA) to PSTN
Local calls	**N**XX-XXXX (7 digits), TON: unknown **N**XX-XXXX (7 digits), TON: subscriber Example: **455-8000**	**N**XX-XXXX (7 digits), TON: unknown **N**XX-XXXX (7 digits), TON: subscriber Example: **455-8000**
National calls	0-**N**XX-**N**XX-XXXX, TON: unknown (0 + 3-digit area + 7 digits) **N**XX-**N**XX-XXXX, TON: national (3-digit area + 7 digits) Example: **0-455-455-8000**	1-**N**XX-**N**XX-XXXX, TON: unknown (1 + 3-digit area + 7 digits) **N**XX-**N**XX-XXXX, TON: national (3-digit area + 7 digits) Example: **1-455-455-8000**
International calls	00 + any number of digits, TON: unknown Any number of digits, TON: international Example: **00-23-455-455-8000**	011 + any number of digits, TON: unknown Any number of digits, TON: international Example: **011-23-455-455-8000**
Emergency calls	112, TON: unknown	911, TON: unknown

Note **N** represents a digit between 2 and 9.

These are gateway codec complexities:

- Low: G.711, clear channel
- Medium: G.729A, G.729AB, G.726, G.722, and Fax Relay
- High: G.723.1, G.728, G.729, G.729B, iLBC, Modem Relay

Task 1: Operate DSPs in Default Codec Complexity Mode Flex

In this task, you will test DSP operations in default codec complexity mode flex.

Activity Procedure

Complete these steps on your HQ gateway:

Step 1 View the automatically created ephones using the **show ephone** command. There should be two ephones with extension numbers that are associated with the ephones. View the preferred codec of each ephone. It should be "g711ulaw."

Note	The ephones are created as a result of the autoregistration when the endpoints first attempt to register. Autoregistration is enabled by default on Cisco Unified Communications Manager Express.

Step 2 From an HQ phone, call a PSTN local number (0-NXXXXXX, where N is 2 to 9 and X is any digit), and answer the call on the PSTN phone.

Step 3 Verify the codec that is used on the VoIP call leg between the HQ phone and the HQ gateway, using two methods:

- Use appropriate **show** commands on the HQ gateway.
- On the IP phone, press the **Settings** button and choose **Status > Call Statistics**.

Note	Both methods should show that the VoIP call leg uses G.711 with a 20-ms payload size.

Step 4 On the HQ router, change the preferred codec of the first ephone (5552001) to iLBC, and reset the phone using these commands:

```
ephone 1
 reset
```

Step 5 From the first HQ phone, call the PSTN phone (0-NXXXXXX), and check the codec that is used during the call. Display DSP usage using different options of the **show voice dsp** command, including the **detailed** keyword.

Step 6 From the PSTN phone, call the first HQ phone (555-2001), and check the codec that is used in the call. It should be the same as the codec that is used in the call that originated in HQ.

Step 7 On the HQ router, enter the voice-card 0 configuration mode, and attempt to change the DSP codec complexity to medium. This operation should fail, due to the existing voice ports.

Activity Verification

You have completed this task when you attain these results:

- You verified that the flex mode supported high-complexity codecs such as iLBC.
- You viewed the DSP usage for voice termination using different options of the **show voice dsp** command.

Task 2: Operate DSPs in Medium Codec Complexity Mode

In this task, you will configure the voice card to support low- and medium-complexity codecs, and you will verify the DSP operations.

Activity Procedure

Complete these steps on your HQ router:

Step 1 Perform this procedure to delete the voice port:

- Enter the voice port configuration mode (for example, **voice-port 0/0/0:15**), and shut down the port.
- Enter the controller configuration mode (for example, the **controller e1 0/0/0**), and shut down the controller.
- In the controller configuration mode, remove the PRI group using the **no pri-group** command.

Note	This will also remove the port command from all POTS dial peers.

Step 2 Enter the voice card configuration, and change the codec complexity to medium.

Step 3 Perform this procedure to re-create the voice port:

- Enter the controller configuration mode, and define the ISDN PRI group using the **pri-group timeslots 1-8** command.
- Activate the controller using the **no shutdown** command.
- Reapply the voice port to all dial peers (for example, add **port 0/0/0:15** command to the dial peers).

Step 4 From the first HQ phone (preferring iLBC), call a PSTN local number (0-NXXXXXX), and check the codec that is used in the call. The codec should be downgraded to a lower-bandwidth, medium-complexity codec. Write down which codec is used, in the space that is provided:

_____

Step 5 With one active conversation, call a PSTN national number (0-0-NXX-XXXXXXX) from the second HQ phone (preferring G.711). Check the codec that is used in the call. Display DSP usage, using various options of the **show voice dsp** command.

Step 6 Change the codec complexity on the HQ gateway back to flex mode. Remember to reapply the voice port to all POTS dial peers.

Activity Verification

You have completed this task when you attain these results:

- You verified that the iLBC codec was no longer supported for voice termination when the DSP resources were configured for medium codec complexity.
- You verified that the codec negotiation for voice termination resulted in a codec that did not exceed the permitted complexity and that it did not consume more bandwidth than the preferred one.

Lab 2-1: Configuring VoIP Call Legs

Complete this lab activity to practice what you learned in the related module.

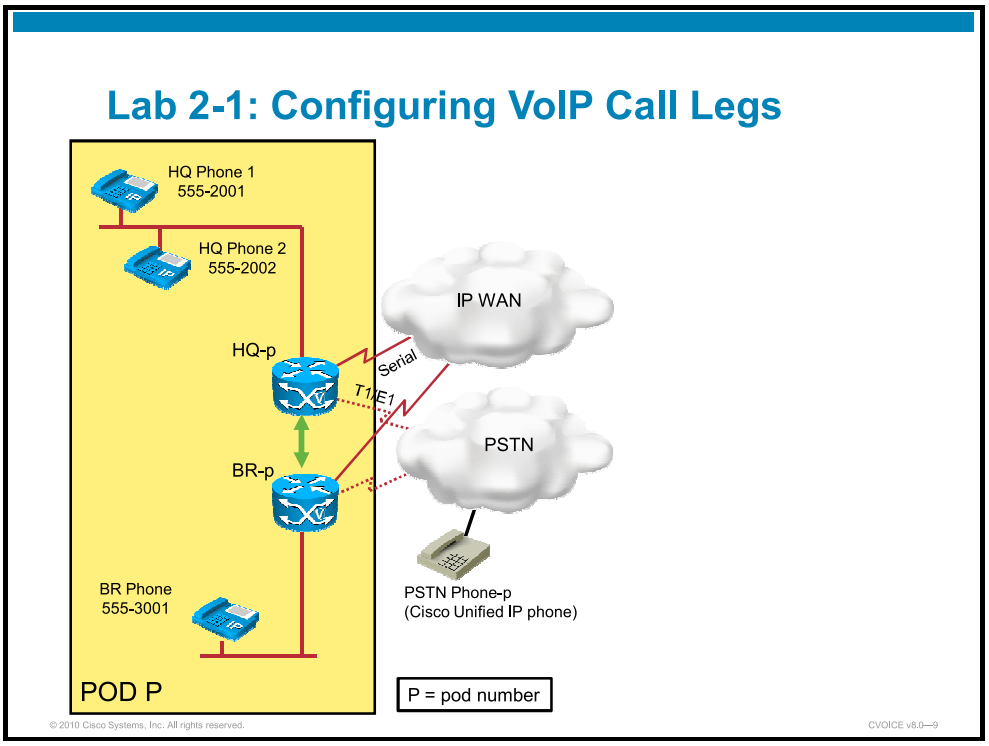
Activity Objective

In this activity, you will configure VoIP dial peers and tune gateway parameters to allow VoIP calls between your HQ and BR sites. After completing this activity, you will be able to meet these objectives:

- Configure basic VoIP dial peer parameters
- Configure codecs and codec classes, and verify codec negotiation
- Configure and verify H.323 fast start and slow start
- Configure the H.323 gateway interface bind feature

Visual Objective

The figure illustrates what you will accomplish in this activity.



Required Resources

These are the resources and equipment that are required to complete this activity:

- Two Cisco Unified IP phones in your HQ site
- One Cisco Unified IP phone in your BR site

Command List

The table describes the commands that are used in this activity.

Cisco IOS Commands

Command	Description
codec codec-name	Specifies which codec is to be used for calls matching this dial peer
codec preference 1-14 codec-name	Configures one entry in the codec list under the **voice class codec** command. Repeat this command as many times as you need, to specify codecs in this list.
dial-peer voice *tag* **voip**	Enters dial-peer configuration mode and specifies VoIP
h323-gateway voip bind srcaddr	Configures the H.323 gateway interface bind feature
session protocol sipv2	Configures the VoIP dial peer to use SIP signaling
session target ipv4:*x.x.x.x*	Specifies the destination IPv4 address for the gateway terminating a VoIP call
voice class codec *tag*	Enters voice class codec configuration mode. In dial peer mode, it attaches the voice class to the dial peer.
voice service voip	Enters voice service voip configuration mode
h323	Enters H.323 mode from voice service voip configuration mode
sip	Enters SIP mode from voice service voip configuration mode
call start slow | fast	Defines H.323 slow start or fast start in H.323 mode
bind control source-interface	Configures the interface bind feature for SIP signaling, in SIP mode
show call active voice	Displays information on active calls
show dial-peer voice (*tag*) **| (summary)**	Displays dial-peer configuration information
show dialplan number *number*	Displays which dial peers are matched when a particular telephone number is dialed
show voice call status	Displays the status of active voice calls
debug voice dialpeer	Monitors the dial peer matching process
debug voip ccapi inout	Displays real-time call control processing and call leg information
debug h225 event	Monitors H.225 events
debug h245 event	Monitors H.245 events
debug h245 asn1	Monitors H.245 ASN.1 library
debug ccsip message	Displays SIP signaling messages

Job Aids

These job aids are available to help you complete the lab activity.

Internal Numbering Plan

	Local HQ Site (EU)	Local BR Site (NA)
Internal numbering	555-2XXX	555-3XXX

Task 1: Configure Basic VoIP on the HQ Gateway

In this task, you will configure a basic VoIP dial peer for HQ-to-BR connectivity.

Activity Procedure

Complete this step on your HQ gateway:

Step 1 Create a VoIP dial peer with these parameters:

- It matches the entire 5553XXX range.
- It points to loopback 0 of your BR gateway (10.p.250.102, where p is your pod number).
- It disables voice activity detection (VAD).
- It has dial peer tag 3000.

Activity Verification

You have completed this task when you attain these results:

- On your BR gateway, enable dial peer debugging. Make sure that your Telnet connection has configured the terminal monitor.
- From your first HQ phone, call the BR phone (5553001) and answer the call. Examine the debugging output, and confirm that no inbound dial peer has been matched on the BR gateway.
- With the call still active, view the active calls using the **show voice call status** command to verify that the inbound dial peer is 0.
- With the call still active, display its VAD setting using the **show call active voice | inc VAD** command. Compare the VAD setting on both gateways. The setting should be different.
- With the call still active, use the appropriate **show** command to examine the codec that is used in the call. End the call.
- Call the BR phone (5553001) from your second HQ phone. Answer the call and examine the codec. You should see that both calls use G.729, despite the fact that the preferred codec of the first HQ phone is iLBC, and the preferred codec of the second HQ phone is G.711.
- View the default codec of the VoIP dial peer, using the appropriate commands. You may want to selectively display the parameter using the **show dial-peer voice 3000 | inc codec** command.

Task 2: Configure Codec on the HQ Gateway

In this task, you will configure a single codec in the VoIP dial peer on your HQ gateway.

Activity Procedure

Complete these steps on your HQ gateway:

Step 1 Set the VoIP dial peer (3000) codec to g723r53.

Step 2 From an HQ phone, call your BR phone (5553001) and answer the call. The call should disconnect as soon as it is answered. This behavior is typical for a codec mismatch.

Step 3 Use H.245 debugging to view the codec negotiation between the gateways. Has the codec negotiation between the gateways been successful?

Note	The Cisco Unified IP Phone 7965 supports these major codecs: G.722, G.711, G.729, and iLBC.

Activity Verification

You have completed this task when you attain these results:

- You verified that the H.245 codec negotiation was successful. The **debug h245 asn1** command displayed the codec proposal that was sent by the HQ gateway (g7231), which was accepted by the BR gateway.
- You identified that the reason for the call disconnect must be that the Cisco Unified IP phones do not support the G.723 codec.

Task 3: Configure Asymmetric Codec Negotiation

In this task, you will configure multiple codec proposals and examine their negotiation.

Activity Procedure

Complete these steps on your HQ and BR gateways:

Step 1 On your HQ gateway, create a codec class with this preference order:

- First codec: g723r53
- Second codec: iLBC
- Third codec: g729br8 (Annex B)

Step 2 On your HQ gateway, remove the G.723 codec from the VoIP dial peer (3000), and attach the codec class to the dial peer.

Step 3 On your BR gateway, create a codec class with this preference order:

- First codec: iLBC
- Second codec: g723r53
- Third codec: g729br8 (Annex B)

Step 4 On your BR gateway, configure a VoIP dial peer (2000) with these parameters:

- The destination pattern should match the DID range of HQ phones (5552XXX).
- The session target should point to your HQ gateway loopback 0 address (10.p.250.101, where p is your pod number).

- It should have an associated codec class that is defined on your BR gateway in the previous step.

Step 5 From an HQ phone, call your BR phone (5553001), and answer the call. The call should work. Examine the inbound dial peer selection on your BR gateway. Check the negotiated codec using the **debug h245 asn1** command.

Step 6 From your BR phone, call an HQ phone (5552001), and answer the call. The call should disconnect. Use available debug methods to examine the cause of the failure.

Activity Verification

You have completed this task when you attain these results:

- You verified that inbound dial peers were successfully matched based on the **destination-pattern** command. The **debug voice dialpeer** command could be used to validate dial-peer matching.
- You examined codec negotiation using the **debug h245 asn1** command. You verified that the codec list was sent by the originating gateway to the terminating gateway. The terminating gateway looks up its prioritized codec list and chooses the first codec, if available, from the received H.245 proposal. The HQ gateway selects g723r53, which is not supported by IP phones, and the call fails.
- You confirmed that the codec selection was based on the priorities that were configured on the terminating gateway.

Task 4: Configure Symmetric Codec Negotiation and Examine Fast Start

In this task, you will configure multiple codec proposals to make VoIP calls work in both directions.

Activity Procedure

Complete these steps on your HQ and BR gateways:

Step 1 On your HQ and BR gateways, modify the codec class to use these codecs only, in this preference order:

- First codec: g729br8 (Annex B)
- Second codec: iLBC

Step 2 Make sure that the codec class is attached to the VoIP dial peers on both gateways.

Step 3 Test VoIP calls in both directions. The calls should work.

Step 4 Use the **debug h245 events** command to ensure which H.323 call setup method (fast start or slow start) is used by default on Cisco voice gateways.

Activity Verification

You have completed this task when you attain these results:

- You verified that VoIP calls worked between the phones in the HQ and BR sites. You may have performed some debugging to examine the dial-peer matching and H.245 codec negotiation.
- You found that Cisco gateways used the fast-start method by default.

Task 5: Configure Slow Start and the Interface Bind Feature

In this task, you will change the default fast-start method to slow start and configure the interface bind feature.

Activity Procedure

Complete these steps:

Step 1 Configure your HQ gateway to use slow-start signaling. Leave your BR gateway at the default setting of fast-start signaling.

Step 2 Make calls between the sites. Use appropriate debug commands on both gateways to ascertain which signaling method is used in which direction. Examine the slow-start signaling process.

Step 3 On both gateways (HQ and BR), source the H.323 signaling traffic from the loopback 0 interfaces.

Step 4 From an HQ phone, call your BR phone. Use the **debug h245 events** command on your BR gateway to verify that the signaling messages are sourced from the loopback 0 IP address.

Activity Verification

You have completed this task when you attain these results:

- You verified that slow start was used for HQ-originated calls, and fast start was used for BR-originated calls.
- You examined the slow-start process and confirmed that H.245 negotiation started after the call was answered and consisted of three main phases: exchange of TCSs, master and slave determination, and OLC exchange.
- You verified that the H.323 gateway interface bind feature caused the signaling packets to be sourced from the defined interface. This can be confirmed with the **debug h225 events** command.

Task 6: Configure SIP Signaling

In this task, you will use SIP as the VoIP signaling protocol.

Activity Procedure

Complete these steps on your HQ gateway:

Step 1 Create VoIP dial peer 3001 with the same destination target and destination pattern as dial peer 3000. Configure the new VoIP dial peer to use SIPv2.

Step 2 Set the preference of the VoIP dial peer 3000 to a worse value, to make it the second choice.

Step 3 From an HQ phone, call your BR phone and use the appropriate debug command to examine the SIP message exchange. Based on the SDP body, decide which offer mechanism (early offer or delayed offer) is used by default on Cisco gateways, and write it in the space that is provided:

_____

Step 4 On your HQ gateway, configure the bind interface feature for SIP signaling and media. Source the traffic from the loopback 0 IP address.

Activity Verification

You have completed this task when you attain these results:

- You examined the SIP signaling using the **debug ccsip messages** command.
- You verified that Cisco gateways use SIP early offer by default, by identifying that the SDP body, which contains the codec proposals, was carried in the INVITE message.
- You shut down the SIP dial peer on your HQ gateway to make sure that H.323 is used in the next exercises.

Lab 3-1: Configuring Cisco Unified Communications Manager Express to Support Endpoints

Complete this lab activity to practice what you learned in the related module.

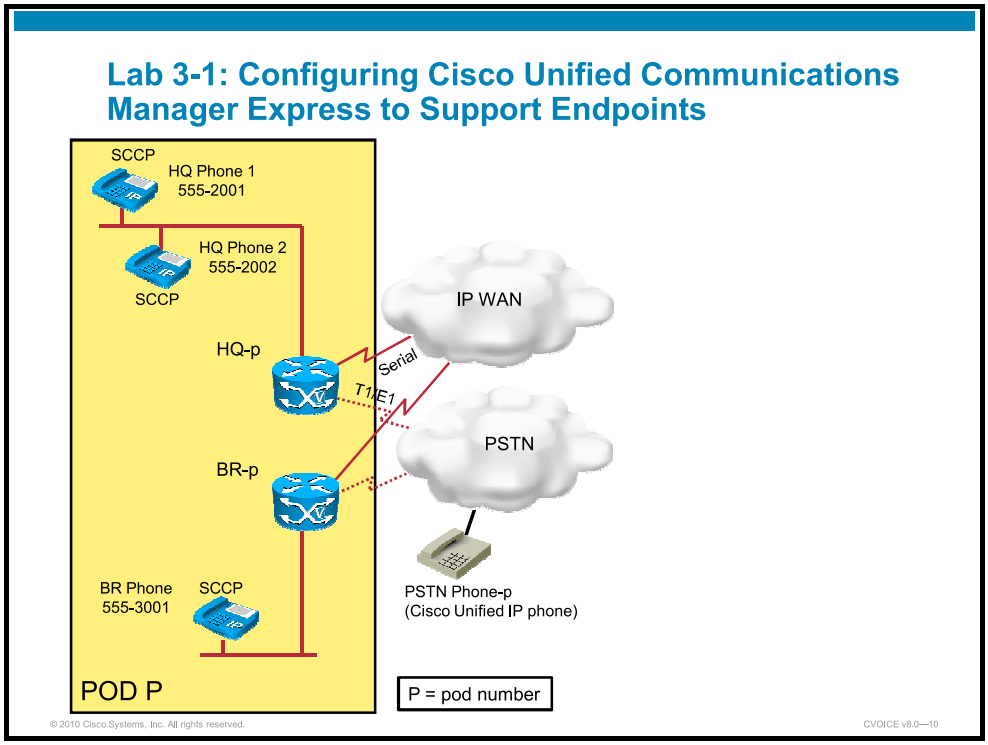
Activity Objective

In this activity, you will implement SCCP endpoints. The HQ site will host two SCCP phones and, optionally, one Cisco IP Communicator. The BR site will host one SCCP phone. After completing this activity, you will be able to meet these objectives:

- Configure SCCP-related Cisco Unified Communications Manager Express parameters
- Implement SCCP endpoints

Visual Objective

The figure illustrates what you will accomplish in this activity.



Required Resources

These are the resources and equipment that are required to complete this activity:

- Two Cisco Unified IP phones in your HQ site
- One Cisco Unified IP phone in your BR site

Command List

The table describes the commands that are used in this activity.

Cisco IOS Commands

Command	Description
ip dhcp pool	Defines a DHCP pool of addresses that must include a network, default router, and option 150
max-dn	Defines the maximum number of directory numbers that are configured in telephony-service configuration mode
max-ephones	Defines the maximum number of SCCP endpoints that are configured in telephony-service configuration mode
load	Defines the binding of the image filename to phone mode that is configured in telephony-service configuration mode
time-format	Defines the time format that is displayed by endpoints that is configured in telephony-service configuration mode
date-format	Defines the date format that is displayed by endpoints that is configured in telephony-service configuration mode
type	Defines the type of the endpoint that is configured in ephone configuration mode. Needed to identify the desired firmware image.
telephony-service	Enters configuration mode for SCCP-based Cisco Unified Communications Manager Express
(no) auto-reg-phone	Controls autoregistration of SCCP endpoints
cnf-file location tftp: | flash:	Defines the configuration file location for SCCP endpoints
cnf-file perphone | perphonetype	Defines the granularity of configuration files for SCCP endpoints
create cnf-files	Writes configuration files into the configured or default cnf-file location
ephone ***ephone-id***	Defines an SCCP endpoint with an identifier and enters ephone configuration mode
mac-address	Configures an endpoint MAC address that is configured in ephone configuration mode
button button-index:dn-index	Associates a phone button with a directory number index. Multiple separator options exist, in addition to ':'
show telephony-service	Displays the SCCP-related Cisco Unified Communications Manager Express system parameters. Multiple options are available to examine specific settings.
show ephone	Displays the status of registered SCCP endpoints
show dial-peer voice	Displays the voice dial peers. The summary option provides a brief output.
debug tftp event	Monitors TFTP server events
debug ephone register	Monitors SCCP endpoint registration

Job Aids

These job aids are available to help you complete the lab activity.

Internal Numbering Plan

	Local HQ Site (EU)	Local BR Site (NA)
Internal numbering	555-2XXX	555-3XXX

Task 1: Delete Existing SCCP Endpoints on HQ Gateway

In this task, you will remove existing SCCP endpoints and block autoregistration on your HQ gateway.

Activity Procedure

Complete these steps on your HQ and BR gateways:

Step 1 Disable autoregistration and autoassignment of SCCP endpoints.

Step 2 Delete the autoconfigured ephones.

Step 3 Delete the manually defined ephone-dns.

Activity Verification

You have completed this task when you attain these results:

- You displayed the existing ephones to make sure that all autoregistered ephones had been deleted.
- You verified that all phones reported “registration rejected.”

Task 2: Configure Cisco Unified Communications Manager Express System Parameters for SCCP Endpoints

In this task, you will configure general system parameters for SCCP-related Cisco Unified Communications Manager Express functionality on the HQ and BR gateways.

Activity Procedure

Complete these steps on your HQ gateway:

Step 1 Verify that the DHCP service is running on the HQ gateway, including option 150. The Cisco Unified Communications Manager Express should use the loopback 0 address for communications with endpoints.

Step 2 Enable IPv4 and IPv6 operations in the telephony service mode, with preference to IPv4.

Step 3 Configure a European-style date and time format to be displayed on the SCCP endpoints (dd-mm-yy and 24h).

Step 4 Set the location of endpoint files to flash and allow per-phone configuration files.

Step 5 Perform the same procedure on your BR gateway to configure Cisco Unified IP phone system settings, with this exception:

- Configure the North American date and time format to be displayed on the SCCP endpoints (mm-dd-yy and 12h).

Activity Verification

You have completed this task when you attain this result:

- If you display the systemwide telephony-service settings, you can verify that the desired parameters are in place.

Task 3: Configure SCCP Endpoints

In this task, you will configure one SCCP endpoint in each site.

Activity Procedure

Complete these steps:

Step 1 On your HQ gateway, create five dual-line SCCP directory numbers with extensions 5552001, 5552002, 5552011, 5552012, and 5552003.

Step 2 On your HQ gateway, configure an ephone [1] with these parameters:

- Use the MAC address of your first HQ phone.
- Attach ephone-dn with extension 5552001 to the first button.
- Attach ephone-dn with extension 5552011 to the second button.
- Select the appropriate phone type (7965).

Step 3 On your HQ gateway, configure an ephone [2] with these parameters:

- Use the MAC address of your second HQ phone.
- Attach ephone-dn with extension 5552002 to the first button.
- Attach ephone-dn with extension 5552012 to the second button.
- Attach ephone-dn with extension 5552003 to the third button.
- Select the appropriate phone type (7965).

Step 4 On your HQ gateway, enable debugging of the ephone registration process.

Step 5 On your HQ gateway, re-create the configuration files.

Note	Phone reset can be triggered either by using the **reset** command in ephone configuration mode, or by pressing the **Settings** button and entering the sequence ****#**** on the phone keypad.

Step 6 Repeat the procedure to configure an SCCP endpoint on your BR gateway, with the extension number 5553001.

Activity Verification

You have completed this task when you attain these results:

- You examined the booting process of the SCCP endpoints.
- You verified successful phone registration at the respective Cisco Unified Communications Manager Express devices.
- You verified that the SCCP phones could call each other within the HQ site and through the IP WAN.

Task 4: Configure Support for Cisco IP Communicator (Optional)

In this task, you will configure the HQ-based Cisco Unified Communications Manager Express to support the Cisco IP Communicator.

Activity Procedure

Complete these steps on your HQ gateway:

Step 1 Install Cisco IP Communicator on your local computer. Your computer must have IP connectivity to the loopback 0 address of the HQ gateway. Depending on your lab setup, you may have to update the routing on your computer or plug your computer into an HQ phone.

Step 2 Start Cisco IP Communicator and select **Menu > Preferences > Network**.

Step 3 In the Device Name section, choose the desired network adapter, and note the device name in the format SEP<*adapter-mac-address*>.

Step 4 In the TFTP Servers section, select **Use these TFTP servers**:, and enter the HQ gateway loopback 0 address (10.p.250.101, where p is your pod).

Step 5 On your HQ gateway, create a new ephone-dn with the extension 555-2004 and a new ephone with the MAC address that was noted in Step 3 and the extension that is attached to its first button.

Activity Verification

You have completed this task when you attain these results:

- You verified that Cisco IP Communicator registered with Cisco Unified Communications Manager Express.
- You verified that calls worked between Cisco IP Communicator and other phones.

Lab 4-1: Implementing Digit Manipulation

Complete this lab activity to practice what you learned in the related module.

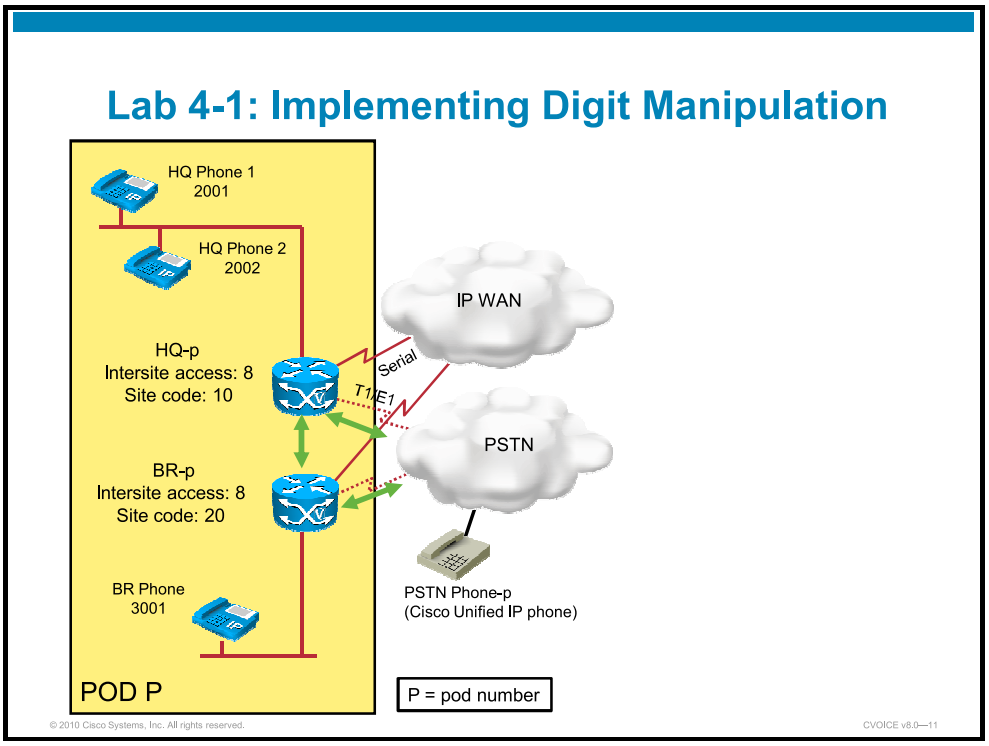
Activity Objective

In this activity, you will change the internal numbering plan to a four-digit scheme and configure digit manipulation to allow PSTN and intersite calls. After completing this activity, you will be able to meet these objectives:

- Configure digit manipulation for inbound PSTN calls
- Configure digit manipulation for outbound PSTN calls
- Configure digit manipulation to enable intersite VoIP calls, using an intersite prefix and site codes

Visual Objective

The figure illustrates what you will accomplish in this activity.



Required Resources

These are the resources and equipment that are required to complete this activity:

- A PSTN phone
- Two Cisco Unified IP phones in your HQ site
- One Cisco Unified IP phone in your BR site

Command List

The table describes the commands that are used in this activity.

Digit Manipulation Commands

Command	Description
digit-strip	Strips all the digits that explicitly match the POTS dial peer. Digit stripping is enabled by default on POTS dial peers.
prefix ***digits***	Specifies the prefix of the dialed digits for a dial peer
forward-digits [0-32]| all | extra	Specifies which digits to forward for voice calls
num-exp ***dialed-digits substitution***	Defines how to expand a telephone extension number into a particular destination pattern
voice translation-rule *rule tag*	Defines a voice translation rule for voice calls
rule ***precedence /match/ /replace/*** *[***type*** ***{match-type replace-type}*** *[***plan*** ***{match-plan replace-plan}]]***	Defines a rule within a voice translation rule
voice translation-profile ***profile-name***	Specifies a translation profile for all incoming VoIP calls
translate {called | calling | redirect-called} ***translation-rule-number***	Associates a translation rule with a voice translation profile
translation-profile {incoming | outgoing} ***name***	Assigns a translation profile to a dial peer
test voice translation-rule ***number input-test-string*** [**type** ***match-type*** [**plan** ***match-type***]	Tests the functionality of a translation rule
debug isdn q931	Monitors ISDN Q931 signaling
debug voice translation	Monitors translation operations

Job Aids

These job aids are available to help you complete the lab activity.

Dial Plan

The table represents the dial plan that will be used in the labs.

Site Internal Numbering Plan and PSTN DID Ranges

	HQ Site (EU)	BR Site (NA)
Internal extensions	2XXX	3XXX
Site codes	810	820
PSTN access code	0	9
Local DID range	555-2XXX	555-3XXX
National DID range	51**p**-555-2XXX	52**p**-555-3XXX
International DID range	55-51**p**-555-2XXX	66-52**p**-555-3XXX

Note	p: 1 to 2 (pod number)

Valid Numbers in Simulated PSTN

	Calls from HQ (EU) to PSTN	Calls from BR (NA) to PSTN
Local calls	**N**XX-XXXX (7 digits), TON: unknown **N**XX-XXXX (7 digits), TON: subscriber Example: **455-8000**	**N**XX-XXXX (7 digits), TON: unknown **N**XX-XXXX (7 digits), TON: subscriber Example: **455-8000**
National calls	0-**N**XX-**N**XX-XXXX, TON: unknown (0 + 3-digit area + 7 digits) **N**XX-**N**XX-XXXX, TON: national (3-digit area + 7 digits) Example: **0-455-455-8000**	1-**N**XX-**N**XX-XXXX, TON: unknown (1 + 3-digit area + 7 digits) **N**XX-**N**XX-XXXX, TON: national (3-digit area + 7 digits) Example: **1-455-455-8000**
International calls	00 + any number of digits, TON: unknown Any number of digits, TON: international Example: **00-23-455-455-8000**	011 + any number of digits, TON: unknown Any number of digits, TON: international Example: **011-23-455-455-8000**
Emergency calls	112, TON: unknown	911, TON: unknown

Note	**N** represents a digit between 2 and 9.

Task 1: Fix Outbound International PSTN Calling from BR

In this task you will fix the issue of ISDN switch-type primary-ni, which is used at the BR gateway. The primary-ni automatically modifies the called-party number for outbound calls, if it is in NANP format (011 followed by 12 digits, which includes a two-digit country code). The primary-ni changes the type of number (TON) to international. Such a called-party number is not compliant with the simulated PSTN that is used in the lab.

Note	If the 011 prefix is used, the correct TON should be unknown.

Activity Procedure

Complete these steps on your BR gateway:

Step 1 Enable ISDN Q931 debugging at the BR gateway. Make sure that the terminal monitor is configured.

Step 2 Place a PSTN international call from your BR phone. For example, dial 9-011-77-455-455-1000#, where 9 is the PSTN access code and # prevents waiting for the interdigit timeout. This call will fail. Observe the type of number that was automatically set for this call by the ISDN process at the BR gateway.

Step 3 Create a translation rule (using tag 1) and translation profile that set the type of number to unknown for all called-party numbers that start with 9011.

Step 4 Associate the translation profile with the ISDN PRI voice port in the outbound direction.

Step 5 Place the same test call again, and it should succeed.

Activity Verification

You have completed this task when you attain these results:

- You observed that, if you are dialing international PSTN number in NANP format, the primary-ni automatically sets TON international.
- You fixed this behavior of the primary-ni by using a voice translation rule and voice translation profile.

Task 2: Change Internal Numbering Plan to Four-Digit Scheme

In this task, you will convert the internal numbering plan to four-digit schemes: 2XXX in the HQ site and 3XXX in the BR site.

Activity Procedure

Complete these steps on your HQ and BR gateways:

Step 1 On your HQ gateway, change the extension number of your SCCP endpoints, using this procedure:

- Modify the ephone-dns:
  - First HQ phone: 555-2001 to 2001 and 555-2011 to 2011
  - Second HQ phone: 555-2002 to 2002, 555-2012 to 2010, and 555-2003 to 2003
  - Optional HQ Cisco IP Communicator: 555-2004 to 2004
- Restart the endpoint (in ephone mode).

Step 2 On your BR gateway, change the extension number of your SCCP endpoint, using this procedure:

- Modify the ephone-dn
  - BR phone: 555-3001 to 3001
- Restart the endpoint (in ephone mode).

Activity Verification

You have completed this task when you attain these results:

- You verified that the endpoints had reregistered and obtained four-digit extension numbers in the respective range.
- You placed test calls and verified that intrasite calls continued to work using the modified numbers. You verified that the intersite calls no longer worked, due to the updated numbering scheme.
- From your HQ and BR phones, you called the PSTN numbers and viewed the calling number that was displayed on the PSTN phone. The calling number was a four-digit number that does not allow callback.

Task 3: Manipulate Calling Number in Outbound PSTN Calls

In this task, you will configure digit manipulation to properly present the calling number when placing calls to the PSTN.

Activity Procedure

Complete these steps:

Step 1 Manipulate the calling number in outbound PSTN calls on your HQ gateway:

- Configure the appropriate translation profiles and rules to convert the calling number from 2XXX to 5552XXX.
- Apply the translation profile to the ISDN PRI voice port in the outgoing direction. Alternatively, you could apply the translation profile to the appropriate POTS dial peers.

Step 2 Manipulate the calling number in outbound PSTN calls on your BR gateway:

- Configure the appropriate translation profiles and rules to convert the calling number from 3XXX to 5553XXX.

Note You will need to reuse the voice translation profile that was configured in Task 1, as you can associate only one translation profile with a single voice port in one direction!

- Make sure that the translation profile is applied to the ISDN PRI voice port in the outgoing direction.

Step 3 Debug the translations using the appropriate Cisco IOS commands.

Note Simulated PSTN works differently from real PSTN. When calling national or international PSTN numbers in the classroom, the PSTN phone displays the calling number as a local number that is configured in this task. Real PSTN would prefix the correct area and country codes, depending on how far the call was sent.

Activity Verification

You have completed this task when you attain these results:

- You tested the translation rule by using the appropriate Cisco IOS commands.
- You placed PSTN calls from your HQ and BR phones and verified that the calling number was presented using the correct DID site range.
- You enabled appropriate debugging on your gateways and monitored the translation operations.

Note If you want to adjust the calling numbers for the simulated PSTN, you can optionally configure multiple translation profiles and apply them to the appropriate outbound dial peers (local, national, and international). The correct area or country code can then be prefixed, based on the selected destination.

Task 4: Manipulate Calling Number and Called Number in Inbound PSTN Calls

In this task, you will configure digit manipulation to properly present the calling number and transform the called number when receiving calls from the PSTN, so that the callback works without editing the number at the phone.

Activity Procedure

Complete these steps:

Step 1 Manipulate the numbers in inbound PSTN calls on your HQ gateway. Configure appropriate translation profiles and rules to meet these needs:

- Apply the correct prefix to the received calling number to enable callback:
  - If the TON is subscriber, prefix the PSTN access code 0 (local calls).
  - If the TON is national, prefix the PSTN access code 0 and another 0 for European national calls.
  - If TON is international, prefix the PSTN access code 0 and 00 for European international calls.
- Convert the called number from the DID format (local, national, or international) to an internally used extension.

Note Simulated PSTN delivers the entire called number, except for the leading zeros. This is different from real PSTN, which delivers the local number after stripping the area and country codes.

- Apply the translation profile to the ISDN PRI voice port in the incoming direction.

Step 2 Manipulate the numbers in inbound PSTN calls on your BR gateway. Configure the appropriate translation profiles and rules to meet these needs:

- Apply the correct prefix to the received calling number to enable callback:
  - If the TON is subscriber, prefix the PSTN access code 9 (local calls).
  - If the TON is national, prefix the PSTN access code 9 and 1 for North American long distance national calls.
  - If the TON is international, prefix the PSTN access code 9 and 011 for North American international calls.
- Convert the called number from the DID format (local, national, or international) to an internally used extension.
- Apply the translation profile to the ISDN PRI voice port in the incoming direction.

Step 3 Debug the translations using the appropriate Cisco IOS commands.

Activity Verification

You have completed this task when you attain these results:

- You tested the translation rule by using the appropriate Cisco IOS commands.

- You called your HQ phones from the PSTN phone (from various lines), using the correct DID numbers (local, national, international), and verified that the calls were delivered to the desired endpoints and that the calling number was presented using the correct prefixes.
- You called your BR phone from the PSTN phone (from various lines), using the correct DID number (local, national, international), and verified that the calls were delivered to the desired endpoint and that the calling number was presented using the correct prefixes.
- You enabled appropriate debugging on your gateways and monitored the translation operations.

Task 5: Manipulate Calling Number and Called Number in Intersite VoIP Calls

In this task, you will configure digit manipulation to enable intersite VoIP calls using intersite prefix 8 and these site codes: 10 for HQ and 20 for BR.

Activity Procedure

Complete these steps:

Step 1 On your HQ gateway, enable HQ-to-BR VoIP calls using intersite prefix 8 and these site codes: 10 for HQ and 20 for BR.

- Configure appropriate translation profiles and rules to meet these needs:
  - Add the prefix 810 to the calling number in an outbound direction towards the BR site.
  - Strip the prefix 810 from the called number in an inbound direction from the BR site.
- Modify the VoIP dial peer (3000) to match the complete number, including the intersite prefix and site code.
- Apply the translation profile for an outbound direction to the VoIP dial peer (3000).
- Configure a new inbound dial peer (use tag 2) that matches all VoIP calls, and apply the translation profile for the inbound direction to this new inbound VoIP dial peer.
- At the inbound VoIP dial peer (2), configure the same preferred codecs as those used by the outbound VoIP dial peer (3000).

Step 2 On your BR gateway, enable BR-to-HQ VoIP calls using intersite prefix 8 and these site codes: 10 for HQ and 20 for BR.

- Configure the appropriate translation profiles and rules to meet these needs:
  - Add the prefix 820 to the calling number in an outbound direction towards the HQ site.
  - Strip the prefix 820 from the called number in an inbound direction from the HQ site.
- Modify the VoIP dial peer (2000) to match the complete number, including the intersite prefix and site code.
- Apply the translation profile for the outbound direction to the VoIP dial peer (2000).

- Configure a new inbound dial peer (use tag 2) that matches all VoIP calls, and apply the translation profile for the inbound direction to this new inbound VoIP dial peer.
- At the inbound VoIP dial peer (2), configure the same preferred codecs as those used by the outbound VoIP dial peer (2000).

Step 3 Place a test call from the HQ site to the BR site, using the intersite prefix and the site code (dial 8203001 from an HQ phone). The call setup should succeed, but you will notice that, shortly after dialing, the caller ID is changed to the extension only (3001). This behavior could confuse a dialer. It can be disabled in both directions by using the following commands, entered at the HQ and BR gateways:

```
voice service voip
 no supplementary-service h225-notify cid-update
```

Activity Verification

You have completed this task when you attain this result:

- You successfully placed intersite calls using the intersite prefix and site codes, and you verified that the calling and called numbers were presented correctly.

Lab 4-2: Implementing Path Selection

Complete this lab activity to practice what you learned in the related module.

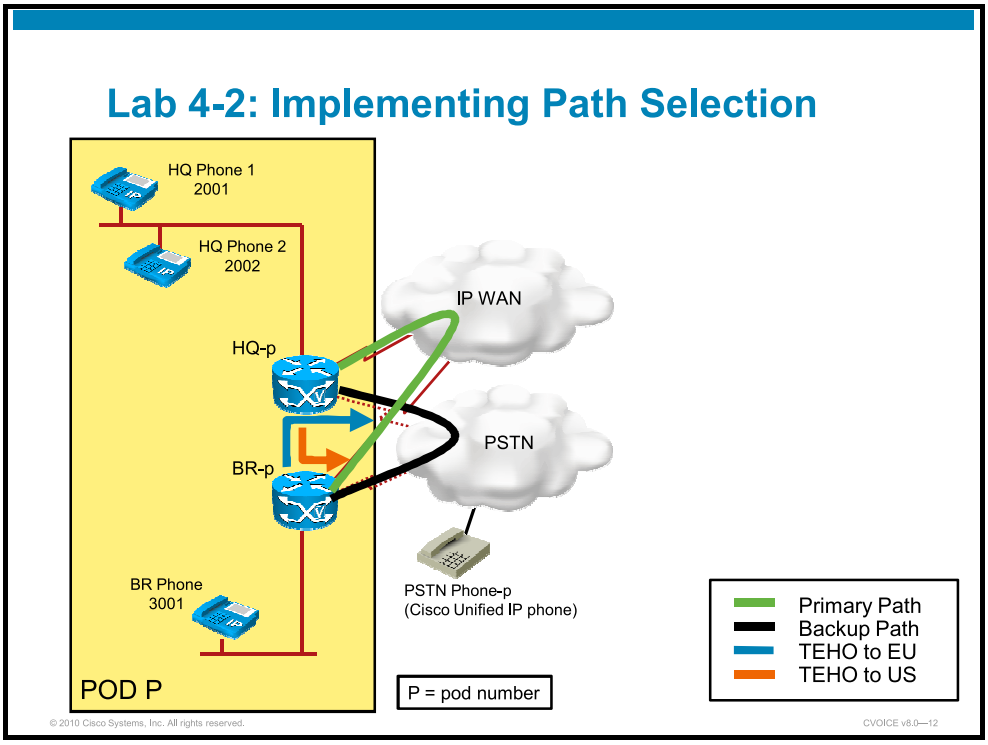
Activity Objective

In this activity, you will implement path selection to ensure continuous VoIP service and toll bypass. After completing this activity, you will be able to meet these objectives:

- Configure a backup PSTN path for intersite calls
- Configure TEHO to provide toll bypass

Visual Objective

The figure illustrates what you will accomplish in this activity.



Required Resources

These are the resources and equipment that are required to complete this activity:

- A PSTN phone
- Two Cisco Unified IP phones in your HQ site
- One Cisco Unified IP phone in your BR site

Command List

The table describes the commands that are used in this activity.

Digit Manipulation Commands

Command	Description
digit-strip	Strips all the digits that explicitly match the POTS dial peer. Digit stripping is enabled by default on POTS dial peers.
prefix ***digits***	Specifies the prefix of the dialed digits for a dial peer
forward-digits [0-32]| all | extra	Specifies which digits to forward for voice calls
num-exp ***dialed-digits substitution***	Defines how to expand a telephone extension number into a particular destination pattern
voice translation-rule ***rule tag***	Defines a voice translation rule for voice calls
rule ***precedence /match/ /replace/*** **[type** ***{match-type replace-type}*** **[plan** ***{match-plan replace-plan}]]***	Defines a rule within a voice translation rule
voice translation-profile ***profile-name***	Specifies a translation profile for all incoming VoIP calls
translate {called | calling | redirect-called} ***translation-rule-number***	Associates a translation rule with a voice translation profile
translation-profile {incoming | outgoing} ***name***	Assigns a translation profile to a dial peer
test voice translation-rule ***number input-test-string*** **[type** ***match-type*** **[plan** ***match-type*****]**	Tests the functionality of a translation rule
debug isdn q931	Monitors ISDN Q931 signaling
debug voice translation	Monitors translation operations

Job Aids

These job aids are available to help you complete the lab activity.

Dial Plan

The table represents the dial plan that will be used in the labs.

Site Internal Numbering Plan and PSTN DID Ranges

	HQ Site (EU)	BR Site (NA)
Internal extensions	2XXX	3XXX
Site codes	810	820
PSTN access code	0	9
Local DID range	555-2XXX	555-3XXX
National DID range	51**p**-555-2XXX	52**p**-555-3XXX
International DID range	55-51**p**-555-2XXX	66-52**p**-555-3XXX

Note	p: 1 to 2 (pod number)

Valid Numbers in Simulated PSTN

	Calls from HQ (EU) to PSTN	Calls from BR (NA) to PSTN
Local calls	**N**XX-XXXX (7 digits), TON: unknown **N**XX-XXXX (7 digits), TON: subscriber Example: **455-8000**	**N**XX-XXXX (7 digits), TON: unknown **N**XX-XXXX (7 digits), TON: subscriber Example: **455-8000**
National calls	0-**N**XX-**N**XX-XXXX, TON: unknown (0 + 3-digit area + 7 digits) **N**XX-**N**XX-XXXX, TON: national (3-digit area + 7 digits) Example: **0-455-455-8000**	1-**N**XX-**N**XX-XXXX, TON: unknown (1 + 3-digit area + 7 digits) **N**XX-**N**XX-XXXX, TON: national (3-digit area + 7 digits) Example: **1-455-455-8000**
International calls	00 + any number of digits, TON: unknown Any number of digits, TON: international Example: **00-23-455-455-8000**	011 + any number of digits, TON: unknown Any number of digits, TON: international Example: **011-23-455-455-8000**
Emergency calls	112, TON: unknown	911, TON: unknown

Note	**N** represents a digit between 2 and 9.

Task 1: Configure Backup PSTN Path for HQ to BR Calling

In this task, you will configure your HQ gateway for a backup PSTN path for calls from HQ to BR.

Activity Procedure

Complete these steps on your HQ gateway:

Step 1 Create a POTS dial peer (use tag 3005) that will back up the intersite calling to BR via PSTN when the VoIP path is unavailable. This POTS dial peer should have the same destination pattern as, but a lower preference than, the VoIP dial peer.

Step 2 Configure the digit manipulation mechanisms to ensure that the call is delivered to the BR via PSTN, and that the calling number that is presented allows callback.

Note	Remember that HQ and BR sites are virtually placed in different countries. Therefore, the intersite that called numbers must have an international format for the call setup to succeed via PSTN.

Step 3 Apply the digit manipulations to the POTS dial peer that is used for the backup.

Activity Verification

You have completed this task when you attain these results:

- You simulated a WAN failure by shutting down the Serial 0/1/0 interface on the HQ gateway.

- While the WAN was in the simulated failure, you placed a call to the BR phone as if you were dialing via the WAN (the intersite prefix 8, site code 20, and 3001). The call should have been redirected via the PSTN.
- The HQ to BR call was presented at the BR phone using a number that allows callback. You tested the callback later, when the WAN was restored. You kept down the WAN for the second task.
- If you attempted a reverse call (BR to HQ), the call should have failed while the WAN was in the simulated failure.

Task 2: Configure Backup PSTN Path for BR to HQ Calling

In this task, you will configure your BR gateway for a backup PSTN path for calls from BR to HQ.

Activity Procedure

Complete these steps on your BR gateway:

Step 1 Create a POTS dial peer (use tag 3005) that will back up the intersite calling to BR via PSTN when the VoIP path is unavailable. This POTS dial peer should have the same destination pattern as, but a lower preference than, the VoIP dial peer.

Step 2 Configure the digit manipulation mechanisms to ensure that the call is delivered to the BR via PSTN, and that the calling number that is presented allows callback.

Note	Remember that HQ and BR sites are virtually placed in different countries. Therefore, the intersite that called numbers must have an international format for the call setup to succeed via PSTN.

Step 3 Apply the digit manipulations to the POTS dial peer that is used for the backup.

Step 4 Fix the translation rule (tag 1) for the primary-ni issue, which was described earlier, by adding a new rule that also sets the TON to unknown if the called-party number starts with 011. This is required, because the current translation rule works for 9011 only. (The PSTN access code 9 would normally be removed by the dial peer 901,1 but this dial peer is not used in the PSTN backup scenario.)

Activity Verification

You have completed this task when you attain these results:

- While the WAN was in the simulated failure (from previous task), you placed a call to an HQ phone as if you were dialing via the WAN (the intersite prefix 8, site code 10, and 2001). The call should have been redirected via the PSTN.
- The BR to HQ call was presented at the HQ phone using a number that allowed callback. You tested the callback later, when the WAN was restored.
- You restored the WAN by activating the Serial 0/1/0 interface.

Task 3: Configure TEHO at HQ for Calls to North America

In this task, you will configure TEHO for calls from the European HQ site to the North American PSTN.

Activity Procedure

Complete these steps:

Step 1 On your HQ gateway, create a VoIP dial peer (use tag 6600) for tail-end hop off to the North American PSTN via the BR gateway. The TEHO dial peer should be used when calling to the virtual North American country with a country code of 66. Remember to use the same codec class as for any other VoIP dial peer at the HQ gateway.

Step 2 Manipulate the sent calling and called numbers for the TEHO calling. The TEHO configuration should not require any changes at the BR gateway. The calling number that is presented should be a correct international number that allows calling back to the TEHO call originator.

Activity Verification

You have completed this task when you attain these results:

- When placing a test call from an HQ phone to the virtual North American country (for instance, 66-455-455-1000#), the call was successfully routed via the BR gateway.
- You enabled appropriate debugging (ISDN, dial-peer) and verified that the expected TEHO path was selected.
- You verified that the correct calling number in international format was presented on the PSTN phone. The call should have rung up at the national, not the international, PSTN phone button.

Task 4: Configure TEHO at BR for Calls to Europe

In this task, you will configure TEHO for calls from the North American BR site to the European PSTN.

Activity Procedure

Complete these steps:

Step 1 On your BR gateway, create a VoIP dial peer (use tag 5500) for tail-end hop off to the European PSTN via the BR gateway, The TEHO dial peer should be used when calling to the virtual European country with a country code of 55. Remember to use the same codec class as for any other VoIP dial peer at the BR gateway.

Step 2 Manipulate the sent calling and called numbers for the TEHO calling. The TEHO configuration should not require any changes at the HQ gateway. The calling number that is presented should be a correct international number that allows calling back to the TEHO call originator.

Activity Verification

You have completed this task when you attain these results:

- When placing a test call from the BR phone to the virtual European country (for instance, 55-455-455-1000#), the call was successfully routed via the HQ gateway.
- You enabled appropriate debugging (ISDN, dial-peer) and verified that the expected TEHO path was selected.
- You verified that the correct calling number in international format was presented on the PSTN phone. The call should have rung up at the national, not the international, PSTN phone button.

Lab 4-3: Implementing Calling Privileges

Complete this lab activity to practice what you learned in the related module.

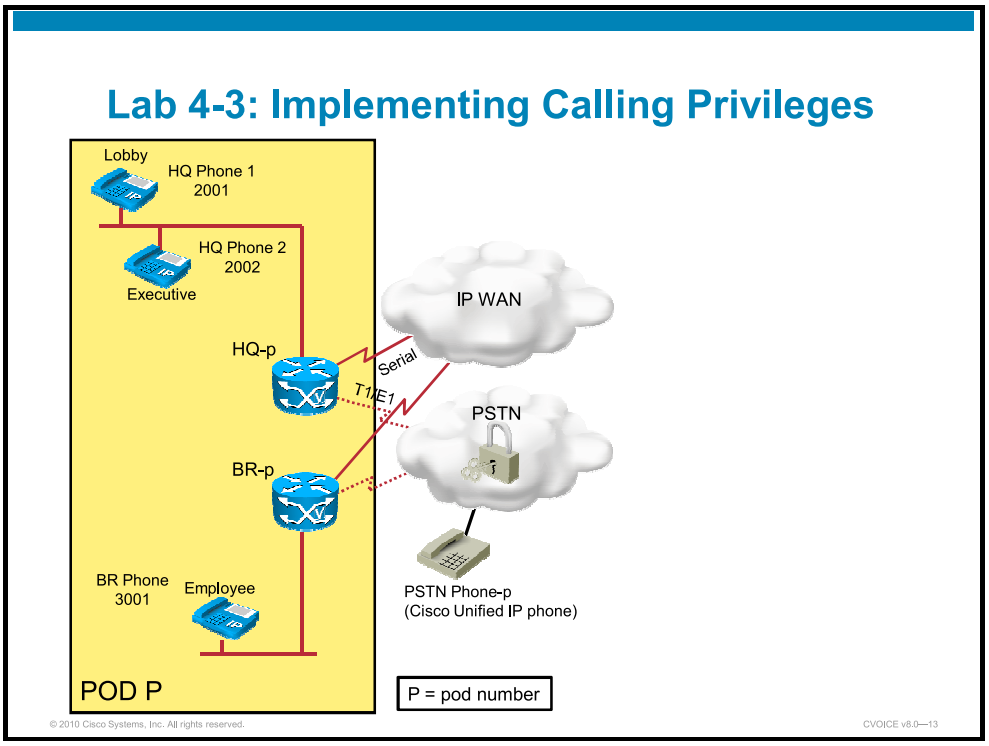
Activity Objective

In this activity, you will implement calling privileges on a gateway using COR. After completing this activity, you will be able to meet these objectives:

- Create COR labels
- Create COR lists and assign members
- Assign COR lists to the appropriate dial peers and Cisco Unified Communications Manager Express endpoints

Visual Objective

The figure illustrates what you will accomplish in this activity.



Required Resources

These are the resources and equipment that are required to complete this activity:

- A PSTN phone
- Two Cisco Unified IP phones in your HQ site
- One Cisco Unified IP phone in your BR site

Command List

The table describes the commands that are used in this activity.

COR Commands

Command	Description
dial-peer cor custom	Specifies that named CORs apply to dial peers
name *cor-name*	Creates a named COR
dial-peer cor list *list-name*	Defines a COR list name
corlist incoming *cor-list-name*	Specifies the COR list to be used when a specified dial peer acts as the incoming dial peer
corlist outgoing *cor-list-name*	Specifies the COR list to be used by outgoing dial peers
show dial-peer cor	Displays COR labels

Job Aids

These job aids are available to help you complete the lab activity.

The table defines call permissions as are required for this lab.

Call Permissions

Endpoint	Description	Permitted to Call
HQ phone 1 second line (2011)	Lobby	PSTN: Emergency Internal: Any
HQ phone 2 first line (2002)	Executive (unrestricted)	Any
HQ phone 2 second line (2012)	Sales	PSTN: Emergency, local, national Internal: Any
BR phone	Employee	PSTN: Emergency, local Internal: Any

All phones should be reachable from PSTN without any restrictions.

Note TEHO destinations and PSTN backup for intersite WAN are not subject to COR configuration, and they can be reached without restrictions.

Site Internal Numbering Plan and PSTN DID Ranges

	HQ Site (EU)	BR Site (NA)
Internal extensions	2XXX	3XXX
Site codes	810	820
PSTN access code	0	9
Local DID range	555-2XXX	555-3XXX
National DID range	51**p**-555-2XXX	52**p**-555-3XXX
International DID range	55-51**p**-555-2XXX	66-52**p**-555-3XXX

Note	p: 1 to 2 (pod number)

Valid Numbers in Simulated PSTN

	Calls from HQ (EU) to PSTN	Calls from BR (NA) to PSTN
Local calls	**N**XX-XXXX (7 digits), TON: unknown **N**XX-XXXX (7 digits), TON: subscriber Example: **455-8000**	**N**XX-XXXX (7 digits), TON: unknown **N**XX-XXXX (7 digits), TON: subscriber Example: **455-8000**
National calls	0-**N**XX-**N**XX-XXXX, TON: unknown (0 + 3-digit area + 7 digits) **N**XX-**N**XX-XXXX, TON: national (3-digit area + 7 digits) Example: **0-455-455-8000**	1-**N**XX-**N**XX-XXXX, TON: unknown (1 + 3-digit area + 7 digits) **N**XX-**N**XX-XXXX, TON: national (3-digit area + 7 digits) Example: **1-455-455-8000**
International calls	00 + any number of digits, TON: unknown Any number of digits, TON: international Example: **00-23-455-455-8000**	011 + any number of digits, TON: unknown Any number of digits, TON: international Example: **011-23-455-455-8000**
Emergency calls	112, TON: unknown	911, TON: unknown

Note	**N** represents a digit between 2 and 9.

Task 1: Configure Call Permissions for the HQ Site

In this task, you will configure COR names (labels), create COR lists with their members, and assign COR lists to the appropriate dial peers and endpoints for call permissions that are applied to the HQ site.

Activity Procedure

Complete these steps on your HQ gateway:

Step 1 Verify that unique dial peers exist for local calls, national calls, international calls, and emergency calls.

Step 2 Create the required COR names.

Note	Naming suggestions for COR names are: emergency, local, national, intl, lobby, executive, and sales.

Step 3 Create the required COR lists.

Note	Naming suggestions for incoming COR lists are: lobby-in, pstn-in, executive-in, and sales-in. Naming suggestions for outgoing COR lists are: lobby-out, executive-out, sales-out, emergency-out, local-out, national-out, and intl-out.

Step 4 Assign the COR lists to appropriate endpoints and dial peers in the correct direction.

Note	TEHO and PSTN backup dial peers are not subject to COR configuration.

Activity Verification

You have completed this task when you attain this result:

- You verified that the call permissions, as they were defined in the job aids for the HQ site, were met by your configuration.

Task 2: Configure Call Permissions for the BR Site

In this task, you will configure COR names (labels), create COR lists with their members, and assign COR lists to the appropriate dial peers and endpoints for call permissions that are applied to the BR site.

Activity Procedure

Complete these steps on your BR gateway:

Step 1 Verify that unique dial peers exist for local calls, national calls, international calls, and emergency calls.

Step 2 Create the required number of COR names.

Step 3 Create the required COR lists.

Step 4 Assign the COR lists to appropriate endpoints and dial peers in the correct direction.

Note TEHO and PSTN backup dial peers are not subject to COR configuration.

Activity Verification

You have completed this task when you attain this result:

- You verified that the call permissions, as they were defined in the job aids for the BR site, were met by your configuration.

Lab 5-1: Implementing Gatekeepers

Complete this lab activity to practice what you learned in the related module.

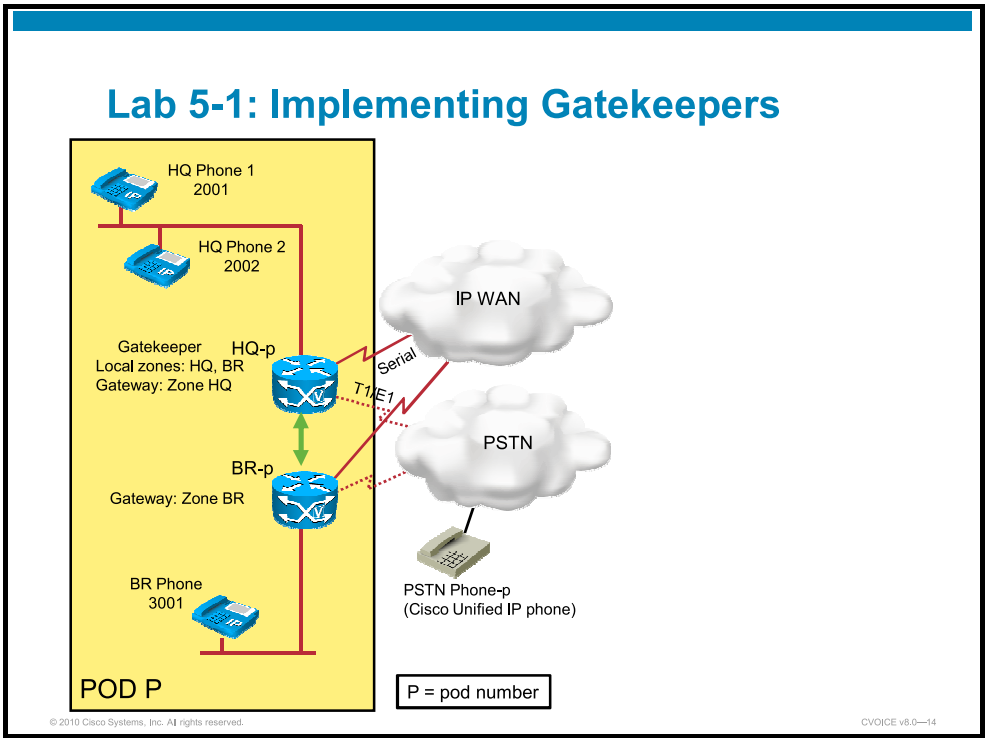
Activity Objective

In this activity, you will configure the corporate gatekeeper HQ as a gatekeeper that controls two zones: HQ and BR. Call routing between the HQ and BR sites will be done via the gatekeeper. After completing this activity, you will be able to meet these objectives:

- Configure an H.323 gatekeeper to support multiple local zones
- Register gateways at the gatekeeper
- Configure technology prefixes
- Implement CAC

Visual Objective

The figure illustrates what you will accomplish in this activity.



Required Resources

These are the resources and equipment that are required to complete this activity:

- Two Cisco Unified IP phones in your HQ site
- One Cisco Unified IP phone in your BR site

Command List

The table describes the commands that are used in this activity.

Gatekeeper Commands

Command	Description
gatekeeper	Enters gatekeeper configuration mode
zone local ***zone-name domain-name*** **[*ras-IP-address*]**	Specifies a zone that is controlled by a gatekeeper
no shutdown	Brings the gatekeeper online
zone prefix ***gatekeeper-name e164-prefix*** **[blast | seq] [gw-priority** ***priority gw-alias*** **[*gw-alias, ...*]]**	Adds a prefix to the gatekeeper zone list
gw-type-prefix ***type-prefix*** **[hopoff** ***gkid1*****] [hopoff** ***gkid2*****] [hopoff** ***gkidn*****] [seq | blast] [default-technology] [gw ipaddr** ***ipaddr*** **[*port*]]**	Adds a technology prefix to the gatekeeper configuration list
bandwidth {interzone | total | session} {default | zone ***zone-name*****}** ***bandwidth-size***	Specifies the maximum aggregate bandwidth for H.323 traffic
show gatekeeper calls	Displays the status of each ongoing call of which a gatekeeper is aware
show gatekeeper status	Displays the overall gatekeeper status, including the authorization and authentication status and zone status
show gatekeeper endpoints	Displays the status of all registered endpoints for a gatekeeper
show gatekeeper gw-type-prefix	Displays the gateway technology prefix table
show gatekeeper zone prefix [all]	Displays the zone prefix table
show gatekeeper zone status	Displays the status of zones that are related to a gatekeeper

Gateway Commands

Command	Description
gateway	Enters gateway configuration mode and enables the gateway to register with a gatekeeper
h323-gateway voip interface	Identifies this as a VoIP gateway interface
h323-gateway voip id ***gatekeeper-id*** **{ipaddr** ***ip-address [port]*****| multicast} [priority** ***priority*****]**	Defines the name and location of the gatekeeper for this gateway
h323-gateway voip h323-id ***interface-id***	Defines the H.323 name of the gateway, identifying this gateway to its associated gatekeeper
h323-gateway voip tech-prefix ***prefix***	Defines the numbers that are used as the technology prefix that the gateway registers with the gatekeeper
session target ras	Enables RAS signaling, which means that a gatekeeper is consulted to translate the E.164 address into an IP address

Gateway and Gatekeeper Monitoring Commands

Command	Description
debug ras	Monitors Registration, Admission, and Status (RAS) messages
debug h225 asn1	Monitors H.225 ASN.1 library messages, which provide a detailed trace of the RAS messages

Job Aids

These job aids are available to help you complete the lab activity.

Gatekeeper and Gateway Addressing

The table defines gatekeeper and gateway addressing.

Internal Numbering Plan

	HQ Site (EU)	BR Site (NA)
Internal extensions	2XXX	3XXX

Gatekeeper and Gateway Addressing

Component	Address	Gateway H.323 ID
HQ gatekeeper	Loopback 0 IP address	-
HQ gateway	Loopback 0 IP address	HQ-gw
BR gateway	Loopback 0 IP address	-

Task 1: Configure Local Zones and Zone Prefixes

In this task, you will configure your HQ router as an H.323 gatekeeper that supports two local zones. You will also configure zone prefixes to enable call routing.

Activity Procedure

Complete these steps on your HQ router:

Step 1 On the HQ gateway, configure a gatekeeper with these two local zones:

- Local zone HQ, domain cisco.com, IP address of loopback 0 interface
- Local zone BR, domain cisco.com

Step 2 Configure prefixes for the local zones HQ and BR. Zone prefixes use the site extensions as seen in the job aids section.

Step 3 Enable the gatekeeper process.

Activity Verification

You have completed this task when you attain these results:

- You viewed the gatekeeper status and verified that the gatekeeper was up.
- When viewing the gatekeeper zone status, you could identify the two local zones: HQ and BR.
- When viewing the gatekeeper zone prefixes, you could verify that they were correct.

Task 2: Configure Gateways to Register with the Gatekeeper

In this task, you will configure the HQ and BR gateways to register with the HQ-based gatekeeper.

Activity Procedure

Complete these steps:

Step 1 On your BR gateway, enable debugging of Registration, Admission, and Status (RAS) messages.

Step 2 Configure your BR gateway to register at the gatekeeper, using these parameters:

- The H.323 interface should be Loopback 0.
- H.323 bind using Loopback 0.
- Registering should be at zone BR.

Step 3 On your BR gateway, configure a new VoIP dial peer (use tag 2002) that routes all calls to the site HQ with extensions 2XXX to the gatekeeper. Remember to apply the same voice-class codec as for other VoIP dial peers.

Step 4 Configure your HQ gateway to register at the gatekeeper using these parameters:

- Interface bind using Loopback 0.
- The H.323 interface should be Loopback 0.
- H.323 bind using Loopback 0.
- The H.323 gateway ID is HQ-gw.
- Disable the registration of extension 2012 (second line of HQ phone 2).
- Registering should be at zone HQ.

Step 5 On your HQ gateway, configure a new VoIP dial peer (use tag 3002) that routes all calls to the site BR with extensions 3XXX to the gatekeeper. Remember to apply the same voice-class codec as for other VoIP dial peers.

Activity Verification

You have completed this task when you attain these results:

- You were able to see the BR gateway registering at the gatekeeper using the RAS debugging and confirmed that the BR gateway had been registered.
- You could confirm that the BR gateway had registered the extension 3001 as E.164-ID.
- You could confirm that the HQ gateway had been registered at the gatekeeper with the correct H.323 ID.
- You could confirm that the HQ gateway had registered all extensions except 2012 as E.164-IDs.
- When you placed an intersite call using an extension, the call succeeded. The calling worked in both directions.
- You examined the RAS registration and call admission messages using the **debug ras** and **debug h225 asn1** commands, and you became familiar with the H.323 debug output.

Task 3: Configure Call Admission Control

In this task, you will calculate the bandwidth requirements for one call and configure the zone bandwidth for calls between the HQ and BR sites.

Activity Procedure

Complete these steps:

Step 1 Determine the codec that is used for intersite calling, and calculate the bandwidth requirements for a single call using the gatekeeper CAC calculation method. Write these in the space that is provided:

_____

_____

Note	Remember that, actually, two different codecs can be negotiated between the gateways that are based on the voice-class codec that is configured. For the calculation and CAC, you must consider both codecs as the initial call setup, though the gatekeeper will take into account the worst codec of these two.

Step 2 Configure the gatekeeper CAC to allow a maximum of one call between the zones in each direction.

Step 3 Enable detailed RAS debugging, and observe that the second interzone call is rejected at the gatekeeper.

Activity Verification

You have completed this task when you attain these results:

- You calculated that a single iLBC call requires 30.4 kb/s (rounded up to 31 kb/s), using the gatekeeper CAC calculation method (2 x codec rate).
- You calculated that a single G.729 call requires 16 kb/s, using the gatekeeper CAC calculation method.
- You verified that one call can be placed successfully between the sites, while the second call will fail.

Lab 5-2: Implementing Cisco Unified Border Element

Complete this lab activity to practice what you learned in the related module.

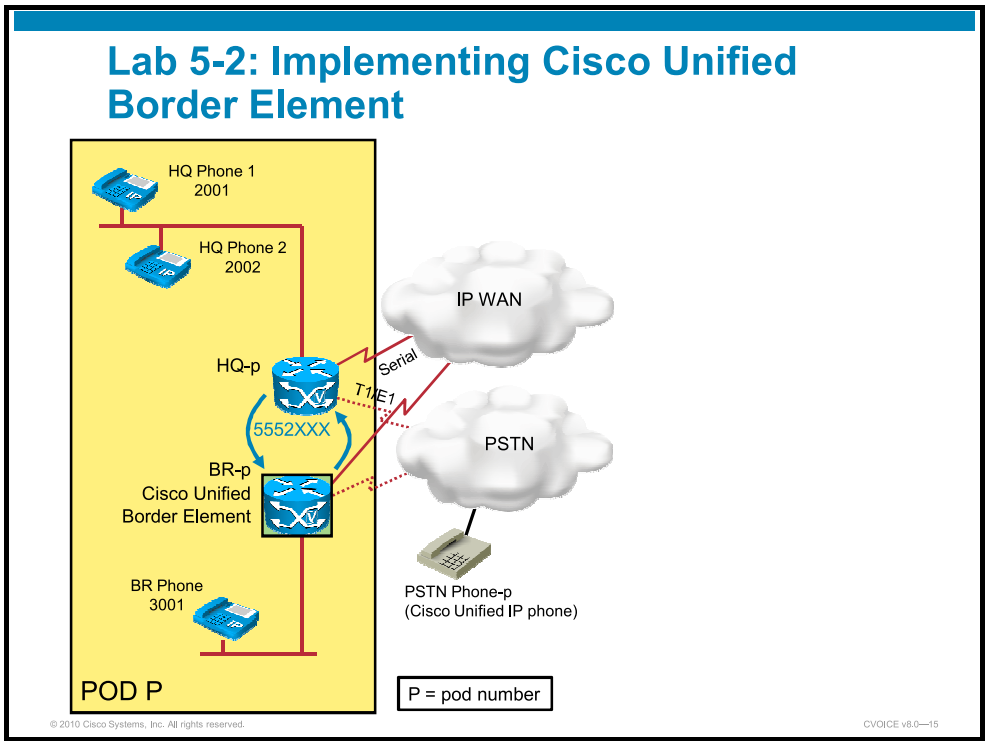
Activity Objective

In this activity, you will implement the Cisco Unified Border Element features: protocol interworking, various media flow methods, and codec transparency. After completing this activity, you will be able to meet these objectives:

- Configure SIP-to-H.323 and H.323-to-H.323 protocol interworking
- Implement codec transparency
- Configure H.323-to-H.323 interworking
- Implement media flow-around and media flow-through

Visual Objective

The figure illustrates what you will accomplish in this activity.



Required Resources

These are the resources and equipment that are required to complete this activity:

- Two Cisco Unified IP phones in your HQ site
- One Cisco Unified IP phone in your BR site

Command List

The table describes the commands that are used in this activity.

Cisco Unified Border Element-Related IOS Commands

Command	Description
`voice service voip`	Enters voice service voip configuration mode
`allow-connections`	Enables Cisco Unified Border Element protocol interworking. Options are: H.323-to-H.323, SIP-to-SIP, H.323-to-SIP, and SIP-to-H.323.
`media flow-around | flow-through`	Configures media flow method. The default is **flow-through**. This command is available in voice service voip, dial peer, or voice class mode.
`h323`	Enters H323 configuration mode (from voice service voip configuration mode)
`call start fast | slow | interwork`	Configures the H.323 signaling method. Default: fast start.
`codec transparent`	Enables the dial peer to transparently pass the codec proposals
`codec`	Configures a codec for a dial peer
`show call active voice brief`	Displays active call parameters, including the RTP addresses
`show voice call status`	Displays brief information about active calls
`show voip rtp connections`	Displays RTP connections for active VoIP call
`debug voip ipipgw`	Monitors Cisco Unified Border Element operations
`debug ccsip messages`	Monitors SIP messages
`debug h225 events`	Monitors H.225 events
`debug h245 events`	Debugs H.245 events
`debug voip dialpeer`	Monitors the matching of inbound and outbound dial peers

Job Aids

No job aids are required to complete this lab activity.

Task 1: Configure Cisco Unified Border Element Functions

In this task, you will reconfigure your dial plan to route calls between your local HQ phones via a BR gateway that works as an H.323-to-H.323 Cisco Unified Border Element.

Activity Procedure

Complete these steps:

Step 1 On your HQ gateway, configure a new VoIP dial peer (use tag 555) to send all calls to 5552XXX to the BR gateway:

- Session target: Loopback 0 IP address of BR gateway
- Codec: G.711 u-law

Step 2 On your HQ and BR gateways, modify the inbound VoIP dial peer (2) to accept only the G.711 u-law codec.

Step 3 On your BR gateway, configure a new VoIP dial peer (use tag 555) to return all calls to 5552XXX back to the HQ gateway:

- Session target: Loopback 0 IP address of HQ gateway
- Codec: G.711 u-law

Step 4 On your HQ gateway, modify the existing voice translation rule that is associated with the inbound VoIP dial peer, to remove 555 from the called number for the inbound call that is being returned back from BR.

Step 5 Try to place a call between your HQ phones using the prefix 555 (for example, dial 5552001 from HQ phone 2). Your call will be blocked at the BR gateway, because it has not been yet enabled for Cisco Unified Border Element functions.

Step 6 Enable Cisco Unified Border Element functions at the BR gateway to allow inbound to outbound 5552XXX calling.

Step 7 On your Cisco Unified Border Element (BR), enable monitoring of Cisco Unified Border Element operations using the **debug voip ipipgw** command, and place a call between your HQ phones, again using the prefix 555. This call should succeed.

Step 8 While the call is active, issue the command **show voip rtp connections** on your Cisco Unified Border Element, to check the two call legs that are associated with the call. Notice that the remote IP address is the same (HQ gateway). Which codec is used for this call? Write it in the space that is provided:

_____

Activity Verification

You have completed this task when you attain these results:

- You successfully placed an H.323-to-H.323 call via the Cisco Unified Border Element and observed how the call was set up using the **debug voip ipipgw** command.
- You could see the two VoIP call legs that terminated at your Cisco Unified Border Element when the call was active. The Cisco Unified Border Element used the default flow-through method.

Task 2: Configure Codec Transparency

In this task, you will configure Cisco Unified Border Element not to participate in codec negotiations.

Activity Procedure

Complete these steps:

Step 1 On your HQ gateway, reapply the voice-class codec to the VoIP dial peer (555) that routes calls between HQ phones via the Cisco Unified Border Element.

Step 2 On your HQ gateway, reapply the voice-class codec to the inbound VoIP dial peer (2).

Step 3 Configure your Cisco Unified Border Element BR to *not* participate in codec negotiations for the calls between HQ phones at the inbound and outbound dial peers (2 and 555), and accept any codec that is determined by the HQ gateway.

Step 4 Place a call between your HQ phones, using the prefix 555, and examine which codec has now been used. Write the answer in the space that is provided:

_____

Activity Verification

You have completed this task when you attain this result:

- You could see that the Cisco Unified Border Element did not actively participate in codec negotiations and that it accepted the codec that was proposed by the HQ gateway. The HQ gateway negotiates codecs that are based on the voice-class codec priority that is configured.

Task 3: Configure SIP-to-H.323 Interworking and Media Flows

In this task, you will reconfigure your Cisco Unified Border Element and HQ gateway for SIP-to-H.323 interworking and modify the Cisco Unified Border Element to media flow-around.

Activity Procedure

Complete these steps:

Step 1 On your HQ gateway, modify the outbound VoIP dial peer (555) that routes calls between HQ phones via the Cisco Unified Border Element, to SIP.

Step 2 On your Cisco Unified Border Element, modify the inbound VoIP dial peer (2) to SIP.

Step 3 Enable your Cisco Unified Border Element for SIP-to-H.323 interworking functions.

Step 4 On your Cisco Unified Border Element, enable the **debug ccsip messages**, and place a call between HQ phones using the prefix 555. Observe the SIP messages that are setting up the call. Which SIP mechanism is used for the call setup, early or delayed offer? Write the answer in the space that is provided:

_____

Step 5 Configure your Cisco Unified Border Element for media flow-around. Place a call, and use the command **show voip rtp connections** to examine that no RTP connections are terminated at the Cisco Unified Border Element.

Activity Verification

You have completed this task when you attain these results:

- You reconfigured your Cisco Unified Border Element for SIP-to-H.323 interworking, and the call via the Cisco Unified Border Element succeeded.
- You observed SIP messages at the Cisco Unified Border Element as the call was setting up.
- You could see that the early offer was used, since the SDP body was included in the SIP INVITE message.
- You could see that, if media flow-around was configured at the Cisco Unified Border Element, RTP streams of an active call were not terminated at the Cisco Unified Border Element.

Lab 6-1: Implementing QoS Using Cisco AutoQoS and Manual Configuration

Complete this lab activity to practice what you learned in the related module.

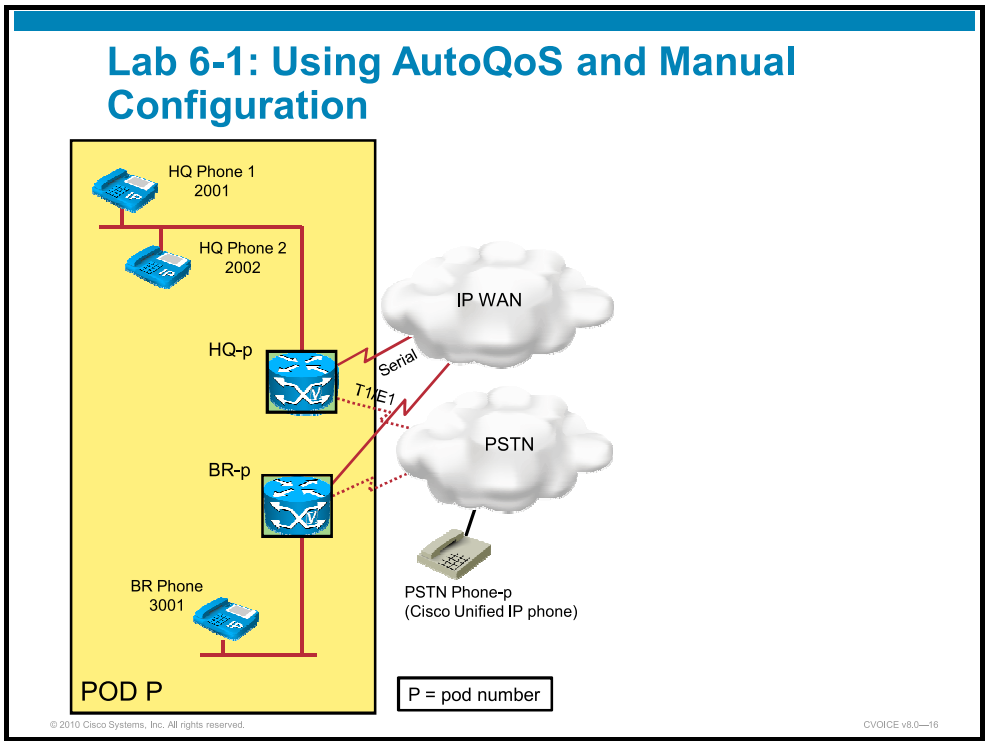
Activity Objective

In this activity, you will implement the best-practice QoS mechanisms using Cisco AutoQoS VoIP, and then you will manually tune the deployed QoS policy. After completing this activity, you will be able to meet these objectives:

- Configure AutoQoS VoIP on routers
- Fine-tune the QoS policy on a switch
- Verify the operations of implemented QoS mechanisms

Visual Objective

The figure illustrates what you will accomplish in this activity.



Required Resources

These are the resources and equipment that are required to complete this activity:

- Two Cisco Unified IP phones in your HQ site
- One Cisco Unified IP phone in your BR site

Command List

The table describes the commands that are used in this activity.

Router QoS Commands

Command	Description
`clock rate`	Configures the clock rate on a serial interface
`bandwidth`	Configures the available bandwidth on an interface
`auto qos voip`	Configures the AutoQoS VoIP feature. Available in interface, subinterface, of Frame Relay DLCI configuration mode.
`priority`	Allocates specified bandwidth to the high-priority queue
`show auto qos`	Displays the QoS policy that is deployed by AutoQoS VoIP
`show policy-map`	Displays the configured policy maps

Job Aids

No job aids are required to complete this lab activity.

Task 1: Evaluate VoIP Quality Without QoS Applied

In this task, you will examine VoIP quality when no QoS mechanisms are being used on the routers.

Activity Procedure

Complete these steps:

Step 1 Ask the instructor to lower the clock rate on the WAN links to your HQ and BR routers to 64,000.

Step 2 Simulate a heavy load by flooding the WAN using the **ping** command:

- From the HQ gateway towards the BR gateway loopback 0 IP address
- Generate ping with 1000 packets, with a packet size of 5000 bytes
- Syntax example (p is your pod number): **ping 10.p.250.102 size 5000 repeat 1000**

Note If you want to interrupt the flood, simultaneously press Ctrl-Shift-6, followed by x.

Step 3 While the WAN link is congested, place an intersite direct call (that does not involve the gatekeeper or Cisco Unified Border Element) by dialing while using a site code. (For instance, from an HQ phone dial 820-3001, or in the opposite direction dial 810-2002.) Examine the audio quality in this way:

- Speak into one receiver and listen to the sound at the other end. You should hear a distinguishable delay.
- On any participating phone, press the **Settings** button, select **Status > Call Statistics**, and examine the parameters that are shown. The expected approximate values are shown in the activity verification section.

Activity Verification

You have completed this task when you attain these results:

- You saw that flooding the network with traffic caused perceptible delay.
- You examined VoIP statistics and determined values in approximately these ranges:
  - Average jitter: 20–500 ms
  - Maximum jitter: 500–700 ms

Task 2: Configure AutoQoS VoIP

In this task, you will configure AutoQoS VoIP on the HQ and BR gateways.

Activity Procedure

Complete these steps:

Step 1 On the HQ gateway, configure the correct bandwidth of 64 kb/s at the Serial subinterface that represents the Frame Relay PVC to BR site (Serial0/1/0.121).

Step 2 On the BR gateway, configure the correct bandwidth of 64 kb/s at the Serial subinterface that represents the Frame Relay PVC to HQ site (Serial0/1/0.111).

Step 3 On the HQ gateway, enter the interface-DLCI configuration mode for the DLCI 121, and enable the AutoQoS VoIP feature while trusting the DSCP markers that are received over the WAN.

Step 4 On the BR gateway, enter the interface-DLCI configuration mode for the DLCI 111, and enable the AutoQoS VoIP feature while trusting the DSCP markers that are received over the WAN.

Step 5 Display the QoS policy that is generated with AutoQoS.

Activity Verification

You have completed this task when you attain this result:

- You examined the QoS policy that was deployed on the gateways when Cisco AutoQoS was activated, using appropriate commands, including the **show auto qos** command.

Task 3: Fine-Tune QoS Policy

In this task, you will fine-tune the QoS policy that has been deployed by the Cisco AutoQoS VoIP feature.

Activity Procedure

Complete these steps:

Step 1 Calculate the bandwidth requirement (including Layer 2) for a single VoIP call, using the following formula:

- BW = (codec payload + Layer 3+ overhead + Layer 2 overhead) * packet rate * 8 bits per byte, with the following arguments:
  - G.729 payload: 20 bytes
  - Layer 3+ overhead (cRTP): 2 bytes
  - Layer 2 overhead (FRF.12): 8 bytes
  - Packet rate: 50 p/s

Step 2 Examine how many calls are supported by the bandwidth that is allocated to the LLQ that is provisioned by AutoQoS VoIP.

Step 3 On the HQ and BR gateways, reduce the LLQ bandwidth to support only two G.729 calls.

Step 4 Simulate a heavy load by flooding the WAN using the **ping** command:

- From the HQ gateway towards the BR gateway loopback 0 IP address
- Generate ping with 1000 packets, with a packet size of 5000 bytes
- Syntax example (p is your pod number): **ping 10.p.250.102 size 5000 repeat 1000**

Step 5 While the WAN link is congested, place an intersite direct call by dialing using a site code. (For instance, from an HQ phone dial 820-3001, or in the opposite direction dial 810-2002.) Examine the audio quality when QoS is implemented. Compare the results with the Task 1 results.

Activity Verification

You have completed this task when you attain these results:

- You calculated that a single G.729 call with cRTP over FRF.12 requires 12 kb/s.
- You calculated that two calls (24 kb/s) require 37.5 percent of a 64 kb/s total bandwidth.
- You allocated 38 percent of total bandwidth to the LLQ and verified the setting using an appropriate command, such as **show policy map**.
- You evaluated VoIP quality with the same congestion in the WAN and noticed an improvement in perceived voice quality due to shorter delay.
- You examined VoIP statistics on the communicating phones and observed the values in approximately these ranges:
  - Average jitter: 2 to 40 ms
  - Maximum jitter: 100 to 180 ms

Answer Key

The correct answers and expected solutions for the activities that are described in this guide appear here.

Lab 1-1 Answer Key: Configuring Voice Ports

The HQ gateway configuration should be like the following:

Task 1, Step 1:

```
ip dhcp pool HQ1-Phones
  ! p is your pod number
   network 10.p.2.0 255.255.255.0
   default-router 10.p.2.101
   option 150 ip 10.p.250.101
!
```

Task 2, Step 1:

```
telephony-service
 max-ephones 5
 max-dn 20
! p is your pod number
 ip source-address 10.p.250.101 port 2000
 auto assign 1 to 2
!
```

Task 2, Step 2:

```
ephone-dn  1  dual-line
 number 5552001
!
ephone-dn  2  dual-line
 number 5552002
!
```

Task 3, Step 1:

```
network-clock-participate wic 0
!
```

Task 3, Step 2:

```
isdn switch-type primary-net5
!
```

Task 3, Step 3:

```
controller E1 0/0/0
 pri-group timeslots 1-8,16
!
```

Task 4, Step 1:

```
dial-peer voice 7 pots
 destination-pattern 0[2-9].....
 port 0/0/0:15
dial-peer voice 10 pots
```



```
 destination-pattern 00[2-9].........
 forward-digits 11
 port 0/0/0:15
dial-peer voice 9011 pots
 destination-pattern 000T
 prefix 00
 port 0/0/0:15
!
```

Task 4, Step 2:

```
dial-peer voice 112 pots
 destination-pattern 112
 port 0/0/0:15
 forward-digits all
dial-peer voice 1120 pots
 destination-pattern 0112
 forward-digits 3
 port 0/0/0:15
!
```

Task 5, Step 1:

```
dial-peer voice 1 pots
 incoming called-number .
 direct-inward-dial
!
```

The BR gateway configuration should be like the following:

Task 1, Step 2:

```
ip dhcp pool BR1-Phones
  ! p is your pod number
   network 10.p.4.0 255.255.255.0
   default-router 10.p.4.102
   option 150 ip 10.p.250.102
!
```

Task 2, Step 4:

```
telephony-service
 max-ephones 5
 max-dn 10
! p is your pod number
 ip source-address 10.p.250.102 port 2000
 auto assign 1 to 1
!
```

Task 2, Step 5:

```
ephone-dn  1  dual-line
 number 5553001
!
```

Task 3, Step 4:

```
network-clock-participate wic 0
```

```
!
isdn switch-type primary-ni
!
controller E1 0/0/0
 pri-group timeslots 1-8
!
```

Task 6, Step 1

```
dial-peer voice 7 pots
 destination-pattern 9[2-9].....
 port 0/0/0:15
dial-peer voice 10 pots
 destination-pattern 91[2-9]..[2-9].....
 port 0/0/0:15
 prefix 1
dial-peer voice 9011 pots
 destination-pattern 9011T
 port 0/0/0:15
 prefix 011
!
```

Task 6, Step 2

```
dial-peer voice 911 pots
 destination-pattern 911
 forward-digits all
 port 0/0/0:15
dial-peer voice 9911 pots
 destination-pattern 9911
 forward-digits 3
 port 0/0/0:15
!
```

Task 7, Step 1

```
dial-peer voice 1 pots
 incoming called-number .
 direct-inward-dial
!
```

Lab 1-2 Answer Key: Configuring DSPs

The HQ gateway configuration should include these commands:

Task 1, Step 4:

```
ephone 1
 codec ilbc
!
```

Task 2, Step 2:

```
voice-card 0
 codec complexity medium
!
```


Task 2, Step 6:

```
voice-card 0
 codec complexity flex
!
```

Lab 2-1 Answer Key: Configuring VoIP Call Legs

The HQ gateway configuration should be like the following:

Task 1, Step 1:

```
dial-peer voice 3000 voip
! p is your pod number
 session target ipv4:10.p.250.102
 destination-pattern 5553...
 no vad
!
```

Task 2, Step 1:

```
dial-peer voice 3000 voip
codec g723r53
!
```

Task 3, Step 1:

```
voice class codec 1
 codec preference 1 g723r53
 codec preference 2 ilbc
 codec preference 3 g729br8
!
```

Task 3, Step 2:

```
dial-peer voice 3000 voip
 no codec g723r53
 voice-class codec 1
!
```

Task 4, Step 1:

```
no voice class codec 1
voice class codec 1
 codec preference 1 g729br8
 codec preference 2 ilbc
!
```

Task 4, Step 2:

```
dial-peer voice 3000 voip
 voice-class codec 1
!
```

Task 5, Step 1:

```
voice service voip
 h323
  call start slow
!
```

Task 5, Step 3:

```
interface Loopback0
 ! p is your pod number
 h323-gateway voip bind srcaddr 10.p.250.101
!
```

Task 6, Step 1:

```
dial-peer voice 3001 voip
! p is your pod number
 session target ipv4:10.p.250.102
 destination-pattern 5553...
 session protocol sipv2
!
```

Task 6, Step 2:

```
dial-peer voice 3000 voip
preference 1
!
```

Task 6, Step 4:

```
voice service voip
 sip
  bind all source-interface Loopback0
!
```

Task 6 verification:

```
dial-peer voice 3001 voip
 shutdown
!
```

The BR gateway configuration should be like the following:

Task 3, Step 3:

```
voice class codec 1
 codec preference 1 ilbc
 codec preference 2 g723r53
 codec preference 3 g729br8
!
```

Task 3, Step 4:

```
dial-peer voice 2000 voip
 destination-pattern 5552...
! p is your pod number
 session target ipv4:10.p.250.101
 voice-class codec 1
!
```

Task 4, Step 1:

```
no voice class codec 1
voice class codec 1
 codec preference 1 g729br8
 codec preference 2 ilbc
```



```
!
```

Task 4, Step 2:

```
dial-peer voice 2000 voip
 voice-class codec 1
!
```

Task 5, Step 3:

```
interface Loopback0
 ! p is your pod number
 h323-gateway voip bind srcaddr 10.p.250.102
!
```

Lab 3-1 Answer Key: Configuring Cisco Unified Communications Manager Express to Support Endpoints

The HQ gateway configuration should be like the following:

Task 1, Step 1:

```
telephony-service
 no auto assign 1 to 2
 no auto-reg-ephone
!
```

Task 1, Step 2:

```
no ephone 1
no ephone 2
!
```

Task 1, Step 3:

```
no ephone-dn  1
no ephone-dn  2
!
```

Task 2, Steps 2 to 4:

```
telephony-service
 protocol mode dual-stack preference ipv4
 cnf-file location flash:
 cnf-file perphone
 time-format 24
 date-format dd-mm-yy
 create cnf-files
!
```

Task 3, Step 1:

```
ephone-dn 1 dual-line
 number 5552001
!
ephone-dn 2 dual-line
 number 5552002
!
ephone-dn 3 dual-line
```

```
 number 5552011
!
ephone-dn 4 dual-line
 number 5552012
!
ephone-dn 5 dual-line
 number 5552003
!
```

Task 3, Step 2:

```
ephone 1
 mac-address 0024.c445.5233
 type 7965
 button 1:1 2:3
```

Task 3, Step 3:

```
ephone 2
 mac-address 0024.C445.4B7F
 type 7965
 button 1:2 2:4 3:5
!
```

Task 3, Step 5

```
telephony-service
 create cnf-files
!
```

Task 4 (Optional), Step 5

```
ephone-dn 6 dual-line
 number 5552004
!
ephone 3
 mac-address 0016.4155.B50B
 type CIPC
 button 1:6
!
```

The BR gateway configuration should include these commands:

Task 1, Step 1:

```
telephony-service
 no auto assign 1 to 1
 no auto-reg-ephone
!
```

Task 1, Step 2:

```
no ephone 1
!
```

Task 1, Step 3:

```
no ephone-dn  1
```



```
!
```

Task 2, Step 5:

```
telephony-service
 protocol mode dual-stack preference ipv4
 cnf-file location flash:
 cnf-file perphone
 create cnf-files
!
```

Task 3, Step 6:

```
ephone-dn  1  dual-line
 number 5553001
!
ephone 1
 mac-address 0024.C445.4B48
 type 7965
 button 1:1
!
telephony-service
create cnf-files
!
```

Lab 4-1 Answer Key: Implementing Digit Manipulation

The HQ gateway configuration should be like the following (this is the pod 1 configuration):

Task 2, Step 1

```
ephone-dn 1
 number 2001
!
ephone-dn 2
 number 2002
!
ephone-dn 3
 number 2011
!
ephone-dn 4
 number 2012
!
ephone-dn 5
 number 2003
!
ephone 1
 restart
!
ephone 2
 restart
!
```

Task 3, Step 1

```
voice translation-rule 1
 rule 1 /^2/ /5552/
!
voice translation-profile pstn-out
 translate calling 1
!
voice-port 0/0/0:15
  translation-profile outgoing pstn-out
!
```

Task 4, Step 1

```
voice translation-rule 2
 rule 1 /.*/ /0&/ type subscriber subscriber
 rule 2 /.*/ /00&/ type national national
 rule 3 /.*/ /000&/ type international international
!
voice translation-rule 3
 rule 1 /^5552/ /2/
 rule 2 /^5115552/ /2/
 rule 3 /^555115552/ /2/
!
voice translation-profile pstn-in
 translate calling 2
 translate called 3
!
voice-port 0/0/0:15
 translation-profile incoming pstn-in
!
```

Task 5, Step 1:

```
voice translation-rule 4
 rule 1 /^2/ /8102/
!
voice translation-profile intersite-out
 translate calling 4
!
dial-peer voice 3000 voip
 destination-pattern 820....
 translation-profile outgoing intersite-out
!
voice translation-rule 5
 rule 1 /^8102/ /2/
!
voice translation-profile intersite-in
 translate called 5
!
dial-peer voice 2 voip
 incoming called-number .
 translation-profile incoming intersite-in
```



```
 voice-class codec 1
!
```

The BR gateway configuration should be like the following (this is the pod 1 configuration):

Task 1, Step 3

```
voice translation-rule 1
 rule 1 /^9011.*/ /&/ type any unknown
!
voice translation-profile fix-011
 translate called 1
!
```

Task 1, Step 4

```
voice-port 0/0/0:15
 translation-profile outgoing fix-011
!
```

Task 2, Step 2:

```
ephone-dn 1
 number 3001
!
ephone 1
 restart
!
```

Task 3, Step 2:

```
voice translation-rule 2
 rule 1 /^3/ /5553/
!
voice translation-profile pstn-out
 translate called 1
 translate calling 2
!
voice-port 0/0/0:15
 translation-profile outgoing pstn-out
!
```

Task 4, Step 2:

```
voice translation-rule 3
 rule 1 /.*/ /9&/ type subscriber subscriber
 rule 2 /.*/ /91&/ type national national
 rule 3 /.*/ /9011&/ type international international
!
voice translation-rule 4
 rule 1 /^5553/ /3/
 rule 2 /^5215553/ /3/
 rule 3 /^665215553/ /3/
!
voice translation-profile pstn-in
 translate calling 3
```

```
 translate called 4
!
voice-port 0/0/0:15
 translation-profile incoming pstn-in
!
```

Task 5, Step 2:

```
voice translation-rule 5
 rule 1 /^3/ /8203/
!
voice translation-profile intersite-out
 translate calling 5
!
dial-peer voice 2000 voip
 destination-pattern 810....
 translation-profile outgoing intersite-out
!
voice translation-rule 6
 rule 1 /^8203/ /3/
!
voice translation-profile intersite-in
 translate called 6
!
dial-peer voice 2 voip
 incoming called-number .
 translation-profile incoming intersite-in
 voice-class codec 1
!
```

Lab 4-2 Answer Key: Implementing Path Selection

The HQ gateway configuration should be like the following (this is the pod 1 configuration):

Task 1, Step 1:

```
dial-peer voice 3005 pots
 preference 2
 destination-pattern 820....
 port 0/0/0:15
!
```

Task 1, Step 2:

```
voice translation-rule 6
 rule 1 /^2/ /555115552/ type any international
!
voice translation-rule 7
 rule 1 /^820/ /0066521555/
!
voice translation-profile pstn-backup
 translate calling 6
 translate called 7
!
```


Task 1, Step 3:

```
dial-peer voice 3005 pots
 translation-profile outgoing pstn-backup
!
```

Task 3, Step 1:

```
dial-peer voice 6600 voip
 destination-pattern 00066T
 session target ipv4:10.1.250.102
 voice-class codec 1
!
```

Task 3, Step 2:

```
voice translation-rule 8
 rule 1 /^00066/ /91/
!
voice translation-profile teho-out
translate calling 6
translate called 8
!
dial-peer voice 6600 voip
 translation-profile outgoing teho-out
!
```

The BR gateway configuration should be like the following (this is the pod 1 configuration):

Task 2, Step 1:

```
dial-peer voice 2005 pots
 preference 2
 destination-pattern 810....
 port 0/0/0:15
!
```

Task 2, Step 2:

```
voice translation-rule 7
 rule 1 /^3/ /665215553/ type any international
!
voice translation-rule 8
 rule 1 /^810/ /01155511555/
!
voice translation-profile pstn-backup
 translate calling 7
 translate called 8
!
```

Task 2, Step 3:

```
dial-peer voice 2005 pots
 translation-profile outgoing pstn-backup
!
```

Task 2, Step 4:

```
voice translation-rule 1
```

```
 rule 1 /^9011.*/ /&/ type any unknown
 rule 2 /^011.*/ /&/ type any unknown
!
```

Task 4, Step 1:

```
dial-peer voice 5500 voip
 destination-pattern 901155T
 session target ipv4:10.1.250.101
 voice-class codec 1
!
```

Task 4, Step 2:

```
voice translation-rule 9
 rule 1 /^901155/ /00/
!
voice translation-profile teho-out
translate calling 7
translate called 9
!
dial-peer voice 5500 voip
 translation-profile outgoing teho-out
!
```

Lab 4-3 Answer Key: Implementing Calling Privileges

The HQ gateway configuration should include these commands:

Task 1, Step 2:

```
dial-peer cor custom
 name emergency
 name local
 name national
 name intl
!
```

Task 1, Step 3:

```
dial-peer cor list emergency-out
 member emergency
!
dial-peer cor list local-out
 member local
!
dial-peer cor list national-out
 member national
!
dial-peer cor list intl-out
 member intl
!
dial-peer cor list lobby
 member emergency
!
```



```
dial-peer cor list sales
 member emergency
 member local
 member national
!
```

Task 1, Step 4:

```
dial-peer voice 7 pots
 corlist outgoing local-out
!
dial-peer voice 10 pots
 corlist outgoing national-out
!
dial-peer voice 9011 pots
 corlist outgoing intl-out
!
dial-peer voice 112 pots
 corlist outgoing emergency-out
!
dial-peer voice 1120 pots
 corlist outgoing emergency-out
!
ephone-dn 3
 number 2011
 corlist incoming lobby
!
ephone-dn 4
 number 2012
 corlist incoming sales
!
```

The BR gateway configuration should include these commands:

Task 2, Step 2:

```
dial-peer cor custom
 name emergency
 name local
 name block
!
```

Task 2, Step 3:

```
dial-peer cor list emergency-out
 member emergency
!
dial-peer cor list local-out
 member local
!
dial-peer cor list employee
 member emergency
 member local
!
```

```
dial-peer cor list block
 member block
!
```

Task 2, Step 4:

```
dial-peer voice 7 pots
 corlist outgoing local-out
!
dial-peer voice 10 pots
 corlist outgoing block
!
dial-peer voice 9011 pots
 corlist outgoing block
!
dial-peer voice 911 pots
 corlist outgoing emergency-out
!
dial-peer voice 9911 pots
 corlist outgoing emergency-out
!
ephone-dn 1
 number 3001
 corlist incoming employee
!
```

Lab 5-1 Answer Key: Implementing Gatekeepers

The HQ gateway configuration should be like the following (this is the pod 1 configuration):

Task 1, Steps 1 to 3:

```
gatekeeper
 zone local HQ cisco.com 10.1.250.101
 zone local BR cisco.com
 zone prefix HQ 2...
 zone prefix BR 3...
 no shutdown
!
```

Task 2, Step 4:

```
interface Loopback0
 ip address 10.1.250.101 255.255.255.255
 h323-gateway voip interface
 h323-gateway voip id HQ ipaddr 10.1.250.101
 h323-gateway voip h323-id HQ-gw
 h323-gateway voip bind srcaddr 10.1.250.101
!
ephone-dn 4
 number 2012 no-reg
!
gateway
!
```


Task 2, Step 5:

```
dial-peer voice 3002 voip
 destination-pattern 3...
 session target ras
 voice-class codec 1
!
```

Task 3, Step 2:

```
gatekeeper
 bandwidth interzone zone HQ 31
 bandwidth interzone zone BR 31
!
```

The BR gateway configuration should be like the following (this is the pod 1 configuration):

Task 2, Step 2:

```
interface Loopback0
 ip address 10.1.250.102 255.255.255.255
 h323-gateway voip interface
 h323-gateway voip id BR ipaddr 10.1.250.101 1719
 h323-gateway voip bind srcaddr 10.1.250.102
!
gateway
!
```

Task 2, Step 3:

```
dial-peer voice 2002 voip
 destination-pattern 2...
 session target ras
 voice-class codec 1
!
```

Lab 5-2 Answer Key: Implementing Cisco Unified Border Element

The HQ gateway configuration should be like the following (this is the pod 1 configuration):

Task 1, Step 1:

```
dial-peer voice 555 voip
 destination-pattern 5552...
 session target ipv4:10.1.250.102
 codec g711ulaw
!
```

Task 1, Step 2:

```
dial-peer voice 2 voip
 translation-profile incoming intersite-in
 incoming called-number .
 no voice-class codec
 codec g711ulaw
!
```

Task 1, Step 4:

```
voice translation-profile intersite-in
 translate called 5
!
voice translation-rule 5
 rule 1 /^8102/ /2/
 rule 2 /^5552/ /2/
!
```

Task 2, Step 1:

```
dial-peer voice 555 voip
 voice-class codec 1
!
```

Task 2, Step 2:

```
dial-peer voice 2 voip
 voice-class codec 1
!
```

Task 3, Step 1:

```
dial-peer voice 555 voip
 session protocol sipv2
!
```

The BR gateway configuration should be like the following (this is the pod 1 configuration):

Task 1, Step 2:

```
dial-peer voice 2 voip
 translation-profile incoming intersite-in
 incoming called-number .
 no voice-class codec
 codec g711ulaw
!
```

Task 1, Step 3:

```
dial-peer voice 555 voip
 destination-pattern 5552...
 session target ipv4:10.1.250.101
 codec g711ulaw
!
```

Task 1, Step 6:

```
voice service voip
 allow-connections h323 to h323
!
```

Task 2, Step 3:

```
dial-peer voice 2 voip
 codec transparent
!
dial-peer voice 555 voip
 codec transparent
```



```
!
```

Task 3, Step 2:

```
dial-peer voice 2 voip
 session protocol sipv2
!
```

Task 3, Step 3:

```
voice service voip
 allow-connections sip to h323
!
```

Task 3, Step 5:

```
voice service voip
 media flow-around
!
```

Lab 6-1 Answer Key: Implementing QoS Using Cisco AutoQoS and Manual Configuration

You should configure your HQ gateway with these commands (this is the pod 1 configuration):

Task 2, Step 1:

```
interface Serial0/1/0.121 point-to-point
 description to BR-1
 bandwidth 64
!
```

Task 2, Step 3:

```
interface Serial0/1/0.121
 frame-relay interface-dlci 121
  auto qos voip trust
!
```

Task 3, Step 3:

```
policy-map AutoQoS-Policy-Trust
 class AutoQoS-VoIP-RTP-Trust
    priority percent 38
!
```

The HQ gateway configuration should be like the following (this is the pod 1 configuration):

```
class-map match-any AutoQoS-VoIP-RTP-Trust
 match ip dscp ef
class-map match-any AutoQoS-VoIP-Control-Trust
 match ip dscp cs3
 match ip dscp af31
!
policy-map AutoQoS-Policy-Trust
 class AutoQoS-VoIP-RTP-Trust
    priority percent 38
 class AutoQoS-VoIP-Control-Trust
    bandwidth percent 5
```

```
 class class-default
    fair-queue
!
interface Serial0/1/0
 no ip address
 encapsulation frame-relay
 no keepalive
 frame-relay traffic-shaping
!
interface Serial0/1/0.121 point-to-point
 description to BR-1
 bandwidth 64
 ip address 10.1.6.101 255.255.255.0
 frame-relay interface-dlci 121
  class AutoQoS-FR-Se0/1/0-121
  auto qos voip trust
 frame-relay ip rtp header-compression
!
map-class frame-relay AutoQoS-FR-Se0/1/0-121
 frame-relay cir 64000
 frame-relay bc 640
 frame-relay be 0
 frame-relay mincir 64000
 frame-relay fragment 80
 service-policy output AutoQoS-Policy-Trust
!
```

You should configure your BR gateway with these commands (this is the pod 1 configuration):

Task 2, Step 2:

```
interface Serial0/1/0.111 point-to-point
 description to HQ-1
 bandwidth 64
!
```

Task 2, Step 4:

```
interface Serial0/1/0.111
 frame-relay interface-dlci 111
  auto qos voip trust
!
```

Task 3, Step 3:

```
policy-map AutoQoS-Policy-Trust
 class AutoQoS-VoIP-RTP-Trust
    priority percent 38
!
```

The BR gateway configuration should be like the following (this is the pod 1 configuration):

```
class-map match-any AutoQoS-VoIP-RTP-Trust
 match ip dscp ef
class-map match-any AutoQoS-VoIP-Control-Trust
```



```
 match ip dscp cs3
 match ip dscp af31
!
policy-map AutoQoS-Policy-Trust
 class AutoQoS-VoIP-RTP-Trust
    priority percent 38
 class AutoQoS-VoIP-Control-Trust
    bandwidth percent 5
 class class-default
    fair-queue
!
interface Serial0/1/0
 no ip address
 encapsulation frame-relay
 no keepalive
 frame-relay traffic-shaping
!
interface Serial0/1/0.111 point-to-point
 description to HQ-1
 bandwidth 64
 ip address 10.1.6.102 255.255.255.0
 frame-relay interface-dlci 111
  class AutoQoS-FR-Se0/1/0-111
  auto qos voip trust
 frame-relay ip rtp header-compression
!
map-class frame-relay AutoQoS-FR-Se0/1/0-111
 frame-relay cir 64000
 frame-relay bc 640
 frame-relay be 0
 frame-relay mincir 64000
 frame-relay fragment 80
 service-policy output AutoQoS-Policy-Trust
!
```

