

CCIE Collaboration Lab Walkthrough

with

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(Collaboration and R/S)**

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CCIE COLLABORATION LAB WALKTHROUGH

Topology

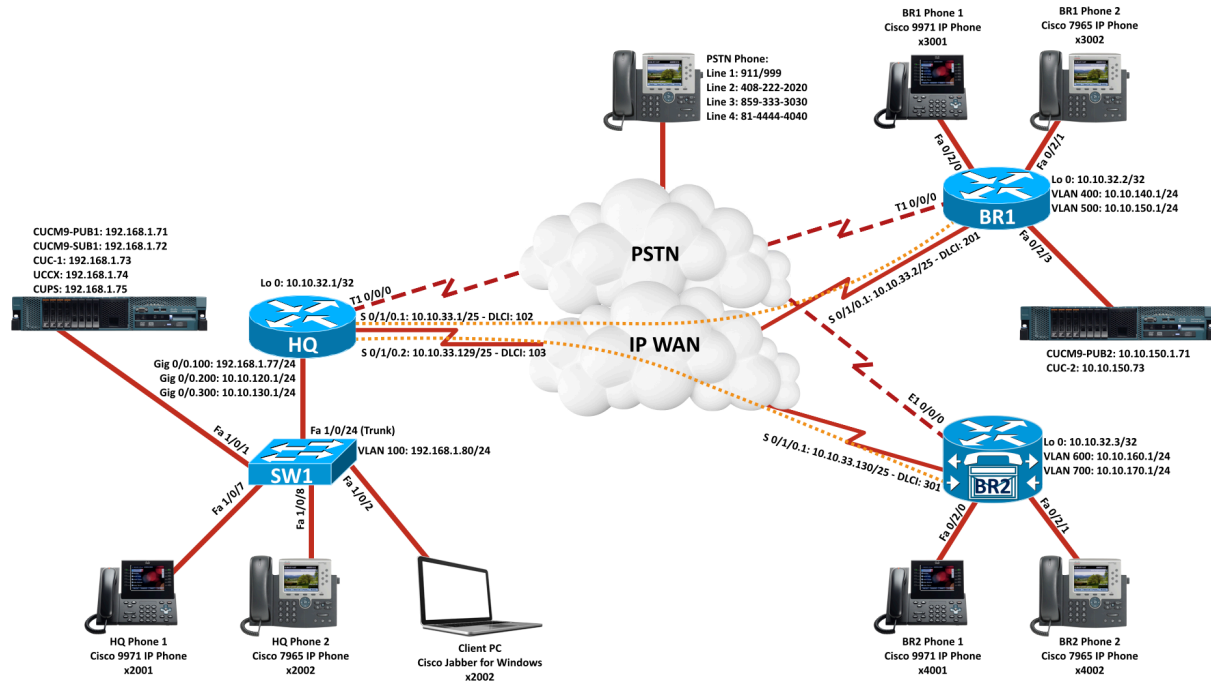


Table 1 – Server IP Addresses

Location	IP Address	Location
CUCM Publisher 1	192.168.1.71	HQ
CUCM Subscriber	192.168.1.72	HQ
Cisco Unity Connection	192.168.1.73	HQ
Cisco Unified Contact Center Express	192.168.1.74	HQ
Cisco Unified Presence Server	192.168.1.75	HQ
CUCM Publisher 2	10.10.150.71	BR1

Table 2 – Device IP Addresses

Location	IP Address
HQ	
- GigabitEthernet 0/0.100 (VLAN 100 HQ Server)	192.168.1.77 /24
- GigabitEthernet 0/0.200 (VLAN 200 HQ Voice)	10.10.120.1 /24
- GigabitEthernet 0/0.300 (VLAN 300 HQ Data)	10.10.130.1 /24
- Serial 0/1/0.1 (PVC to BR1)	10.10.33.1 /25

- Serial 0/1/0.2 (PVC to BR2)	10.10.33.129 /25
- Loopback 0	10.10.32.1 /32
SW1	
- VLAN 100	192.168.1.80 /24
BR1	
- VLAN 400 (BR1 Voice)	10.10.140.1 /24
- VLAN 500 (BR1 Data)	10.10.150.1 /24
- Serial 0/1/0.1 (PVC to HQ)	10.10.33.2 /25
- Loopback 0	10.10.32.2 /32
BR2	
- VLAN 600 (BR2 Voice)	10.10.160.1 /24
- VLAN 700 (BR2 Data)	10.10.170.1 /24
- Serial 0/1/0.1 (PVC to HQ)	10.10.33.130 /25
- Loopback 0	10.10.32.3 /32
PSTN Router	
- FastEthernet 0/0 (to PSTN IP Phone)	10.1.200.229 /24
- FastEthernet 0/1 (VLAN 100 HQ Server)	192.168.1.78 /24

Table 3 – Voice and Data VLAN IDs

	DATA	PHONES	SERVERS
HQ	300	200	100
BR1	500	400	N/A
BR2	700	600	N/A

Table 4 – HQ Phone Types and Directory Numbers

HQ	Type	Protocol	DNs
Phone 1	9971	SIP	2001
Phone 2	7965	SCCP	2002

Table 5 – BR1 Phone Types and Directory Numbers

BR1	Type	Protocol	DNs
Phone 1	9971	SIP	3001
Phone 2	7965	SCCP	3002

Table 6 – BR2 Phone Types and Directory Numbers

BR2	Type	Protocol	DNs
Phone 1	9971	SIP	4001
Phone 2	7965	SCCP	4002

Table 7 – PSTN Information

Router	Signaling	Framing	Line Coding	ISDN Switch Type	Active Timeslots
HQ	T1 PRI	ESF	B8ZS	Primary-NI	1-3
BR1	T1 PRI	ESF	B8ZS	Primary-NI	1-3
BR2	E1 PRI	CRC4	HDB3	Primary-Net5	1-3

Table 8 – PSTN Phone Numbers

Line	Number	Country Code	Local to Site
1	911	N/A	HQ and BR1
2	999	N/A	BR2
3	408-222-2020	1 (US)	HQ
4	859-333-3030	1 (US)	BR1
5	4444-4040	81 (Japan)	BR2
6	44-5555-6666	55 (Brazil)	Backbone

Table 9 – Usernames and Passwords

Username	Password	Description
administrator	kwtrain01	Administrative user for CUCM, CUC, UCCX, and CUPS
uccxadmin	kwtrain01	CTI user for UCCX

(1) VLANs, IP Address Assignment, and Network Time Protocol (NTP) Configuration

1.1 VLANs (2 Points)

Configure switch ports connecting to IP phones (at all sites) as multi-VLAN access ports, based on the VLAN information provided in Table 3.

1.2 IP Address Assignment (3 Points)

- The CUCM publisher server at HQ should act as the DHCP server for IP phones at HQ.
- These IP phones at HQ should receive IP addresses in the range **10.10.120.10 /24 – 10.10.120.20 /24**.
- The BR1 router should act as the DHCP server for IP phones at BR1.
- IP phones at BR1 should receive IP addresses in the range **10.10.140.10 /24 – 10.10.140.20 /24**.
- The BR2 router should act as the DHCP server for IP phones at BR2.
- BR2 Phone 1** should always receive an IP address of **10.10.160.10 /24**, and **BR2 Phone 2** should always receive an IP address of **10.10.160.11 /24**.

NOTE: You do not need to configure DHCP to provide IP addresses to any PCs that might exist in any of the Data VLANs.

1.3 Network Time Protocol (NTP) Configuration (2 Points)

- Configure the HQ router to obtain time from an NTP server on the backbone with an IP address of **192.168.1.78**. The HQ router should then act as the NTP server for the BR1 and BR2 routers, in addition to the CUCM Publishers.
- All IP Phones should display the appropriate time for their time zone. The following table lists time zone information for each of your sites.

Location	Time Zone
HQ	Pacific UTC -8 (DST)
BR1	Eastern UTC -5 (DST)
BR2	Tokyo UTC +9 (No DST)

(2) Cisco Unified Communications Manager (CUCM) and Cisco Unified Communications Manager Express (CUCME) Base Configuration

2.1 CUCM SIP (Model 9971) IP Phones (3 Points)

- Register HQ Phone 1 to the HQ cluster and BR1 Phone 1 to the BR1 CUCM Publisher. Assign extension numbers as specified in Tables 4 and 5.
- The display on HQ Phone 1 and BR1 Phone 1 should be formatted as shown below.

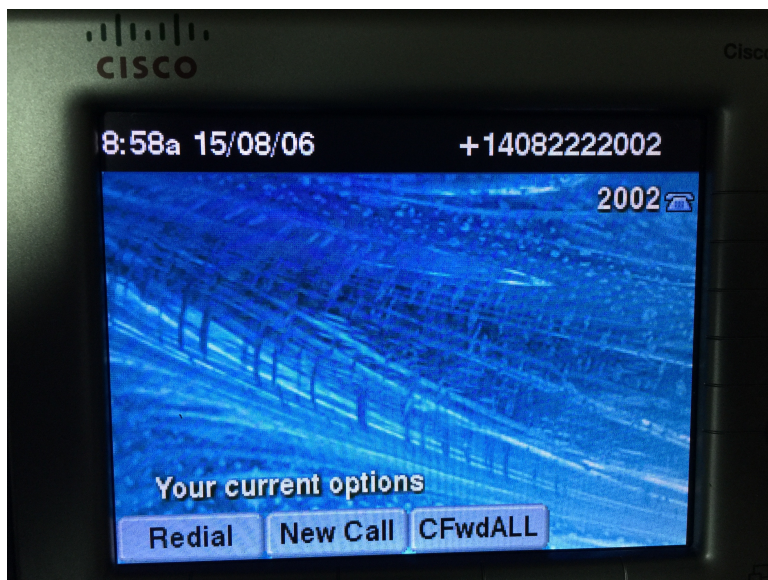


- IP phones should display a globalized dialing number in the right corner (i.e. HQ Phone 1 should display +14082222001, and BR1 Phone 1 should display +18593333001).

- Ensure that HQ Phone 1 is configured with a redundant TFTP server, such that HQ's CUCM Subscriber (192.168.1.72) is the primary TFTP server, and HQ's CUCM Publisher (192.168.1.71) is the backup TFTP server.
- IP phones should be able to call one another using 4-digit dialing. Caller ID information (both name and number) from the calling IP phone should be displayed on the called IP phone. You can use your own naming convention for the IP phones.

2.2 CUCM SCCP (Model 7965) IP Phones (3 Points)

- Register HQ Phone 2 to the HQ cluster and BR1 Phone 2 to the BR1 CUCM Publisher. Assign extension numbers as specified in Tables 4 and 5.
- The display on HQ Phone 2 and BR1 Phone 2 should be formatted as shown below.



- IP phones should display a globalized dialing number in the right corner (i.e. HQ Phone 2 should display +14082222002, and BR1 Phone should display +18593333002).
- Ensure that HQ Phone 1 is configured with a redundant TFTP server, such that HQ's CUCM Subscriber (192.168.1.72) is the primary TFTP server, and HQ's CUCM Publisher (192.168.1.71) is the backup TFTP server.
- IP phones should be able to call one another using 4-digit dialing. Caller ID information (both name and number) from the calling IP phone should be displayed on the called IP phone. You can use your own naming convention for the IP phones.

2.3 CUCM IP Phone Customization (3 Points)

- Calls between IP phones in the HQ site should use the G.711 ulaw codec.
- Calls between IP phones in the BR1 site should use the G.711 alaw codec.

2.4 CUCME SIP IP Phone Registration (3 Points)

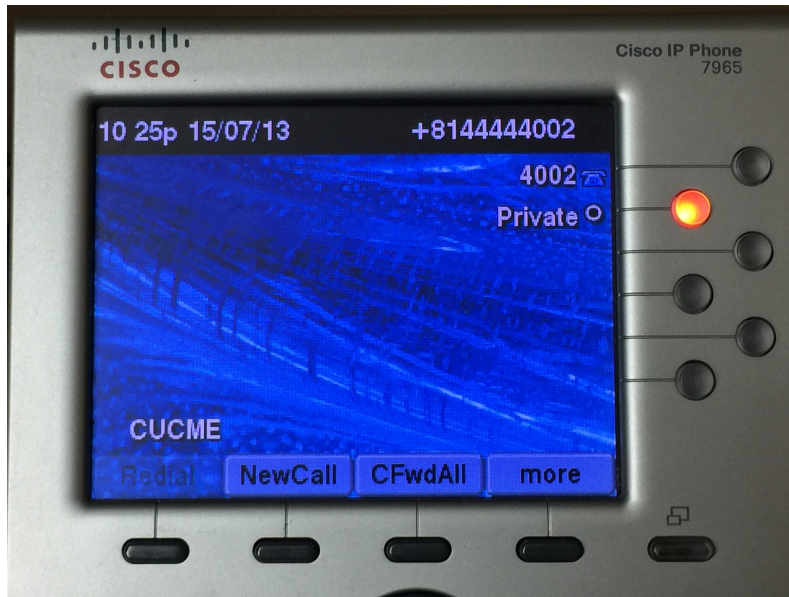
- BR2 Phone 1 should register as a SIP phone with router BR2 (acting as a CUCME router) and be assigned the directory number specified in Table 6.
- Other than the specific time, date, background, and phone number, configure the display of BR2 Phone 1 to look like the display shown below.



BR2 Phone 1's Display

2.5 CUCME SCCP IP Phone Registration (3 Points)

- BR2 Phone 2 should register as an SCCP phone with router BR2 (acting as a CUCME router) and be assigned the directory number specified in Table 6.
- Other than the specific time, date, background, and phone number, configure the display of BR2 Phone 2 to look like the display shown below.



BR2 Phone 2's Display

2.6 CUCME IP Phone Customization (3 Points)

On BR2 Phone 1 (i.e. the Cisco 9971 IP Phone), you must enter a password of **Cisco** to access the **Administrator Settings** screen. Remove this requirement to enter a password to access this screen.

(3) VoIP Gateway Configuration

Use the following parameters when configuring T1 and E1 interfaces:

- **For the T1 Controllers:**
 - **Switch Type:** primary-ni
 - **Framing:** B8ZS
 - **Line Code:** ESF
- **For the E1 Controller:**
 - **Switch Type:** primary-net5
 - **Framing:** CRC4
 - **Line Code:** HDB3
- Take clocking for Layer 1 from Network side.
- Your PRI circuit Layer 2 should be user side.
- Calling names should be sent to the PSTN.

3.1 HQ Cisco IOS H.323 T1-PRI Gateway (2 Points)

- Configure CUCM to point to HQ router controller T1 0/0/0 as a Cisco IOS H.323 T1 PRI gateway. Make sure that all inbound and output H.323 traffic is sourced from the voice VLAN's subinterface 10.10.120.1. Use only 3 channels of the T1 PRI.
- The telco is sending 10-digits Direct-Inward-Dial (DID) for inbound PSTN calls. Test the inbound calls to HQ IP phones 408222XXXX, where XXXX is the extension range of HQ IP phones.
- Verify gateway functionality by making outgoing calls to the 911 emergency number. Calls made to this number should display 10-digit caller ID as 408222XXXX.
- There is no need to test 9911 calling.

3.2 BR1 Cisco IOS MGCP T1-PRI Gateway (2 Points)

- Configure CUCM to register the BR1 router controller T1 0/0/0 as a Cisco IOS MGCP T1 PRI gateway. Make sure that all inbound and outbound MGCP traffic is sourced from the Voice VLAN's interface. Use only 3 channels of the T1 PRI.
- The telco is sending 10-digits Direct-Inward-Dial (DID) for inbound PSTN calls. Test the inbound calls to BR1 IP phones 859333XXXX, where XXXX is the extension range for BR1 phones.
- Verify gateway functionality by making outgoing calls to the 911 emergency number. Calls made to this number should display 10-digit caller ID as 859333XXXX.
- There is no need to test 9911 calling.

3.3 BR2 CUCME H.323 Gateway (2 Points)

- Configure the BR2 router as a Cisco Unified Communications Manager Express (CUCME) gateway. Use only 3 channels of the E1 PRI.
- Make sure all inbound and outbound H.323 traffic is sourced from the Voice VLAN's interface.
- The Telco is sending 8-digits Direct-Inward-Dial (DID) for inbound PSTN calls. Test the inbound calls to BR2 IP phones 4444XXXX, where XXXX is the extension range of BR2 IP phones.
- Verify gateway functionality by making outgoing calls to the 999 emergency number. Calls made to this number should display 8-digit caller ID as 4444XXXX.

3.4 SIP Trunk (4 Points)

- Configure router **HQ** as a **CUBE**.
- The CUBE should be used for inter-site calls between HQ and BR1 (initiated from either the HQ or BR1 site).
- The **SIP** signaling protocol should be used for these calls.
- The **iLBC** codec should be used for these calls.
- Verify that an audio call can be established between HQ and BR1 by going to HQ Phone 2 and calling BR1 Phone 2. Then, go to BR1 Phone 2, and call HQ Phone 2.

3.5 Troubleshooting Video Call Between HQ and BR1 (4 Points)

- Place a call from HQ Phone 1 to BR1 Phone 1. Then, place a call from BR1 Phone 1 to HQ Phone 1. In both instances, notice that you are experiencing one-way video (unless you accounted for this issue in a prior task).
- Issue the **debug ccsip messages** command on the CUBE, and place a call from HQ Phone 1 to BR1 Phone 1.
- Copy the contents of the debug output to a Microsoft Notepad document.
- If you experienced one-way video in the way, identify (in the Microsoft Notepad document) why you experienced one-way video. Then, resolve the one-way video issue.
- However, if you accounted for this issue in a prior task, identify (in the Microsoft Notepad document) why are are experiencing two-way video.
- Save the Microsoft Notepad document to your computer's desktop.

3.6 SIP Gateway (2 Points)

- Configure the CUCM cluster at HQ and the CUCM9-PUB2 server at BR1 with SIP trunks pointing to the BR2 router.
- Configure the BR2 router as a SIP gateway that will communicate with the CUCM cluster at HQ and the CUCM9-PUB2 server at BR1.
- Both HQ and BR1 should be able to call IP phones at the BR2 site using 4-digit dialing.
- IP phones at the BR2 site should be able to call IP phones at either the HQ or BR1 site using 4-digit dialing.
- In addition to audio calls, video calls between Cisco 9971 IP Phones should be supported.

(4) Route Plan Configuration

4.1 HQ Call Routing (4 Points)

- HQ callers dial a 9 to get an outside line, and this leading digit should be stripped before sending the DNIS out to the PSTN.
- HQ callers should be able to dial 911, in order to reach emergency services. The ANI should be 10 digits in length and have a TON of **National**. The DNIS should have a TON of **Unknown**. Only the HQ gateway should be used for this call type.
- HQ callers should be able to call local numbers, by dialing a **9**, followed by seven digits. Note that the first of those seven digits (i.e. the office code) must be in the range of 2 – 9. The ANI should be 7 digits in length and have a TON of **Subscriber**. The DNIS should also have a TON of **Subscriber**. Only the HQ gateway should be used for this call type.
- HQ callers should be able to call long distance numbers, by dialing a 9, followed by a 1, followed by ten digits. Note that the first of those ten digits (i.e. the area code) must be in the range of 2 – 9. The office code must also be in the range of 2 – 9. The ANI should be 10 digits in length and have a TON of **National**. The DNIS should include the leading 1 and have a TON of **National**. Only the HQ gateway should be used for this call type.

- HQ callers should be able to call international numbers, by dialing a 9, followed by 011, followed by a variable length dial string. After dialing the variable length dial string, the caller should not have to wait more than three seconds before CUCM places the call. Alternately, the caller should be allowed to press the # button to indicate they have finished dialing. The ANI should be in E.164 format and have a TON of **International**. The DNIS should include the leading 011 and have a TON of **International**. Only the HQ gateway should be used for this call type.

4.2 BR1 Call Routing (4 Points)

- BR1 callers dial a 9 to get an outside line, and this leading digit should be stripped before sending the DNIS out to the PSTN.
- BR1 callers should be able to dial 911, in order to reach emergency services. The ANI should be 10 digits in length and have a TON of **National**. The DNIS should have a TON of **Unknown**. Only the BR1 gateway should be used for this call type.
- BR1 callers should be able to call local numbers, by dialing a 9, followed by seven digits. Note that the first of those seven digits (i.e. the office code) must be in the range of 2 – 9. The BR1 gateway is the preferred gateway for this call type. If the BR1 gateway is used, the ANI should be 7 digits in length and have a TON of **Subscriber**. The DNIS should also have a TON of **Subscriber**.
- BR1 callers should be able to call long distance numbers, by dialing a 9, followed by a 1, followed by ten digits. Note that the first of those ten digits (i.e. the area code) must be in the range of 2 – 9. The office code must also be in the range of 2 – 9. The ANI should be 10 digits in length and have a TON of **National**. The DNIS should not include the leading 1 and have TON of **National**. Only the BR1 gateway should be used for this call type.
- BR1 callers should be able to call international numbers, by dialing a 9, followed by 011, followed by a variable length dial string. After dialing the variable length dial string, the caller should not have to wait more than three seconds before CUCM places the call. Alternately, the caller should be allowed to press the # button to indicate they have finished dialing. The ANI should be in E.164 format and have a TON of **International**. The DNIS should not include the leading 011 and have TON of **International**. Only the BR1 gateway should be used for this call type.

4.3 BR2 Call Routing (3 Points)

- BR2 callers dial a 9 to get an outside line, and this leading digit should be stripped before sending the DNIS out to the PSTN.
- BR2 callers should be able to dial 999, in order to reach emergency services. The ANI should be 8 digits in length and have a TON of **Subscriber**. The DNIS should have a TON of **Unknown**.
- BR2 callers should be able to call local numbers, by dialing a 9, followed by eight digits, the first of which is in the range 1-9. The ANI should be 8 digits in length and have a TON of **Subscriber**. The DNIS should also have a TON of **Subscriber**.
- BR2 callers should be able to call international numbers, by dialing a 9, followed by 00, followed by a variable length dial string. The ANI should be in E.164 format and have a TON of **International**. The DNIS should not include the leading 00 and have TON of **International**.

4.4 Number Globalization and Localization (4 Points)

- When a call comes into HQ Phone 2 from the 408-222-2020 PSTN phone number, the ANI should appear as a seven-digit number on the IP phone's display. However, the number should appear in a fully globalized E.164 format when viewed in HQ Phone 1's call history.
- You should be able to select this E.164 number in HQ Phone 2's call history, and press the **Dial** softkey to place a call to the corresponding PSTN number. The HQ gateway is the preferred gateway for this call, with the BR1 gateway acting as a backup.
- If the **HQ gateway** is used, the **ANI** should be **7 digits** with a **TON** of **Subscriber**, and the **DNIS** should have a **TON** of **Subscriber**. However, if the **BR1 gateway** must be used (due to the HQ gateway not being available), the **ANI** should be **10 digits** with a **TON** of **National**, and the **DNIS** should have a **TON** of **National**.

4.5 URI Dialing (2 Points)

- **HQ Phone 1** has a URI ID of **hquser1@cisco.local**
- Add a speed dial button 6 of HQ Phone 2 that will call HQ Phone 1 using its URI ID.

(5) Media Resource Configuration

5.1 Video Conference Bridge (3 Points)

- Configure HQ as a video conferencing bridge that registers with the HQ CUCM cluster.
- You should be able to setup a three-way video conference call initiated from HQ Phone 1 with conference attendees of BR1 Phone 1 and BR2 Phone 1.

5.2 Ad-Hoc Conferencing (3 Points)

- Configure BR2 to support a single ad-hoc conference, using a hardware conference bridge.

(6) Quality of Service (QoS) Configuration

6.1 Cisco Catalyst Switch QoS (3 Points)

Configure QoS settings on switch SW1 such that:

- RTP media (i.e. both voice and video) with appropriate CoS values should be placed in the egress priority queues of interfaces connecting to IP phones and the trunk interface connecting to the HQ router.
- The switch should map the signaling CoS value (CoS=3) and RTP media CoS values (Video CoS = 4 and Voice CoS = 5) to appropriate DSCP values.
- Frames with a CoS value of 6 and 7 should be placed in Queue 2.

- Frames with a CoS value of 3 should be placed in Queue 3.
- Frames with a CoS value of 0, 1, or 2 should be placed in Queue 4.
- However, for ports connecting to IP phones (and only ports connecting to IP phones), frames with a CoS value of 0 should be dropped after Queue 4 reaches 25 percent capacity, and frames with a CoS value of 1 should be dropped after Queue 4 reaches 50 percent capacity.
- For the interface connecting switch SW1 to router HQ (i.e. the trunking interface), Queue 1 should be limited to 12.5 percent of the interface's bandwidth, while Queues 2, 3, and 4 share 40, 20, and 40 percent respectively of the remaining interface bandwidth.

6.2 Router QoS (3 Points)

The link between router HQ and router BR2 is 1.544 Mbps. Configure the HQ and BR2 routers to meet the following QoS configuration requirements:

- Configure Multilink PPP (MLP) as a Link Fragmentation and Interleaving (LFI) mechanism over this link.
- Do not configure RTP Header Compression (cRTP).
- Configure Low Latency Queuing (LLQ) such that the expedite queue has enough bandwidth to accommodate 5 simultaneous iLBC voice calls using a packetization rate of 50 packets per second (PPS).

(7) Unified Messaging Configuration

7.1 Cisco Unity Connection (CUC) SCCP Integration Configuration (3 Points)

Integrate the HQ Cisco Unity Connection (CUC) server with the HQ Cisco Unified Communications Manager (CUCM) cluster using the following parameters:

- Integration Type: **SCCP**
- AXL Username/Password: **administrator/ kwtrain01**
- Voicemail Pilot DN: **2500**
- Voicemail Port DNs: **2501** and **2502**
- MWI On DN: **2510**
- MWI Off DN: **2511**

After CUC has been successfully integrated with CUCM, create voice mailboxes for the following users:

- hqphone1
- hqphone2

Each user should have a PIN of **12345**.

Each user's phone should divert to voicemail if the user is in a call, or if the user does not answer within **10 seconds**.

Message waiting indicator (MWI) lights should function for all users.

7.2 Cisco Unity Connection (CUC) SIP Integration Configuration (3 Points)

Integrate the BR1 Cisco Unity Connection (CUC) server with the BR1 Cisco Unified Communications Manager (CUCM) cluster using the following parameters:

- Integration Type: **SIP**
- AXL Username/Password: **administrator / kwtrain01**
- Voicemail Pilot DN: **3500**
- Two voicemail ports

After CUC has been successfully integrated with CUCM, create voice mailboxes for the following users:

- br1phone1
- br1phone2

Each user should have a PIN of **12345**.

Each user's phone should divert to voicemail if the user is in a call, or if the user does not answer within **10 seconds**.

Message waiting indicator (MWI) lights should function for all users.

7.3 CUC Customization (2 Points)

When a caller is listening to the voicemail greeting of hqphone1, they should be allowed to press **8** to be diverted to the voicemail box of **hqphone2**.

7.4 Cisco Unity Express (CUE) Module Initialization (2 Points)

Set the Cisco Unity Express (CUE) module in BR2 to factory default settings, if it is not already set to factory defaults. Initialize the CUE module using the following parameters:

- CUE Module's IP Address: **10.10.160.2/24**
- CUE Module's Hostname: **CUE**
- CUE Domain Name: **kwtrain.com**
- NTP Server's IP Address: **192.168.1.78**
- Administrative Username: **administrator/cisco**

7.5 Integrating CUE with CUCME (3 Points)

The voicemail system should be accessible using the following parameters:

- Voicemail DN: **4500**
- MWI On DN: **4510**
- MWI Off DN: **4511**

Integrate CUE with Cisco Communications Manager Express (CUCME), and create a voice mailbox for the following users:

- br2phone1
- br2phone2

The PIN for each user should be **12345**.

Each user's phone should divert to voicemail if the user is on a call, or if the user does not answer within 10 seconds.

MWI indicators should function for all users.

7.6 CUE Customization (2 Points)

Configure CUE such that when a caller dials **4550**, the message they hear begins with:

"Congratulations! You have successfully completed this task."

(8) Cisco Unified Contact Center Express (UCCX) Configuration

8.1 UCCX Integration (4 Points)

UCCX should be integrated with the CUCM using the following parameters:

- UCCX Server's Administrative Username / Password: **administrator / kwtrain01**
- UCCX Server's CTI Username / Password: **uccxadmin / kwtrain01**
- UCCX Server's RM Username / Password: **rm / kwtrain01**
- UCCX Server's Application Trigger (CTI Route Point): **2700**
- UCCX Server's CTI Ports: **2701, 2702**
- UCCX Server's Contact Service Queue (CSQ): **CSQ**
- IP Phone Agents: **hqphone1, hqphone2**
- CUCM IP XML Phone Services: **IP Phone Agent**

The IP phone agents should be able to log into the IP Phone Agent service without specifying their login credentials.

8.2 UCCX Scripting (4 Points)

Modify the UCCX's **icd.aef** script to meet the following requirements:

- The script should be assigned to an application named **CALLCENTER**.
- The HQ PSTN phone should be able to dial **222-2700** in order to reach the CALLCENTER application.
- If no agents are available, the caller should be queued.
- When the caller is placed in the queue, they should hear the following message:
"Your estimated wait time is x minutes," where **x** is the caller's estimated wait time, in minutes.

(9) IM and Presence Configuration

9.1 IM and Presence Integration with CUCM (4 Points)

- Integrate the HQ CUCM cluster with the HQ IM and Presence Server.
- The IM and Presence Server should belong to the **cisco.local** domain.
- The integration should support the exchange of presence information between the HQ CUCM cluster and the IM and Presence server.

9.2 Cisco Jabber for Windows (3 Points)

- Configure the Cisco Jabber for Windows client on the HQ Client PC using the existing username of **hqphone2**.
- Place a call from the HQ Jabber client to HQ Phone 1 (i.e. the Cisco 9971 IP Phone at HQ). The HQ Jabber client should see video from HQ Phone 1.

SOLUTION

Module 1: Mapping Out Your Day

Create a chart that lists the tasks to be performed on each device.

SW1	HQ	BR1	BR2
1.1 VLANs 6.1 QoS	1.1 VLANs 1.2 DHCP 1.3 NTP 3.1 HQ H.323 GW 3.4 SIP Trunk 3.5 Troubleshooting Video Call (wait) 4.1 HQ Call Routing 5.1 Video Conference Bridge 6.2 QoS	1.1 VLANs 1.2 DHCP 1.3 NTP 3.2 BR1 MGCP GW (wait)	1.1 VLANs 1.2 DHCP 1.3 NTP 2.4 CUCME SIP Phone 2.5 CUCME SCCP Phone 2.6 Phone Customization 3.3 BR2 CUCME H.323 GW 3.6 SIP GW 4.3 BR2 Call Routing 5.2 Ad-Hoc Conferencing 6.2 QoS 7.4 CUE Installation 7.5 CUE Integration with CUCME 7.6 CUE Customization
CUCM		CUC	CUE
1.2 DHCP 1.3 NTP 2.1 CUCM SIP Phones 2.2 CUCM SCCP Phones 2.3 Phone Customization 3.1 HQ H.323 GW 3.2 BR1 MGCP GW 3.4 SIP Trunk 3.5 Troubleshooting Video Call 3.6 SIP GW 4.1 HQ Call Routing 4.2 BR1 Call Routing 4.4 Number Globalization/Localization 4.5 URI Dialing 5.1 Video Conference Bridge	7.1 CUC SCCP Integration 7.2 CUC SIP Integration 8.1 UCCX Integration 9.1 IM and Presence Integration with CUCM	7.1 CUC SCCP Integration 7.2 CUC SIP Integration 7.3 CUC Customization 8.2 UCCX Scripting	7.4 CUE Installation 7.5 CUE Integration with CUCME 7.6 CUE Customization
		UCCX	IM & Presence
		8.1 UCCX Integration 8.2 UCCX Scripting	9.1 IM and Presence Integration with CUCM 9.2 Cisco Jabber for Windows

Backup configurations on SW1, HQ, BR1, and BR2.

SW1:

```
SW1#copy running-config BACKUP
```

```
Destination filename [BACKUP]?
```

```
4173 bytes copied in 1.544 secs (2703 bytes/sec)
```

```
SW1#
```

REPEAT ON ROUTERS HQ, BR1, and BR2.

Create a list of server IP addresses, IP addresses of voice VLAN interfaces, and IP addresses of router loopback interfaces.

```
CUCM9-PUB1: 192.168.1.71
CUCM9-SUB1: 192.168.1.72
CUCM9-PUB2: 10.10.150.71
HQ Gig 0/0.200: 10.10.120.1
```

```
HQ Lo 0: 10.10.32.1
BR1 VLAN 400: 10.10.140.1
BR1 Lo 0: 10.10.32.2
BR2 VLAN 600: 10.10.160.1
BR2 Lo 0: 10.10.32.3
```

Enable all services on all CUCM servers, except for DHCP.

CUCM9- PUB1:

Navigation > Cisco Unified Serviceability > Go > (Login) > Tools > Service Activation > Check All Services > (Uncheck Cisco DHCP Monitor Service) > Save > OK

The screenshot shows the Cisco Unified Serviceability web interface. The top navigation bar includes 'Navigation', 'Cisco Unified Serviceability', and a 'Go' button. Below the navigation bar, there's a 'Service Activation' section with buttons for 'Save', 'Set to Default', and 'Refresh'. The 'Status' section shows 'Ready'. The 'Select Server' section has a dropdown menu set to 'CUCM-PUB1' and a 'Go' button. Below this, there's a table of services with checkboxes and their activation status.

Service Name	Activation Status
<input checked="" type="checkbox"/> Cisco CallManager	Activated
<input checked="" type="checkbox"/> Cisco Messaging Interface	Activated
<input checked="" type="checkbox"/> Cisco Unified Mobile Voice Access Service	Activated
<input checked="" type="checkbox"/> Cisco IP Voice Media Streaming App	Activated
<input checked="" type="checkbox"/> Cisco CTIManager	Activated
<input checked="" type="checkbox"/> Cisco Extension Mobility	Activated
<input checked="" type="checkbox"/> Cisco Extended Functions	Activated
<input type="checkbox"/> Cisco DHCP Monitor Service	Deactivated
<input checked="" type="checkbox"/> Cisco Intercluster Lookup Service	Activated
<input checked="" type="checkbox"/> Cisco Location Bandwidth Manager	Activated
<input checked="" type="checkbox"/> Cisco Dialed Number Analyzer Server	Activated
<input checked="" type="checkbox"/> Cisco Dialed Number Analyzer	Activated
<input checked="" type="checkbox"/> Cisco Tftp	Activated

REPEAT ON SERVERS CUCM9-SUB1 and CUCM9-PUB2.

Remove DNS reliance, and configure Enterprise Parameters/Service Parameters on CUCM servers

Navigate to **System > Server > Find** on CUCM9-PUB1.

Any server listed by name needs to be updated such that it's known by its IP address. In this example the "CUCM9-PUB1" server is listed by name.

CUCM9-PUB1 > Host Name/IP Address > 192.168.1.71 > Save > OK

Although not all of the following are requirements for this particular lab, get in a habit of configuring the following:

Navigate to **System > Enterprise Parameters** on **CUCM9-PUB1**.

REPEAT ON CUCM9-PUB2 SERVER.

However, in our case, the CUCM9-PUB2 server is already known by its IP address.

Auto Registration Phone Protocol: **SCCP**

BLF For Call Lists: **Enabled**

Advertise G.722 Codec: **Disabled** (**NOTE:** Do this if the lab specifies G.711 should be used within a site.)

Under the **Phone URL Parameters** section, replace any occurrences of the publisher's server's name with its IP address.

Save and **Reset** when complete.

REPEAT ON CUCM9-PUB2 SERVER.

Navigate to **System > Service Parameters** on **CUCM9-PUB1**.

Select the **Publisher** server from the **Server** drop-down menu.

Select **Cisco CallManager** from the **Service** drop-down menu.

T302 Timer: 3000

H225 T302 Timer: 3000

Stop Routing on Unallocated Number Flag: False

Stop Routing on User Busy Flag: False

G722 Codec Enabled: Disabled (**NOTE:** Do this if the lab specifies G.711 should be used within a site.)

Default Interregion Max Audio Bit Rate: 16 kbps (iLBC, G.728) (**NOTE:** Do this if the lab specifies iLBC (as opposed to G.729) should be used within a site.)

Automated Alternate Routing Enable: True

Enabled Enterprise Feature Access: True

Enabled Mobile Voice Access: True

Matching Caller ID with Remote Destination: Partial Match

Number of Digits for Caller ID Partial Match: 7

REPEAT ON CUCM9-PUB2 SERVER.

Module 2: Doing EVERYTHING on Switch SW1

Task 1.1 VLANs

Confirm CDP operation

```
SW1(config)#cdp run  
SW1(config)#cdp timer 5  
SW1(config)#cdp advertise-v2
```

Assign IP phone ports to appropriate VLANs

```
SW1(config)#interface range fa 1/0/7-8  
SW1(config-if)#switchport access vlan 300  
SW1(config-if)#switchport voice vlan 200
```

Set the appropriate Native VLAN for the trunk port

```
SW1(config)#interface fa 1/0/24  
SW1(config-if)#switchport trunk native vlan 300
```

Task 6.1 QoS

Enable QoS, and configure CoS to DSCP mappings

```
SW1(config)#mls qos  
SW1(config)#mls qos map cos-dscp 0 8 16 24 34 46 48 56
```

Configure CoS to Queue/Threshold mappings

```
SW1(config)#mls qos srr-queue output cos-map queue 1 threshold 3 4 5  
SW1(config)#mls qos srr-queue output cos-map queue 2 threshold 3 6 7  
SW1(config)#mls qos srr-queue output cos-map queue 3 threshold 3 3  
SW1(config)#mls qos srr-queue output cos-map queue 4 threshold 1 0  
SW1(config)#mls qos srr-queue output cos-map queue 4 threshold 2 1  
SW1(config)#mls qos srr-queue output cos-map queue 4 threshold 3 2
```

Configure Queue-Set #2 Drop Thresholds for Queue #4

```
SW1(config)#mls qos queue-set output 2 threshold 4 25 50 100 100
```

Assign IP phone ports to Queue-Set #2

```
SW1(config)#interface range fa 1/0/7-8  
SW1(config-if)#queue-set 2
```

Instruct IP phone ports to trust CoS markings if, and only if, they came from an IP phone

```
SW1(config-if)#mls qos trust cos
```

```
SW1(config-if)#mls qos trust device cisco-phone
```

Configure IP phone ports with a Priority Queue

```
SW1(config-if)#priority-queue out
```

Configure the trunk port to trust DSCP markings

```
SW1(config)#interface fa 1/0/24  
SW1(config-if)#mls qos trust dscp
```

Configure the trunk port with a Priority Queue

```
SW1(config-if)#priority-queue out
```

Configure Shared Round Robin and Shaped Round Robin on the trunk port

```
SW1(config-if)#srr-queue bandwidth share 1 40 20 40  
SW1(config-if)#srr-queue bandwidth shape 8 0 0 0
```

Module 3: Configuring Router HQ (Part 1 of 2)

Task 1.1 VLANs

NOTE: The subinterfaces for the VLANs coming in over the trunk from switch SW1 are already correctly configured.

```
HQ#sh run  
... OUTPUT OMITTED ...  
interface GigabitEthernet0/0  
  description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$  
  no ip address  
  duplex auto  
  speed auto  
!  
interface GigabitEthernet0/0.100  
  encapsulation dot1Q 100  
  ip address 192.168.1.77 255.255.255.0  
!  
interface GigabitEthernet0/0.200  
  encapsulation dot1Q 200  
  ip address 10.10.120.1 255.255.255.0  
!  
interface GigabitEthernet0/0.300  
  encapsulation dot1Q 300 native  
  ip address 10.10.130.1 255.255.255.0
```

... OUTPUT OMITTED ...

Task 1.2 DHCP

Add an IP Helper Address to the Voice VLAN's subinterface

```
HQ#conf t  
Enter configuration commands, one per line. End with CNTL/Z.  
HQ(config)#int gig 0/0.200  
HQ(config-subif)#ip helper-address 192.168.1.71  
HQ(config-subif)#end  
HQ#
```

Task 1.3 NTP

Configure Network Time Protocol (NTP)

```
HQ#conf t  
Enter configuration commands, one per line. End with CNTL/Z.  
HQ(config)#ntp server 192.168.1.78  
HQ(config)#ntp source loopback0  
HQ(config)#clock timezone PST -8  
HQ(config)#clock summer-time PDT recurring  
HQ(config)#end  
HQ#
```

Task 3.1 HQ H.323 Gateway

Configure the ISDN switch type

```
HQ#conf term  
Enter configuration commands, one per line. End with CNTL/Z.  
HQ(config)#isdn switch-type primary-ni
```

Configure the T1 controller

```
HQ(config)#card type t1 0 0  
HQ(config)#network-clock-participate wic 0  
HQ(config)#network-clock-select 1 T1 0/0/0  
HQ(config)#controller T1 0/0/0  
HQ(config-controller)#linecode b8zs  
HQ(config-controller)#framing esf  
HQ(config-controller)#pri-group timeslots 1-3  
HQ(config-controller)#no shutdown  
HQ(config-controller)#int serial 0/0/0:23  
HQ(config-if)#isdn outgoing display-ie  
HQ(config-if)#isdn outgoing ie redirecting-number
```

Configure an H.323 interface

```
HQ(config-if)#int gig 0/0.200  
HQ(config-subif)#h323-gateway voip interface  
HQ(config-subif)#h323-gateway voip bind srcaddr 10.10.120.1
```

Configure an H.323 voice class to allow a dial peer to fail over after 3 seconds

```
HQ(config-subif)#voice class h323 1  
HQ(config-class)#h225 timeout tcp establish 3
```

Configure a codec voice class to support multiple codecs

```
HQ(config-class)#voice class codec 1  
HQ(config-class)#codec preference 1 g711ulaw  
HQ(config-class)#codec preference 2 ilbc
```

Configure an inbound dial peer to suppress secondary dial tone

```
HQ(config-class)#dial-peer voice 1 pots  
HQ(config-dial-peer)#incoming called-number .  
HQ(config-dial-peer)#direct-inward-dial
```

Configure dial peers to point to the CUCM Subscriber and Publisher for calls destined for 2...

```
HQ(config-dial-peer)#dial-peer voice 2 voip  
HQ(config-dial-peer)#destination-pattern 2...$  
HQ(config-dial-peer)#session target ipv4:192.168.1.72  
HQ(config-dial-peer)#voice-class h323 1  
HQ(config-dial-peer)#voice-class codec 1  
HQ(config-dial-peer)#dtmf-relay h245-alphanumeric  
HQ(config-dial-peer)#no vad  
HQ(config-dial-peer)#dial-peer voice 3 voip  
HQ(config-dial-peer)#preference 1  
HQ(config-dial-peer)#destination-pattern 2...$  
HQ(config-dial-peer)#session target ipv4:192.168.1.71  
HQ(config-dial-peer)#voice-class h323 1  
HQ(config-dial-peer)#voice-class codec 1  
HQ(config-dial-peer)#dtmf-relay h245-alphanumeric  
HQ(config-dial-peer)#no vad
```

Strip incoming DNIS from 10 digits down to 4 digits

```
HQ(config-dial-peer)#voice translation-rule 1  
HQ(cfg-translation-rule)#rule 1 /.*(2...$)/ /\1/  
HQ(cfg-translation-rule)#voice translation-profile STRIP  
HQ(cfg-translation-profile)#translate called 1
```

```
HQ(cfg-translation-profile)#voice-port 0/0/0:23
HQ(config-voiceport)#translation-profile incoming STRIP
```

Module 4: Configuring Router HQ (Part 2 of 2)

Task 3.4 SIP Trunk

Configure CUBE

```
HQ(config)#voice service voip
HQ(conf-voi-serv)#no ip address trusted authenticate
HQ(conf-voi-serv)#mode border-element (NOTE: This will require a
router reboot.
HQ(conf-voi-serv)#allow-connections sip to sip
HQ(conf-voi-serv)#allow-connections sip to h323
HQ(conf-voi-serv)#allow-connections h323 to sip
HQ(conf-voi-serv)#allow-connections h323 to h323
HQ(conf-voi-serv)#sip
HQ(conf-serv-sip)#bind control source-interface gig 0/0.200
HQ(conf-serv-sip)#bind media source-interface gig 0/0.200
HQ(conf-serv-sip)#dial-peer voice 5 voip (NOTE: This will act as the
incoming dial peer for calls using the CUBE.)
HQ(config-dial-peer)#session protocol sipv2
HQ(config-dial-peer)#incoming called-number [23]...$
HQ(config-dial-peer)#codec transparent (NOTE: The Region configuration
will ensure that the iLBC codec is used between HQ and BR1 Phones.)
HQ(config-dial-peer)#dtmf-relay rtp-nte sip-notify sip-kpml
HQ(config-dial-peer)#no vad
HQ(config-dial-peer)#dial-peer voice 6 voip
HQ(config-dial-peer)#destination-pattern 3...$
HQ(config-dial-peer)#session protocol sipv2
HQ(config-dial-peer)#session target ipv4:10.10.150.71
HQ(config-dial-peer)#dtmf-relay rtp-nte sip-notify sip-kpml
HQ(config-dial-peer)#codec transparent
HQ(config-dial-peer)#no vad
HQ(config-dial-peer)#dial-peer voice 7 voip
HQ(config-dial-peer)#destination-pattern 2...$
HQ(config-dial-peer)#session protocol sipv2
HQ(config-dial-peer)#session target ipv4:192.168.1.72
HQ(config-dial-peer)#codec transparent
HQ(config-dial-peer)#dtmf-relay rtp-nte sip-notify sip-kpml
HQ(config-dial-peer)#no vad
HQ(config-dial-peer)#dial-peer voice 8 voip
HQ(config-dial-peer)#destination-pattern 2...$
HQ(config-dial-peer)#session protocol sipv2
HQ(config-dial-peer)#session target ipv4:192.168.1.71
HQ(config-dial-peer)#preference 1
HQ(config-dial-peer)#codec transparent
HQ(config-dial-peer)#dtmf-relay rtp-nte sip-notify sip-kpml
```

```
HQ(config-dial-peer)#no vad
```

Task 4.1 HQ Call Routing

Configure an outbound dial peer

```
HQ(config-voiceport)#dial-peer voice 4 pots
HQ(config-dial-peer)#destination-pattern .T
HQ(config-dial-peer)#port 0/0/0:23
```

Task 5.1 Video Conference Bridge

Configure an outbound dial peer

```
HQ(config)#voice-card 0
HQ(config-voicecard)#voice-service dsp-reservation 30
    50% of DSPs allocated for voice and TDM video services.
    Remaining DSPs allocated for IP video service
HQ(config-voicecard)#dsp services dspfarm
HQ(config-voicecard)#exit

HQ(config)#sccp local gig 0/0.200
HQ(config)#sccp ccm 192.168.1.72 identifier 1 version 7.0+
HQ(config)#sccp ccm 192.168.1.71 identifier 2 version 7.0+
HQ(config)#sccp

HQ(config)#dspfarm profile 1 conference video homogeneous
HQ(config-dspfarm-profile)#codec g711ulaw
HQ(config-dspfarm-profile)#codec g711alaw
HQ(config-dspfarm-profile)#codec g729ar8
HQ(config-dspfarm-profile)#codec g729abr8
HQ(config-dspfarm-profile)#codec g729r8
HQ(config-dspfarm-profile)#codec g729br8
HQ(config-dspfarm-profile)#codec ilbc
HQ(config-dspfarm-profile)# codec h264 cif frame-rate 30 bitrate
320kbps
HQ(config-dspfarm-profile)#maximum conference-participants 8
HQ(config-dspfarm-profile)#maximum sessions 4
HQ(config-dspfarm-profile)#associate application SCCP
HQ(config-dspfarm-profile)#no shutdown
HQ(config-dspfarm-profile)#exit

HQ(config)#sccp ccm group 1
HQ(config-sccp-ccm)#associate ccm 1 priority 1
HQ(config-sccp-ccm)#associate ccm 2 priority 2
HQ(config-sccp-ccm)#associate profile 1 register HQ-VID-CFB
HQ(config-sccp-ccm)#end
HQ#
```

Task 6.2 Router QoS

Using the QoS SRND (a document available to you in the real lab), lookup (in Table 3-10) how much bandwidth is required for an iLBC call using a 50 packet per second packetization rate, over a Multilink PPP connection

The table tells us that the required bandwidth is 35.2 kbps per call. We want to support 5 simultaneous calls. So, we multiply 35.2 kbps by 5, giving us a total bandwidth requirement of **176 kbps**.

Temporarily set HQ's S 0/1/0.2 to a bandwidth of 768kbps or less, to force AutoQoS to enable Link Fragmentation and Interleaving (LFI), specifically Multilink PPP (MLP)

```
HQ#conf term  
Enter configuration commands, one per line. End with CNTL/Z.  
HQ(config)#int s 0/1/0.2  
HQ(config-subif)#bandwidth 768
```

Enable AutoQoS on the DLCI

```
HQ(config-subif)#frame-relay interface-dlci 103  
HQ(config-fr-dlci)#auto qos voip trust fr-atm
```

Reset the DLCI's bandwidth back to a value back to a T1 speed (i.e. 1.544 Mbps)

```
HQ(config-fr-dlci)#int s 0/1/0.2  
HQ(config-subif)#bandwidth 1544  
HQ(config-subif)#end
```

Check to make sure cRTP was not configured by AutoQoS, and remove the cRTP configuration if it exists (NOTE: Since we gave the fr-atm option to configure MLP, as opposed to FRF.12, we don't have a command enabling cRTP).

```
HQ#show run | b 0/1/0.2  
interface Serial0/1/0.2 point-to-point  
bandwidth 1544  
frame-relay interface-dlci 103 ppp Virtual-Template200  
class AutoQoS-FR-Se0/1/0-103  
auto qos voip trust fr-atm
```

Check the Policy Map that AutoQoS configured

```
HQ#show run | s policy-map  
policy-map AutoQoS-Policy-Trust  
class AutoQoS-VoIP-RTP-Trust  
priority percent 70  
class AutoQoS-VoIP-Control-Trust
```

```
bandwidth percent 5
class class-default
fair-queue
```

Change the priority bandwidth from 70 percent of the interface's bandwidth to the 176 kbps value calculated earlier

HQ#**conf term**

Enter configuration commands, one per line. End with CNTL/Z.

```
HQ(config)#policy-map AutoQoS-Policy-Trust
HQ(config-pmap)#class AutoQoS-VoIP-RTP-Trust
HQ(config-pmap-c)#priority 176
HQ(config-pmap-c)#end
```

REBOOT HQ WHILE WORKING ON BR2. THIS LETS THE MODE BORDER-ELEMENT COMMAND TAKE EFFECT.

Configure the other side of the link on BR2, beginning by determining the interface connected to HQ

BR2#**show cdp neigh**

Capability Codes: R - Router, T - Trans Bridge, B - Source Route Bridge
S - Switch, H - Host, I - IGMP, r - Repeater, P - Phone,
D - Remote, C - CVTA, M - Two-port Mac Relay

Device ID	Local Intrfce	Holdtme	Capability	Platform	Port ID
HQ.kwtrain.com	Ser 0/1/0.1	145	R S I	CISCO2911	Ser 0/1/0.2
SEP0CD9969026A3	Fas 0/2/0	174	H M	IP Phone	Port 1
SEP001C58FB7601	Fas 0/2/1	152	H M	IP Phone	Port 1

Check current bandwidth configuration of S 0/1/0.1 (NOTE: No bandwidth command implies a default configuration.)

BR2#**show run | b 0/1/0.1**

```
interface Serial0/1/0.1 point-to-point
ip address 10.10.33.130 255.255.255.128
frame-relay interface-dlci 301
```

Temporarily set BR1's S 0/1/0.2 to a bandwidth of 768kbps or less, to force AutoQoS to enable Link Fragmentation and Interleaving (LFI), specifically Multilink PPP (MLP)

BR2#**conf term**

Enter configuration commands, one per line. End with CNTL/Z.

```
BR2(config)#int s 0/1/0.1
BR2(config-subif)#bandwidth 768
```

Enable AutoQoS on the DLCI

```
BR2(config-subif)#frame-relay interface-dlci 301
BR2(config-fr-dlci)#auto qos voip trust fr-atm
```


Reset the DLCI's bandwidth back to a value back to a T1 speed (i.e. 1.544 Mbps)

```
BR2(config-fr-dlci)#int s 0/1/0.1  
BR2(config-subif)#bandwidth 1544  
BR2(config-subif)#end
```

Check to make sure cRTP was not configured by AutoQoS, and remove the cRTP configuration if it exists (NOTE: Since we gave the fr-atm option to configure MLP, as opposed to FRF.12, we don't have a command enabling cRTP).

```
BR2#show run | b 0/1/0  
interface Serial0/1/0  
  no ip address  
  encapsulation frame-relay IETF  
  frame-relay traffic-shaping
```

Check the Policy Map that AutoQoS configured

```
BR2#show run | s policy-map  
policy-map AutoQoS-Policy-Trust  
  class AutoQoS-VoIP-RTP-Trust  
    priority percent 70  
  class AutoQoS-VoIP-Control-Trust  
    bandwidth percent 5  
  class class-default  
    fair-queue
```

Change the priority bandwidth from 70 percent of the interface's bandwidth to the 176 kbps value calculated earlier

```
BR2#conf term  
Enter configuration commands, one per line. End with CNTL/Z.  
BR2(config)#policy-map AutoQoS-Policy-Trust  
BR2(config-pmap)#class AutoQoS-VoIP-RTP-Trust  
BR2(config-pmap-c)#priority 176  
BR2(config-pmap-c)#end
```

Check reachability to the Loopback 0 interface on router HQ (NOTE: If the ping fails, save the configurations on both routers, and reboot

```
BR2#ping 10.10.32.1  
Type escape sequence to abort.  
Sending 5, 100-byte ICMP Echos to 10.10.32.1, timeout is 2 seconds:  
!!!!  
Success rate is 100 percent (5/5), round-trip min/avg/max = 1/1/4 ms  
BR2#
```

Module 5: Configuring Router BR1

Task 1.1 VLANs

Verify CDP is correctly configured

BR1#**conf term**

Enter configuration commands, one per line. End with CNTL/Z.

BR1(config)#**cdp run**

BR1(config)#**cdp timer 5**

BR1(config)#**cdp advertise-v2**

Identify ports connected to IP phones

BR1#**show cdp neighbor**

Capability Codes: R - Router, T - Trans Bridge, B - Source Route Bridge
S - Switch, H - Host, I - IGMP, r - Repeater, P - Phone,
D - Remote, C - CVTA, M - Two-port Mac Relay

Device ID	Local Intrfce	Holdtme	Capability	Platform	Port ID
HQ.kwtrain.com	Ser 0/1/0.1	154	R S I	CISCO2911	Ser 0/1/0.1
SEP0CD996919A55	Fas 0/2/0	176	H M	IP Phone	Port 2
SEP0024C40D8CFC	Fas 0/2/1	153	H M	IP Phone	Port 1

Configure the Voice and Data VLANs on ports connecting to IP phones

BR1#**conf t**

Enter configuration commands, one per line. End with CNTL/Z.

BR1(config)#**int range fa 0/2/0-1**

BR1(config-if-range)#**switchport access vlan 500**

BR1(config-if-range)#**switchport voice vlan 400**

BR1(config-if-range)#**spanning-tree portfast**

BR1(config-if-range)#**end**

Verify that Voice and Data Switch Virtual Interfaces (SVIs) exist

BR1#**show ip int brief**

Interface	IP-Address	OK?	Method	Status	Protocol
Embedded-Service-Engine0/0	unassigned	YES	NVRAM	administratively down	down
GigabitEthernet0/0	unassigned	YES	NVRAM	administratively down	down
GigabitEthernet0/1	unassigned	YES	NVRAM	administratively down	down
GigabitEthernet0/2	unassigned	YES	NVRAM	administratively down	down
Serial0/1/0	unassigned	YES	NVRAM	up	up
Serial0/1/0.1	10.10.33.2	YES	NVRAM	up	up
FastEthernet0/2/0	unassigned	YES	unset	up	up
FastEthernet0/2/1	unassigned	YES	unset	up	up
FastEthernet0/2/2	unassigned	YES	unset	down	down
FastEthernet0/2/3	unassigned	YES	unset	up	up
Loopback0	10.10.32.2	YES	NVRAM	up	up
Vlan1	unassigned	YES	unset	down	down
Vlan400	10.10.140.1	YES	NVRAM	up	up
Vlan500	10.10.150.1	YES	NVRAM	up	up

Task 1.2 DHCP

Exclude range of IP addresses from being assigned via DHCP

```
BR1#conf t
Enter configuration commands, one per line. End with CNTL/Z.
BR1(config)#ip dhcp excluded-address 10.10.140.1 10.10.140.9
BR1(config)#ip dhcp excluded-address 10.10.140.21 10.10.140.254
BR1(config)#ip dhcp pool IPPHONES
BR1(dhcp-config)#network 10.10.140.0 255.255.255.0
BR1(dhcp-config)#option 150 ip 10.10.150.71
BR1(dhcp-config)#default-router 10.10.140.1
BR1(dhcp-config)#end
BR1#
```

Task 1.3 NTP

Specify the NTP server (router HQ's Loopback 0 interface)

```
BR1#conf t
Enter configuration commands, one per line. End with CNTL/Z.
BR1(config)#ntp server 10.10.32.1
```

Set the time zone parameters for BR1

```
BR1(config)#clock timezone EST -5
BR1(config)#clock summer-time EDT recurring
BR1(config)#end
BR1#
```

Task 3.2 BR1 Cisco IOS MGCP T1-PRI Gateway

We'll wait on Task 3.2 until we define the MGCP gateway in the CUCM server.

Module 6: Configuring Router BR2 (Part 1 of 2)

Task 1.1 VLANs

Verify CDP is correctly configured

```
BR2#conf t
Enter configuration commands, one per line. End with CNTL/Z.
BR2(config)#cdp run
BR2(config)#cdp timer 5
BR2(config)#cdp advertise-v2
BR2(config)#end
BR2#
```

Identify port connected to IP phones

```
BR2#sh cdp neigh
```

Capability Codes: R - Router, T - Trans Bridge, B - Source Route Bridge
S - Switch, H - Host, I - IGMP, r - Repeater, P - Phone,
D - Remote, C - CVTA, M - Two-port Mac Relay

Device ID	Local Intrfce	Holdtme	Capability	Platform	Port ID
HQ.kwtrain.com	Ser 0/1/0.1	148	R S I	CISCO2911	Ser 0/1/0.2
SEP0CD9969026A3	Fas 0/2/0	157	H M	IP Phone	Port 1
SEP001C58FB7601	Fas 0/2/1	138	H M	IP Phone	Port 1

Configure the Voice and Data VLANs on ports connecting to IP phones

```
BR2#conf t
```

```
BR2 (config)#int range fa 0/2/0-1
```

```
BR2 (config-if-range)#switchport access vlan 700
```

```
BR2 (config-if-range)#switchport voice vlan 600
```

```
BR2 (config-if-range)#spanning-tree portfast
```

```
BR2 (config-if-range)#end
```

```
BR2#
```

Check to see if Voice and Data Switch Virtual Interfaces (SVIs) exist

```
BR2#show ip int brief
```

Interface	IP-Address	OK?	Method	Status	Protocol
Embedded-Service-Engine0/0	unassigned	YES	NVRAM	administratively down	down
GigabitEthernet0/0	unassigned	YES	NVRAM	administratively down	down
GigabitEthernet0/1	unassigned	YES	NVRAM	administratively down	down
GigabitEthernet0/2	unassigned	YES	NVRAM	administratively down	down
Serial0/1/0	unassigned	YES	NVRAM	up	up
Serial0/1/0.1	unassigned	YES	unset	up	up
FastEthernet0/2/0	unassigned	YES	unset	up	up
FastEthernet0/2/1	unassigned	YES	unset	up	up
FastEthernet0/2/2	unassigned	YES	unset	down	down
FastEthernet0/2/3	unassigned	YES	unset	down	down
SM1/0	unassigned	YES	NVRAM	administratively down	down
SM1/1	unassigned	YES	NVRAM	administratively down	down
Loopback0	10.10.32.3	YES	NVRAM	up	up
Virtual-Access1	10.10.33.130	YES	NVRAM	up	up
Virtual-Access2	unassigned	YES	unset	down	down
Virtual-Access3	10.10.33.130	YES	NVRAM	up	up
Virtual-Template200	10.10.33.130	YES	NVRAM	down	down
Vlan1	unassigned	YES	unset	down	down

No SVIs have been created for VLANs 600 and 700; so they need to be added

```
BR2#conf t
```

Enter configuration commands, one per line. End with CNTL/Z.

```
BR2 (config)#int vlan 600
```

```
BR2 (config-if)#ip address 10.10.160.1 255.255.255.0
```

```
BR2 (config-if)#int vlan 700
```

```
BR2 (config-if)#ip address 10.10.170.1 255.255.255.0
```

```
BR2 (config-if)#end
```

```
BR2#
```

Task 1.2 DHCP

Configure a DHCP Pool for BR2 Phone 1

```
BR2#conf term  
Enter configuration commands, one per line. End with CNTL/Z.  
BR2(config)#ip dhcp pool BR2PH1  
BR2(dhcp-config)#host 10.10.160.10 255.255.255.0  
BR2(dhcp-config)#client-identifier 010C.D996.9026.A3  
BR2(dhcp-config)#default-router 10.10.160.1  
BR2(dhcp-config)#option 150 ip 10.10.32.3  
BR2(dhcp-config)#exit
```

Configure a DHCP Pool for BR2 Phone 1

```
BR2(config)#ip dhcp pool BR2PH2  
BR2(dhcp-config)#host 10.10.160.11 255.255.255.0  
BR2(dhcp-config)#client-identifier 0100.1C58.FB76.01  
BR2(dhcp-config)#default-router 10.10.160.1  
BR2(dhcp-config)#option 150 ip 10.10.32.3  
BR2(dhcp-config)#end  
BR2#
```

Task 1.3 NTP

Specify the NTP server (router HQ's Loopback 0 interface)

```
BR2#conf term  
Enter configuration commands, one per line. End with CNTL/Z.  
BR2(config)#ntp server 10.10.32.1
```

Specify the time zone parameters for BR2

```
BR2(config)#clock timezone TKY +9  
BR2(config)#end  
BR2#
```

Task 2.4 CUCME SIP IP Phone Registration

Create the VoIP voice service for the Cisco 9971 SIP phone

```
BR2#conf term  
Enter configuration commands, one per line. End with CNTL/Z.  
BR2(config)#voice service voip  
BR2(conf-voi-serv)#allow-connections sip to sip  
BR2(conf-voi-serv)#allow-connections sip to h323  
BR2(conf-voi-serv)#allow-connections h323 to sip  
BR2(conf-voi-serv)#allow-connections h323 to h323  
BR2(conf-voi-serv)#sip
```

```
BR2(conf-serv-sip)#bind all source-interface lo0
BR2(conf-serv-sip)#registrar server
BR2(conf-serv-sip)#end
BR2#
```

Determine if IP phone firmware files are available via TFTP

```
BR2#show run | i tftp-server
BR2#
```

If firmware files are not available via TFTP, determine the location of the IP phone firmware files in flash

```
BR2#show flash
-#- --length-- -----date/time----- path
...OUTPUT OMITTED...
46          0 Jul 12 2015 04:17:54 +09:00 phones
47          0 Jul 12 2015 04:18:12 +09:00 phones/7945-7965
48      4639974 Jul 12 2015 04:19:46 +09:00 phones/7945-7965/apps45.9-2-1TH1-13.sbn
49      575590 Jul 12 2015 04:21:24 +09:00 phones/7945-7965/cnu45.9-2-1TH1-13.sbn
50     2211969 Jul 12 2015 04:22:06 +09:00 phones/7945-7965/cvm45sccp.9-2-1TH1-13.sbn
51      356907 Jul 12 2015 04:22:50 +09:00 phones/7945-7965/dsp45.9-2-1TH1-13.sbn
52     1886651 Jul 12 2015 04:23:34 +09:00 phones/7945-7965/jar45sccp.9-2-1TH1-13.sbn
53          656 Jul 12 2015 04:24:08 +09:00 phones/7945-7965/SCCP45.9-2-1S.loads
54          660 Jul 12 2015 04:25:10 +09:00 phones/7945-7965/term45.default.loads
55          660 Jul 12 2015 04:25:48 +09:00 phones/7945-7965/term65.default.loads
56           0 Jul 12 2015 04:18:18 +09:00 phones/9971
57      10728 Jul 12 2015 04:29:36 +09:00 phones/9971/dkern9971.100609R2-9-2-2SR1-
9.sebn
58     1508080 Jul 12 2015 04:30:12 +09:00 phones/9971/kern9971.9-2-2SR1-9.sebn
59     36692736 Jul 12 2015 04:31:30 +09:00 phones/9971/rootfs9971.9-2-2SR1-9.sebn
60     145016 Jul 12 2015 04:32:18 +09:00 phones/9971/sboot9971.031610R1-9-2-2SR1-
9.sebn
61      67000 Jul 12 2015 04:32:50 +09:00 phones/9971/skern9971.022809R2-9-2-2SR1-
9.sebn
62          1297 Jul 12 2015 04:33:36 +09:00 phones/9971/sip9971.9-2-2SR1-9.loads
...OUTPUT OMITTED...
```

If firmware files are not available via TFTP, make each of the firmware files available via TFTP, using an alias for each file, because the files are in subdirectories

Use a Microsoft Notepad document to create the configuration to past into router R2.

```
tftp-server flash:phones/7945-7965/apps45.9-2-1TH1-13.sbn alias apps45.9-2-1TH1-13.sbn
tftp-server flash:phones/7945-7965/cnu45.9-2-1TH1-13.sbn alias cnu45.9-2-1TH1-13.sbn
tftp-server flash:phones/7945-7965/cvm45sccp.9-2-1TH1-13.sbn alias cvm45sccp.9-2-1TH1-
13.sbn
tftp-server flash:phones/7945-7965/dsp45.9-2-1TH1-13.sbn alias dsp45.9-2-1TH1-13.sbn
tftp-server flash:phones/7945-7965/jar45sccp.9-2-1TH1-13.sbn alias jar45sccp.9-2-1TH1-
13.sbn
tftp-server flash:phones/7945-7965/SCCP45.9-2-1S.loads alias SCCP45.9-2-1S.loads
tftp-server flash:phones/7945-7965/term45.default.loads alias term45.default.loads
tftp-server flash:phones/7945-7965/term65.default.loads alias term65.default.loads
tftp-server flash:phones/9971/dkern9971.100609R2-9-2-2SR1-9.sebn alias
dkern9971.100609R2-9-2-2SR1-9.sebn
tftp-server flash:phones/9971/kern9971.9-2-2SR1-9.sebn alias kern9971.9-2-2SR1-9.sebn
tftp-server flash:phones/9971/rootfs9971.9-2-2SR1-9.sebn alias rootfs9971.9-2-2SR1-
```

```

9.sebn
tftp-server flash:phones/9971/sboot9971.031610R1-9-2-2SR1-9.sebn alias
sboot9971.031610R1-9-2-2SR1-9.sebn
tftp-server flash:phones/9971/skern9971.022809R2-9-2-2SR1-9.sebn alias
skern9971.022809R2-9-2-2SR1-9.sebn
tftp-server flash:phones/9971/sip9971.9-2-2SR1-9.loads alias sip9971.9-2-2SR1-9.loads

```

Globally configure CUCME to register SIP IP phones

```

BR2#conf t
BR2 (config)#voice register global
BR2 (config-register-global)#mode cme
BR2 (config-register-global)#source-address 10.10.32.3
BR2 (config-register-global)#bandwidth video tias-modifier 512000
negotiate end-to-end
BR2 (config-register-global)#max-dn 20
BR2 (config-register-global)#max-pool 10
BR2 (config-register-global)#ntp 10.10.32.1
BR2 (config-register-global)#timezone 44
BR2 (config-register-global)#load 9971 sip9971.9-2-2SR1-9.loads
BR2 (config-register-global)#tftp-path flash:
BR2 (config-register-global)#time-format 24
BR2 (config-register-global)#camera
BR2 (config-register-global)#video
BR2 (config-register-global)#create profile
BR2 (config-register-global)#end
BR2#

```

Create a directory number for the Cisco 9971 IP Phone

```

BR2#conf term
Enter configuration commands, one per line. End with CNTL/Z.
BR2 (config)#voice register dn 1
BR2 (config-register-dn)#number 4001
BR2 (config-register-dn)#name BR2 Phone 1

```

Define the Cisco 9971 IP Phone

```

BR2 (config)#voice register pool 1
BR2 (config-register-pool)#id mac OCD9.9690.26A3
BR2 (config-register-pool)#type 9971
BR2 (config-register-pool)#number 1 dn 1
BR2 (config-register-pool)#dtmf-relay sip-kpml (NOTE: Do not use sip-notify.
Otherwise, DTMF tones might not be correctly interpreted by Cisco Unity Express (CUE).)
BR2 (config-register-pool)#description +8144444001
BR2 (config-register-pool)#codec g711ulaw
BR2 (config-register-pool)#no vad
BR2 (config-register-pool)#end
BR2#

```

Task 2.5 CUCME SCCP IP Phone Registration

Globally define the CUCME SCCP service

```
BR2#conf term
Enter configuration commands, one per line.  End with CNTL/Z.
BR2(config)#telephony-service
BR2(config-telephony)#max-ephones 10
BR2(config-telephony)#max-dn 20 no-reg both
BR2(config-telephony)#ip source-address 10.10.32.3
BR2(config-telephony)#load 7965 SCCP45.9-2-1S
BR2(config-telephony)#time-zone 44
BR2(config-telephony)#date-format yy-mm-dd
BR2(config-telephony)#time-format 12
BR2(config-telephony)#system message CUCME
BR2(config-telephony)#create cnf-files
```

Create a directory number for the Cisco 7965 IP Phone

```
BR2(config-telephony)#ephone-dn 1 octo-line
BR2(config-ephone-dn)#number 4002 no-reg both
BR2(config-ephone-dn)#description +8144444002
BR2(config-ephone-dn)#name BR2 Phone 2
```

Define the Cisco 7965 IP Phone

```
BR2(config-ephone-dn)#ephone 1
BR2(config-ephone)#mac-address 001C.58FB.7601
BR2(config-ephone)#type 7965
BR2(config-ephone)#button 1:1
BR2(config-ephone)#privacy-button
BR2(config-ephone)#end
BR2#
```

Task 2.6 CUCME IP Phone Customization

To remove the password requirement, we need to modify the IP phone's .XML configuration file. We can make this edit by:

- Displaying the contents of the .XML file in our terminal emulator
- Copying and pasting the that text into Microsoft Notepad
- Editing the file
- Uploading the file from our desktop to a CUCM server (which acts as a TFTP server)
- Making the file available via TFTP
- Downloading the modified .XML file from the CUCM (acting as a TFTP server) to router BR2's flash
- Resetting the IP phone

Display the contents of the .XML file

BR2#show flash | i .xml

```
24      3412 Oct 25 2011 05:26:42 +09:00 xml.template
25      10230 Oct 25 2011 05:26:54 +09:00 xml-test.html
28      3939 Jul 13 2015 22:21:06 +09:00 SEP0CD9969026A3.cnf.xml
29      825 Jul 13 2015 22:21:06 +09:00 featurePolicyDefault.xml
30      4342 Jul 13 2015 22:21:04 +09:00 softkeyDefault_kpml.xml
31      4376 Jul 13 2015 22:21:06 +09:00 softkeyDefault.xml
32      69 Jul 13 2015 22:21:04 +09:00 syncinfo.xml
37      3282 Jan 11 2012 04:02:54 +09:00 its/vrf1/XMLDefault.cnf.xml
39      903 Dec 6 2014 02:19:54 +09:00 its/united_states_7960-tones.xml
40      8777 Dec 6 2014 02:19:54 +09:00 its/united_states_7960-font.xml
41      1313 Dec 6 2014 02:19:56 +09:00 its/united_states_7960-kate.xml
42      19750 Dec 6 2014 02:19:56 +09:00 its/united_states_7960-dictionary.xml
43      2740 Dec 6 2014 02:19:56 +09:00 its/united_states_SCCP-dictionary.xml
```

BR2#more SEP0CD9969026A3.cnf.xml

```
<device>
<deviceProtocol>SIP</deviceProtocol>
<devicePool>
<dateTimeSetting>
<dateTemplate>M/D/Y</dateTemplate>
<timeZone>Tokyo Standard Time</timeZone>
<ntp>
<ntp priority="0">
<name>10.10.32.1</name>
<ntpMode>directedbroadcast</ntpMode>
</ntp>
</ntp>
</dateTimeSetting>
<callManagerGroup>
<members>
<member priority="0">
<callManager>
<ports>
<sipPort>5060</sipPort>
</ports>
<processNodeName>10.10.32.3</processNodeName>
</callManager>
</member>
</members>
</callManagerGroup>
</devicePool>
<sipProfile>
<sipProxies>
<registerWithProxy>true</registerWithProxy>
</sipProxies>
<sipCallFeatures>
<cnfJoinEnabled>true</cnfJoinEnabled>
<localCfwdEnable>true</localCfwdEnable>
<callForwardURI>service-uri-cfwdall</callForwardURI>
<callPickupURI>service-uri-pickup</callPickupURI>
<callPickupGroupURI>service-uri-gpickup</callPickupGroupURI>
<callHoldRingback>2</callHoldRingback>
<semiAttendedTransfer>true</semiAttendedTransfer>
<anonymousCallBlock>2</anonymousCallBlock>
<callerIdBlocking>2</callerIdBlocking>
<dndControl>2</dndControl>
<remoteCcEnable>true</remoteCcEnable>
</sipCallFeatures>
<sipStack>
```

```

<remotePartyID>true</remotePartyID>
</sipStack>
<sipLines>
<line button="1" lineIndex="1">
<featureID>9</featureID>
<featureLabel>4001</featureLabel>
<proxy>USECALLMANAGER</proxy>
<port>5060</port>
<name>4001</name>
<displayName>BR2 Phone 1</displayName>
<autoAnswer>
<autoAnswerEnabled>2</autoAnswerEnabled>
</autoAnswer>
<callWaiting>1</callWaiting>
<authName></authName>
<authPassword></authPassword>
<sharedLine>false</sharedLine>
<messagesNumber></messagesNumber>
<ringSettingActive>5</ringSettingActive>
<forwardCallInfoDisplay>
<callerName>true</callerName>
<callerNumber>true</callerNumber>
<redirectedNumber>true</redirectedNumber>
<dialNumber>true</dialNumber>
</forwardCallInfoDisplay>
</line>
</sipLines>
<enableVad>false</enableVad>
<preferredCodec>g711ulaw</preferredCodec>
<dialTemplate></dialTemplate>
<kpml>1</kpml>
<phoneLabel>+8144444001</phoneLabel>
<stutterMsgWaiting>2</stutterMsgWaiting>
<disableLocalSpeedDialConfig>true</disableLocalSpeedDialConfig>
<dscpForAudio>184</dscpForAudio>
<dscpVideo>136</dscpVideo>
</sipProfile>
<commonProfile>
<phonePassword>Cisco</phonePassword>
<callLogBlfEnabled>2</callLogBlfEnabled>
</commonProfile>
<featurePolicyFile>featurePolicyDefault.xml</featurePolicyFile>
<loadInformation>sip9971.9-2-2SR1-9.loads</loadInformation>
<vendorConfig>
</vendorConfig>
<commonConfig>
<videoCapability>1</videoCapability>
<ciscoCamera>1</ciscoCamera>
</commonConfig>
<sshUserId></sshUserId>
<sshPassword></sshPassword>
<userId></userId>
<phoneServices>
<provisioning>2</provisioning>
<phoneService type="1" category="0">
<name>Missed Calls</name>
<phoneLabel></phoneLabel>
<url>Application: Cisco/MissedCalls</url>
<vendor></vendor>
<version></version>
</phoneService>
<phoneService type="1" category="0">
<name>Received Calls</name>

```

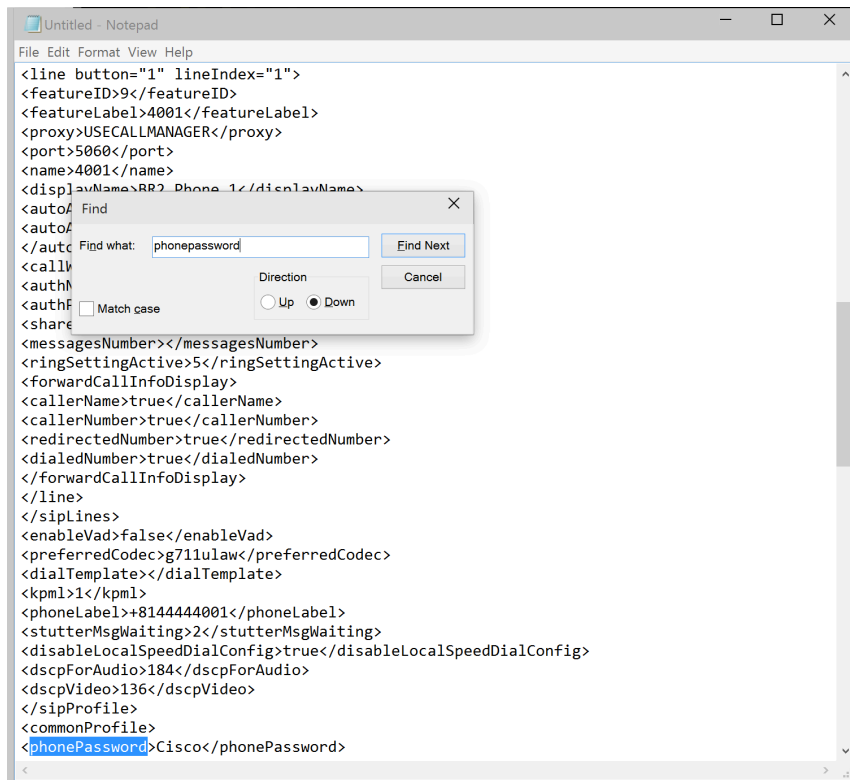
```

<phoneLabel></phoneLabel>
<url>Application:Cisco/ReceivedCalls</url>
<vendor></vendor>
<version></version>
</phoneService>
<phoneService type="1" category="0">
<name>Placed Calls</name>
<phoneLabel></phoneLabel>
<url>Application:Cisco/PlacedCalls</url>
<vendor></vendor>
<version></version>
</phoneService>
<phoneService type="2" category="0">
<name>Voicemail</name>
<phoneLabel></phoneLabel>
<url>Application:Cisco/Voicemail</url>
<vendor></vendor>
<version></version>
</phoneService>
</phoneServices>
<versionStamp>0005611535321036</versionStamp>
<userLocale>
<name>English_United_States</name>
<langCode>en</langCode>
</userLocale>
<networkLocale>United_States</networkLocale>
<networkLocaleInfo>
<name>United_States</name>
</networkLocaleInfo>
<authenticationURL></authenticationURL>
<directoryURL></directoryURL>
<servicesURL>http://10.10.32.3:80/CMEserverForPhone/serviceurl</servicesURL>
<dscpForSCCPPhoneServices>0</dscpForSCCPPhoneServices>
<dscpForCm2Dvce>96</dscpForCm2Dvce>
<transportLayerProtocol>2</transportLayerProtocol>
</device>

```

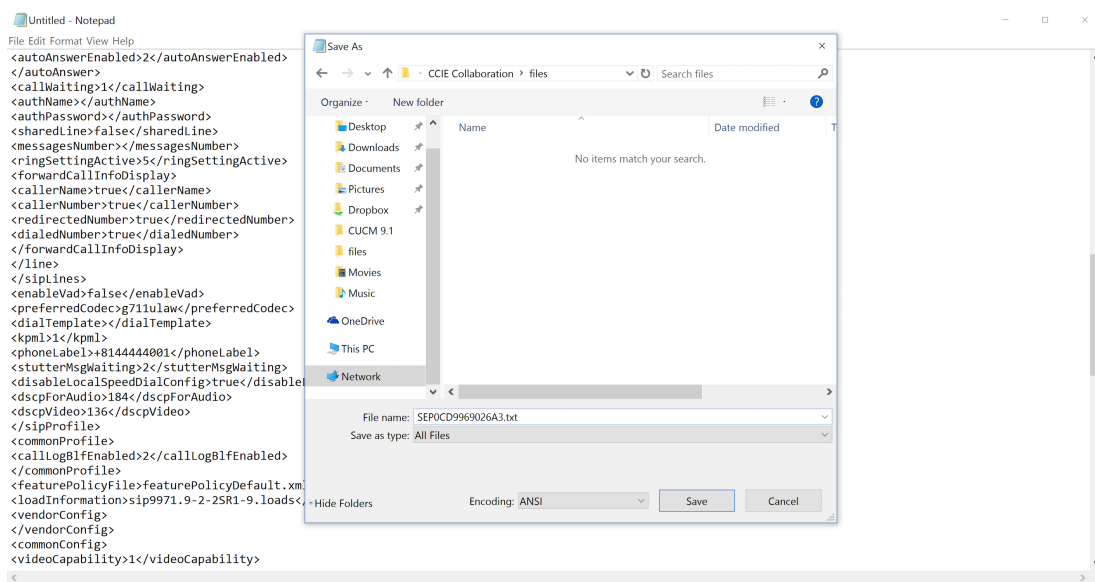
Copy the contents of the file, and past into Microsoft Notepad

Search for the line containing **phonepassword**.



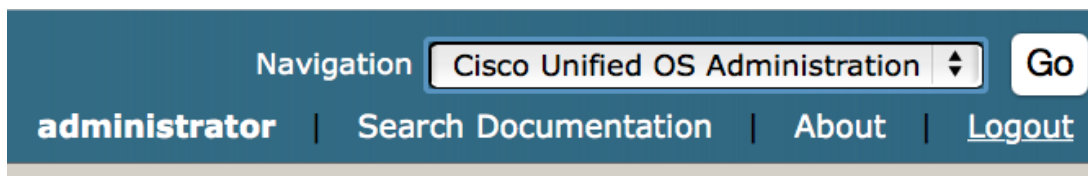
Edit the file

Delete that line from the document, and save the document with the same name as the original .XML file, **except** replace the .cnf.xml extension with a .txt extension. If you leave the original extension, the CUCM server might try disallow you from placing an .XML configuration file on its TFTP server.

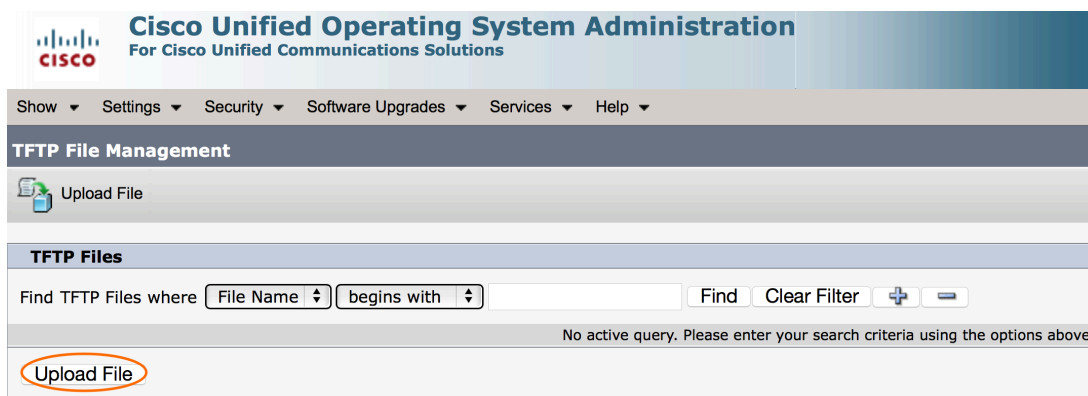


Upload the file to the BR1 CUCM Publisher

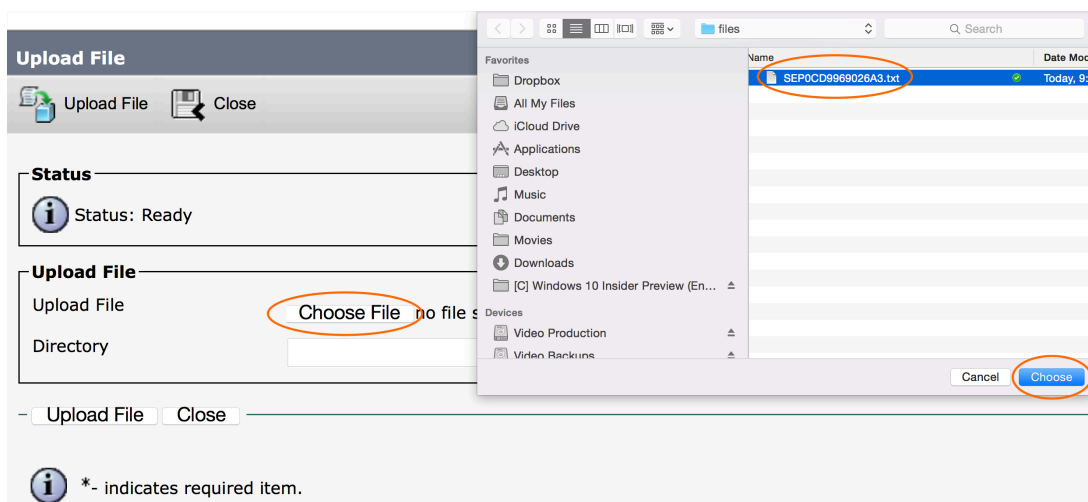
(On HQ PUB) Navigation > Cisco Unified OS Administration > Go (Then login.)



Software Upgrades > TFTP File Management > Upload File



Choose File > (Navigate to modified .XML file, which now has a .txt extension.) > Choose



Enter **/ccie** (or some other subdirectory name) as the **Directory**. Do **NOT** use the root directory, or your file might be removed when you restart the TFTP server.

Upload File

Close

Status

Status: Ready

Upload File

Upload File

Choose File

SEP0CD9969026A3.txt

Directory

/ccie

Upload File

Close

*- indicates required item.

Upload File

After the file uploads, click the **Close** button.

Make the file available via TFTP

The TFTP service needs to be restarted on the CUCM server in order for the file to be available via TFTP.

Navigation > Cisco Unified Serviceability > Go (Then login.)

Navigation

Cisco Unified Serviceability

Go

administrator

Search Documentation

About

Logout

Tools > Control Center – Feature Services > Cisco Tftp > Restart > OK

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Start Stop **Restart** Refresh Page

Status:
Ready

Select Server
Server* 10.10.150.71 Go

Performance and Monitoring Services					
	Service Name	Status:	Activation Status	Start Time	Up Time
<input type="radio"/>	Cisco Serviceability Reporter	Started	Activated	Tue Jul 21 09:34:54 2015	0 days 00:28:02
<input type="radio"/>	Cisco CallManager SNMP Service	Started	Activated	Tue Jul 21 09:34:56 2015	0 days 00:28:00

Directory Services					
	Service Name	Status:	Activation Status	Start Time	Up Time
<input type="radio"/>	Cisco DirSync	Started	Activated	Tue Jul 21 09:34:57 2015	0 days 00:27:59

CM Services					
	Service Name	Status:	Activation Status	Start Time	Up Time
<input type="radio"/>	Cisco CallManager	Started	Activated	Tue Jul 21 09:36:56 2015	0 days 00:26:00
<input type="radio"/>	Cisco Messaging Interface	Not Running	Activated		
<input type="radio"/>	Cisco Unified Mobile Voice Access Service	Started	Activated	Tue Jul 21 09:45:10 2015	0 days 00:17:46
<input type="radio"/>	Cisco IP Voice Media Streaming App	Started	Activated	Tue Jul 21 09:34:47 2015	0 days 00:28:09
<input type="radio"/>	Cisco CTIManager	Started	Activated	Tue Jul 21 09:34:50 2015	0 days 00:28:06
<input type="radio"/>	Cisco Extension Mobility	Started	Activated	Tue Jul 21 09:45:06 2015	0 days 00:17:50
<input type="radio"/>	Cisco DHCP Monitor Service	Started	Activated	Tue Jul 21 09:34:59 2015	0 days 00:27:57
<input type="radio"/>	Cisco Intercluster Lookup Service	Started	Activated	Tue Jul 21 09:35:00 2015	0 days 00:27:56
<input type="radio"/>	Cisco Location Bandwidth Manager	Started	Activated	Tue Jul 21 09:34:44 2015	0 days 00:28:12
<input type="radio"/>	Cisco Dialed Number Analyzer Server	Started	Activated	Tue Jul 21 09:35:03 2015	0 days 00:27:53
<input type="radio"/>	Cisco Dialed Number Analyzer	Started	Activated	Tue Jul 21 09:45:10 2015	0 days 00:17:46
<input checked="" type="radio"/>	Cisco Tftp	Started	Activated	Tue Jul 21 09:35:08 2015	0 days 00:27:48

Download the .XML file to BR2

```
BR2#copy tftp flash
Address or name of remote host []? 10.10.150.71
Source filename []? ccie/SEP0CD9969026A3.txt
Destination filename [SEP0CD9969026A3.txt]?
Accessing tftp://10.10.150.71/ccie/SEP0CD9969026A3.txt...
Loading ccie/SEP0CD9969026A3.txt from 10.10.150.71 (via Virtual-Access3): !
[OK - 4042 bytes]
```

4042 bytes copied in 0.472 secs (8564 bytes/sec)

BR2#

NOTE: Do **NOT** enter a leading / before the source filename.

Copy the newly downloaded .txt file to a .cnf.xml file

```
BR2#copy SEP0CD9969026A3.txt SEP0CD9969026A3.cnf.xml
Destination filename [SEP0CD9969026A3.cnf.xml]?
%Warning:There is a file already existing with this name
Do you want to over write? [confirm]
Copy in progress...C
4042 bytes copied in 0.304 secs (13296 bytes/sec)
```

BR2#

Finally, reset the Cisco 9971 IP Phone to make the new configuration file take effect

```
BR2#conf term
Enter configuration commands, one per line. End with CNTL/Z.
BR2(config)#voice register pool 1
BR2(config-register-pool)#reset
BR2(config-register-pool)#end
BR2#
```

While there are many tasks that we'll not verify, to preserve time on lab day, this task can be verified very quickly. Simply press the **Settings** button on the IP phone, and then touch **Administrator Settings** on the IP phone screen. You should be taken to the Administrator Settings menu without being prompted for a password.

Module 7: Configuring Router BR2 (Part 2 of 2)

Task 3.3 BR2 CUCME H.323 Gateway

Define the port type as an E1; specify ISDN switch type; and specify clocking

```
BR2#conf term
Enter configuration commands, one per line. End with CNTL/Z.
BR2(config)#card type e1 0 0
BR2(config)#isdn switch-type primary-net5
BR2(config)#network-clock-participate wic 0
BR2(config)#network-clock-select 1 e1 0/0/0
```

Configure the E1 controller

```
BR2(config)#controller e1 0/0/0
BR2(config-controller)#linecode hdb3
BR2(config-controller)#framing crc4
BR2(config-controller)#pri-group timeslots 1-3
BR2(config-controller)#no shutdown
BR2(config-controller)#interface serial 0/0/0:15
BR2(config-if)#isdn outgoing display-ie
BR2(config-if)#isdn outgoing ie redirecting-number
```

Configure the H.323 interface

```
BR2(config-if)#int vlan 600
BR2(config-if)#h323-gateway voip interface
BR2(config-if)#h323-gateway voip bind srcaddr 10.10.160.1
BR2(config-if)#exit
```

Configure an H.323 Voice Class to allow a dial peer to fail over after three seconds

```
BR2(config)#voice class h323 1
BR2(config-class)#h225 timeout tcp establish 3
BR2(config-class)#exit
BR2#
```


Configure an Codec Voice Class to allow either the G.711 uLaw codec or the iLBC codec

```
BR2(config)#voice class codec 1
BR2(config-class)#codec preference 1 g711ulaw
BR2(config-class)#codec preference 2 ilbc
BR2(config-class)#end
BR2#
```

Task 3.6 SIP Gateway

The prior commands have already configured router BR2 as a SIP gateway. The portion of task 3.6 relating to how BR2 IP phones should be able to call HQ and BR1 IP phones is related to BR2's route plan, which will be configured in Task 4.3 (below).

Task 4.3 BR2 Call Routing

Create a table listing the dialed pattern ANI/TON, DNIS/TON, and digit manipulation

BR2 Route Plan

Pattern	Route List	Route Group	ANI/TON	DNIS/TON	DM
999\$			8/S	999/U	No Digit-Strip
9[1-9].....\$			8/S	8/S	None
900T			E.164/I	Var./I	None
2...\$			4-Digit DN	4-Digit DN	None
3...\$			4-Digit DN	4-Digit DN	None

Create a Microsoft Notepad document (to be pasted into the router's configuration) with the configuration of the Dial Peers, Voice Translation Rules, and Voice Translation Profiles

Prevent two-stage dialing for incoming calls

```
BR2#conf term
Enter configuration commands, one per line. End with CNTL/Z.
BR2(config)#dial-peer voice 1 pots
BR2(config-dial-peer)#incoming called-number .
BR2(config-dial-peer)#direct-inward-dial
```

```
BR2(config-dial-peer)#exit
```

Strip incoming DNIS to a four-digit number

```
BR2(config)#voice translation-rule 1
BR2(cfg-translation-rule)#rule 1 /.*(4...$)/ /\1/
BR2(cfg-translation-rule)#exit
BR2(config)#voice translation-profile STRIP
BR2(cfg-translation-profile)#translate called 1
BR2(cfg-translation-profile)#exit
BR2(config)#voice-port 0/0/0:15
BR2(config-voiceport)#translation-profile in STRIP
BR2(config-voiceport)#exit
BR2(config)#
```

Configure voice translation to format number and TON information for 999 calls

```
BR2(config)#voice translation-rule 2
BR2(cfg-translation-rule)#rule 1 /^4...$/ /4444\0/ type any subscriber plan any isdn
BR2(cfg-translation-rule)#rule 2 // // type any unknown plan any isdn
BR2(cfg-translation-rule)#voice translation-profile 999
BR2(cfg-translation-profile)#translate calling 2
BR2(cfg-translation-profile)#translate called 2
BR2(cfg-translation-profile)#exit
```

Create a dial peer for 999 calls

```
BR2(config)#dial-peer voice 2 pots
BR2(config-dial-peer)#destination-pattern 999$
BR2(config-dial-peer)#no digit-strip
BR2(config-dial-peer)#port 0/0/0:15
BR2(config-dial-peer)#translation-profile out 999
BR2(config-dial-peer)#exit
BR2(config)#
```

Configure voice translation to format number and TON information for local calls

```
BR2(config)#voice translation-rule 3
BR2(cfg-translation-rule)#rule 1 /^4...$/ /4444\0/ type any subscriber plan any isdn
BR2(cfg-translation-rule)#rule 2 // // type any subscriber plan any isdn
BR2(cfg-translation-rule)#voice translation-profile LOCAL
BR2(cfg-translation-profile)#translate calling 3
BR2(cfg-translation-profile)#translate called 3
BR2(cfg-translation-profile)#exit
```

Create a dial peer for local calls

```
BR2(config)#dial-peer voice 3 pots
BR2(config-dial-peer)#destination-pattern 9[1-9].....$
BR2(config-dial-peer)#port 0/0/0:15
BR2(config-dial-peer)#translation-profile out LOCAL
BR2(config-dial-peer)#exit
BR2(config)#
```

Configure voice translation to format number and TON information for international calls

```
BR2(config)#voice translation-rule 4
BR2(cfg-translation-rule)#rule 1 /^4...$/ /+814444\0/ type any international plan any isdn
BR2(cfg-translation-rule)#rule 2 // // type any international plan any isdn
```

```
BR2(cfg-translation-rule)#exit
BR2(config)#voice translation-profile INTL
BR2(cfg-translation-profile)#translate calling 4
BR2(cfg-translation-profile)#translate called 4
BR2(cfg-translation-profile)#exit
```

Create a dial peer for international calls

```
BR2(config)#dial-peer voice 4 pots
BR2(config-dial-peer)#destination-pattern 900T
BR2(config-dial-peer)#port 0/0/0:15
BR2(config-dial-peer)#translation-profile out INTL
BR2(config-dial-peer)#end
BR2#
```

Create a dial peers for HQ calls

```
BR2(config)#dial-peer voice 5 voip
BR2(config-dial-peer)#destination-pattern 2...$
BR2(config-dial-peer)#session protocol sipv2
BR2(config-dial-peer)#session target ipv4:192.168.1.72
BR2(config-dial-peer)#voice-class codec 1
BR2(config-dial-peer)#dtmf-relay sip-kpml
BR2(config-dial-peer)#no vad
BR2(config-dial-peer)#exit
BR2(config)#dial-peer voice 6 voip
BR2(config-dial-peer)#destination-pattern 2...$
BR2(config-dial-peer)#session target ipv4:192.168.1.71
BR2(config-dial-peer)#voice-class codec 1
BR2(config-dial-peer)#dtmf-relay sip-kpml
BR2(config-dial-peer)#no vad
BR2(config-dial-peer)#preference 1
BR2(config-dial-peer)#exit
```

Create a dial peers for BR1 calls

```
BR2(config)#dial-peer voice 7 voip
BR2(config-dial-peer)#destination-pattern 3...$
BR2(config-dial-peer)#session protocol sipv2
BR2(config-dial-peer)#session target ipv4:10.10.150.71
BR2(config-dial-peer)#voice-class codec 1
BR2(config-dial-peer)#dtmf-relay sip-kpml
BR2(config-dial-peer)#no vad
BR2(config-dial-peer)#end
BR2#
```

Task 5.2 Ad-Hoc Conferencing

Create an ephone-dn for ad-hoc conferencing

```
BR2#conf t
Enter configuration commands, one per line. End with CNTL/Z.
BR2(config)#ephone-dn 2 octo-line (NOTE: The ephone-dn has to be configured for an ad-hoc conference. It
should be configured as an octo-line. Otherwise, two dual-line ephone-dns would be needed.)
BR2(config-ephone-dn)#number 4444 no-reg both (NOTE: The number 4444 was randomly chosen, because
a specific ad-hoc conference number was not specified.)
BR2(config-ephone-dn)#conference ad-hoc
BR2(config-ephone-dn)#exit
```

Configure a hardware conference bridge

```
BR2(config)#voice-card 0
BR2(config-voicecard)#dsp services dspfarm
BR2(config-voicecard)#exit

BR2(config)#sccp local lo0
BR2(config)#sccp ccm 10.10.32.3 identifier 1 version 7.0+
BR2(config)#sccp

BR2(config)#dspfarm profile 1 conference
BR2(config-dspfarm-profile)#codec g711ulaw
BR2(config-dspfarm-profile)#codec g711alaw
BR2(config-dspfarm-profile)#codec g729r8
BR2(config-dspfarm-profile)#codec g729ar8
BR2(config-dspfarm-profile)#codec g729br8
BR2(config-dspfarm-profile)#codec g729abr8
BR2(config-dspfarm-profile)#codec ilbc
BR2(config-dspfarm-profile)#max session 1
BR2(config-dspfarm-profile)#associate application SCCP
BR2(config-dspfarm-profile)#no shut
BR2(config-dspfarm-profile)#exit

BR2(config)#sccp ccm group 1
BR2(config-sccp-ccm)#associate ccm 1 priority 1
BR2(config-sccp-ccm)#associate profile 1 register BR2-CFB
BR2(config-sccp-ccm)#exit
```

Associate a hardware conference bridge with CUCME

```
BR2(config)#telephony-service
BR2(config-telephony)#sdspfarm units 2
BR2(config-telephony)#sdspfarm tag 1 BR2-CFB
BR2(config-telephony)#conference hardware
BR2(config-telephony)#end
BR2#
```

Module 8: Configuring Cisco Unity Express (CUE)

Task 7.4 Cisco Unity Express (CUE) Module Initialization

Configure the Service Engine interface on router BR2

```
BR2#conf term
Enter configuration commands, one per line. End with CNTL/Z.
BR2(config)#int sm 1/0 (NOTE: You can identify the interface by issuing the show ip int brief command.)
BR2(config-if)#ip unnumbered vlan 600 (NOTE: The IP unnumbered feature is used, because we are told that
the CUE module's IP address should be 10.10.160.2 /24, which is part of the same subnet as interface Vlan600.)
BR2(config-if)#service-module ip address 10.10.160.2 255.255.255.0
BR2(config-if)#service-module ip default-gateway 10.10.160.1 (NOTE: The default gateway is the IP
address of the Vlan600 SVI.)
BR2(config-if)#no shutdown
BR2(config-if)#exit
```

Add a static route for the router to get to the CUE module

```
BR2(config)#ip route 10.10.160.2 255.255.255.255 sm 1/0
```

```
BR2(config)#end
BR2#
```

Connect to and reconfigure the CUE module

```
BR2#service-module sm 1/0 session
Trying 10.10.160.1, 2067 ... Open
se-10-10-160-2# offline
!!!WARNING!!!: If you are going offline to do a backup, it is recommended
that you save the current running configuration using the 'write' command,
prior to going to the offline state.
```

Putting the system offline will disable management interfaces.

```
Are you sure you want to go offline?[confirm]
se-10-10-160-2(offline)# restore factory default
!!!WARNING!!!: This operation will cause all configuration and data
on the system to be erased. This operation is not reversible.
```

```
Do you wish to continue?[confirm]
Restoring the system. Please wait .....done
System will be restored to factory default when it reloads.
```

Press any key to reload:

System reloading

NOTE: The reload process takes a few minutes. While we wait, we can do some additional CUE integration configuration on BR2.

Create a dial peer to reach CUE

```
BR2#conf term
Enter configuration commands, one per line. End with CNTL/Z.
BR2(config)#dial-peer voice 4500 voip
BR2(config-dial-peer)#destination-pattern 45..
BR2(config-dial-peer)#session protocol sipv2
BR2(config-dial-peer)#session target ipv4:10.10.160.2
BR2(config-dial-peer)#dtmf-relay sip-notify
BR2(config-dial-peer)#codec g711ulaw
BR2(config-dial-peer)#no vad
BR2(config-dial-peer)#exit
```

Create ephone-dns for MWI

```
BR2(config)#ephone-dn 3
BR2(config-ephone-dn)#number 4510.... no-reg both
BR2(config-ephone-dn)#mwi on
BR2(config-ephone-dn)#ephone-dn 4
BR2(config-ephone-dn)#number 4511.... no-reg both
BR2(config-ephone-dn)#mwi off
BR2(config-ephone-dn)#exit
```

Configure CUCME for voicemail

```
BR2(config)#telephony-service
BR2(config-telephony)#voicemail 4500
BR2(config-telephony)#mwi relay
BR2(config-telephony)#create cnf-files
Creating CNF files
BR2(config-telephony)#voice register global
```

```
BR2(config-register-global)#voicemail 4500
BR2(config-register-global)#create profile
BR2(config-register-global)#exit
BR2(config)#
```

Add MWI server information to SIP configuration

```
BR2(config)#sip-ua
BR2(config-sip-ua)#mwi-server ipv4:10.10.160.2 unsolicited
BR2(config-sip-ua)#exit
```

Configure call forwarding behavior for ephone-dn and voice register pool

```
BR2(config)#ephone-dn 1
BR2(config-ephone-dn)#call-forward busy 4500
BR2(config-ephone-dn)#call-forward noan 4500 timeout 10
BR2(config-ephone-dn)#huntstop channel 1
BR2(config-ephone-dn)#voice register dn 1
BR2(config-register-dn)#call-forward b2bua busy 4500
BR2(config-register-dn)#call-forward b2bua noan 4500 timeout 10
BR2(config-register-dn)#huntstop channel 1
BR2(config-register-dn)#exit
```

Apply these updates to the IP phones

```
BR2(config)#ephone 1
BR2(config-ephone)#reset
resetting 001C.58FB.7601
BR2(config-ephone)#voice register pool 1
BR2(config-register-pool)#reset
BR2(config-register-pool)#end
BR2#
```

Reconnect to the CUE module, and go through the initialization wizard

...OUTPUT OMITTED...

```
IMPORTANT::
IMPORTANT::      Welcome to Cisco Systems Service Engine
IMPORTANT::      post installation configuration tool.
IMPORTANT::
IMPORTANT:: This is a one time process which will guide
IMPORTANT:: you through initial setup of your Service Engine.
IMPORTANT:: Once run, this process will have configured
IMPORTANT:: the system for your location.
IMPORTANT::
IMPORTANT:: If you do not wish to continue, the system will be halted
IMPORTANT:: so it can be safely removed from the router.
IMPORTANT::
```

```
Do you wish to start configuration now (y,n)? y
Are you sure (y,n)? y
```

```
Enter Hostname
(my-hostname, or enter to use se-10-10-160-2): CUE
```

```
Enter Domain Name
(mydomain.com, or enter to use localdomain):
Using localdomain as default
```

```
IMPORTANT:: DNS Configuration:
```

IMPORTANT::
IMPORTANT:: This allows the entry of hostnames, for example foo.cisco.com, instead
IMPORTANT:: of IP addresses like 1.100.10.205 for application configuration. In order
IMPORTANT:: to set up DNS you must know the IP address of at least one of your
IMPORTANT:: DNS Servers.

Would you like to use DNS (y,n)?**n**

WARNING: If DNS is not used, IP addresses will be required.

Are you sure (y,n)? **y**

Enter IP Address of the Primary NTP Server
(IP address, or enter for 10.10.160.1): **192.168.1.78**
Found server 192.168.1.78

Enter IP Address of the Secondary NTP Server
(IP address, or enter to bypass):

Please identify a location so that time zone rules can be set correctly.
Please select a continent or ocean.

1) Africa	4) Arctic Ocean	7) Australia	10) Pacific Ocean
2) Americas	5) Asia	8) Europe	
3) Antarctica	6) Atlantic Ocean	9) Indian Ocean	

#? **5**

Please select a country.

1) Afghanistan	18) Israel	35) Palestine
2) Armenia	19) Japan	36) Philippines
3) Azerbaijan	20) Jordan	37) Qatar
4) Bahrain	21) Kazakhstan	38) Russia
5) Bangladesh	22) Korea (North)	39) Saudi Arabia
6) Bhutan	23) Korea (South)	40) Singapore
7) Brunei	24) Kuwait	41) Sri Lanka
8) Cambodia	25) Kyrgyzstan	42) Syria
9) China	26) Laos	43) Taiwan
10) Cyprus	27) Lebanon	44) Tajikistan
11) East Timor	28) Macau	45) Thailand
12) Georgia	29) Malaysia	46) Turkmenistan
13) Hong Kong	30) Mongolia	47) United Arab Emirates
14) India	31) Myanmar (Burma)	48) Uzbekistan
15) Indonesia	32) Nepal	49) Vietnam
16) Iran	33) Oman	50) Yemen
17) Iraq	34) Pakistan	

#? **19**

The following information has been given:

Japan

Therefore TZ='Asia/Tokyo' will be used.
Is the above information OK?

1) Yes
2) No
#? **1**

Local time is now: Thu Jul 23 02:45:30 JST 2015.
Universal Time is now: Wed Jul 22 17:45:30 UTC 2015.
executing app post_install

Enter Call Agent

1) Cisco Unified Communications Manager (CUCM) -- default
2) Cisco Unified Communications Manager Express (CUCME)
#? **2**

```

Setting Call Agent to CUCME
executing app post_install done
Configuring the system. Please wait...
Changing owners and file permissions.
Tightening file permissions ...
Change owners and permissions complete.
INIT: Switching to runlevel: 4
INIT: Sending processes the TERM signal
==> Starting CDP
STARTED: ntp_startup.sh
STARTED: LDAP_startup.sh
STARTED: SQL_startup.sh
STARTED: dnwldr_startup.sh
STARTED: HTTP_startup.sh
STARTED: probe
STARTED: fndn_udins_wrapper
STARTED: superthread_startup.sh
STARTED: /usr/wfavvid/run-wfengine.sh
STARTED: /usr/bin/launch_ums.sh

```

Waiting 61 ...

```

IMPORTANT::
IMPORTANT::      Administrator Account Creation
IMPORTANT::
IMPORTANT:: Create an administrator account. With this account,
IMPORTANT:: you can log in to the Cisco Unity Express GUI and
IMPORTANT:: run the initialization wizard.
IMPORTANT::

```

```

Enter administrator user ID:
  (user ID): administrator
Enter password for administrator:
  (password):
Confirm password for administrator by reentering it:
  (password):

```

```

SYSTEM ONLINE
CUE#

```

Task 7.5 Integrating CUE with CUCME

Create a trigger for the voicemail service

```

CUE# conf term
Enter configuration commands, one per line. End with CNTL/Z.
CUE(config)# ccn trigger sip phonenumber 4500
Adding new trigger
CUE(config-trigger)# application voicemail
CUE(config-trigger)# enabled
CUE(config-trigger)# maxsessions 6
CUE(config-trigger)# end

```

Define voicemail users

```

CUE(config)# username br2phone1 create
CUE(config)# username br2phone2 create

```

Associate voicemail users with their phone number


```
CUE(config)# username br2phone1 phonenumber 4001
CUE(config)# username br2phone2 phonenumber 4002
```

Create mailboxes for voicemail users

```
CUE(config)# voice mailbox owner br2phone1
CUE(config-mailbox)# end
CUE(config)# voice mailbox owner br2phone2
CUE(config-mailbox)# end
```

Update the CUE module with the correct MWI numbers

```
CUE(config)# ccn application ciscoMWIapplication aa
Modifying existing application
CUE(config-application)# parameter "strMWI_OFF_DN" "4511"
CUE(config-application)# parameter "strMWI_ON_DN" "4510"
CUE(config-application)# exit
```

NOTE: On each of the BR2 phones, press the Messages button, and walk through the setup process to set the PIN to **12345**.

Task 7.6 CUE Customization

Create an Auto Attendant trigger

```
CUE(config)# ccn trigger sip phonenumber 4550
Adding new trigger
CUE(config-trigger)# application autoattendant
CUE(config-trigger)# enabled
CUE(config-trigger)# maxsession 2
CUE(config-trigger)# end
```

A test call can now be placed to **4550**. The auto attendant application should answer with the default greeting. To record an alternative greeting, Administration via Telephone (AvT) can be configured. AvT uses the Prompt Management application.

Create a Prompt Management trigger

```
CUE(config)# ccn trigger sip phonenumber 4551
Adding new trigger
CUE(config-trigger)# application promptmgmt
CUE(config-trigger)# enabled
CUE(config-trigger)# maxsessions 1
CUE(config-trigger)# end
```

NOTE: By default, users do not have permission to use AvT.

Add one of the BR2 phone users to the Administrator group

```
CUE(config)# groupname Administrators member br2phone1
CUE(config)# end
CUE# copy run star
```

From BR2 Phone 1, dial 4551. When prompted, enter the phone number of 4001 (followed by #) and a PIN of 12345 (followed by #). Use the AvT interface to record and activate an alternate greeting for the Auto Attendant function.

Call phone number 4550 from one of the BR2 phones, and confirm the opening greeting begins with the alternate greeting we recorded.

Module 9: Basic CUCM Configuration

Task 1.2 IP Address Assignment

NOTE: We don't want to activate the DHCP server on our CUCM servers until all of the phone configuration is ready to go. That way, the phones can obtain an IP address and be ready to download their configuration file.

Task 1.3 NTP

Navigate to Cisco Unified OS Administration screen on CUCM9-PUB1, and add an NTP server

Navigation > Cisco Unified OS Administration > Go > (Login) > Settings > NTP Servers

Make sure the Loopback 0 IP address on router HQ (10.10.32.1) is the only NTP server. In our example, 192.168.1.78 is configured as the NTP server. Therefore, we need to add 10.10.32.1 as an NTP server, and then remove 192.168.1.78 as an NTP server.

Add New

Hostname or IP Address: **10.10.32.1**

Save

NTP Server Configuration

Save

Status

Status: Ready

NTP Server Settings

Hostname or IP Address*

Save

*- indicates required item.

NTP Server Configuration

Add New
 Select All
 Clear All
 Delete Selected

Status

- Update successful
- 2 records found

NTP Server	
<input type="checkbox"/>	Hostname or IP Address
<input checked="" type="checkbox"/>	192.168.1.78
<input type="checkbox"/>	10.10.32.1

(Check 192.168.1.78) > Delete Selected > OK

Repeat on CUCM9-PUB2

Task 2.1 CUCM SIP (Model 9971) IP Phones

Navigate back to the Cisco Unified CM Administration screen on CUCM9-PUB1

Navigation > Cisco Unified CM Administration > Go > (Login)

Several tasks are performed to initially register the IP phones at the HQ site. These tasks include:

- Modify the default Cisco Unified CM Group
- Create an NTP reference to be used by SIP IP phones
- Modify the default Date/Time Group
- Modify the default Region (for use by HQ IP phones), and create two new ones (for the CUBE and BR2 gateways)
- Create Locations for the CUBE and BR2 trunks
- Create a basic set of Partitions and Calling Search Spaces
- Create a basic set of Media Resource Groups (MRGs) and Media Resource Group Lists (MRGLs)
- Modify the default Device Pool (for use by HQ IP phones), and create two new ones (for the CUBE and BR2 gateways)
- Create a user for each IP phone
- Create a SIP Profile
- Add HQ Phone 1
- Add HQ Phone 2

- Activate the Cisco DHCP Monitor service on CUCM9-PUB1

Modify the default Cisco Unified CM Group

System > Cisco Unified CM Group > Find > Default

Name: **SUB-PUB**

Selected Cisco Unified Communications Managers:

- **CM_192.168.1.72**
- **CM_CUCM9-PUB1**

Save

Ok

Reset

Reset

Close

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes links for System, Call Routing, Media Resources, Advanced Features, Device, Application, and User Management. The main heading is "Cisco Unified CM Group Configuration". Below this, there are action buttons: Save, Delete, Copy, Reset, Apply Config, and Add New. The "Status" section shows an "Update successful" message. The "Cisco Unified Communications Manager Group Information" section states "Cisco Unified Communications Manager Group: SUB-PUB (used by 8 devices)". The "Cisco Unified Communications Manager Group Settings" section shows the "Name*" field set to "SUB-PUB" and the "Auto-registration" checkbox checked. The "Cisco Unified Communications Manager Group Members" section shows two lists: "Available Cisco Unified Communications Managers" (empty) and "Selected Cisco Unified Communications Managers*" (containing "CM_192.168.1.72" and "CM_CUCM-PUB1").

Create an NTP reference to be used by SIP IP phones




System > Phone NTP Reference > Add New


IP Address: **10.10.32.1**

Mode: **Unicast**


Save

Phone NTP Reference Configuration

 Save  Delete  Add New

Status
 Add successful

Phone NTP Reference Information
IP Address*
Description
Mode*

 *- indicates required item.

[Modify the default Date/Time Group](#)

System > Date/Time Group > Find > CMLocal

Group Name: **HQ**

Time Zone: **(GMT-8:00) America/Los_Angeles**

Separator: **/ (slash)**

Date Format: **Y/M/D**

Time Format: **12-Hour**

Add Phone NTP References

(Check 10.10.32.1)

Add Selected

Save

Apply Config

OK

Date/Time Group Configuration

Save
 Delete
 Copy
 Reset
 Apply Config
 Add New

Status

- Update successful
- Click on the Reset button to have the changes take effect.

Date/Time Group Information

Date/Time Group: HQ (used by 8 devices)

Group Name*

Time Zone* Entries with + are compatible with [legacy phone loads](#)

Separator* (applies to Date Format only)

Date Format*

Time Format*

Phone NTP References for this Date/Time Group

Selected Phone NTP References**

*- indicates required item.
 **Selected Phone NTP References are ordered by highest priority

Modify the default Region (for use by HQ IP phones), and create two new ones (for the CUBE and BR2 gateways)

System > Region Information > Region > Find > Default

Name: HQ

Save

OK

System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help

Region Configuration

Related Links: [Back To Find/List](#)

Save
 Delete
 Reset
 Apply Config
 Add New

Status

- Update successful
- Click on the Reset button to have the changes take effect.

Region Information

Name*

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
HQ	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	384
NOTE: Regions not displayed Use System Default Use System Default Use System Default			

Modify Relationship to other Regions

Regions	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
HQ	<input type="text" value="Keep Current Setting"/>	<input type="text" value="Keep Current Setting"/>	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text" value=""/> kbps

Similarly, create Regions with the following names:

- CUBE
- BR2

We should now have three Regions defined:

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Find and List Regions

[Add New](#)
[Select All](#)
[Clear All](#)
[Delete Selected](#)

Status
i 3 records found

Regions (1 - 3 of 3) Rows per Page 50 ▾

Find Regions where Name Find [Clear Filter](#) [+](#) [-](#)

	Name ^
<input type="checkbox"/>	BR2
<input type="checkbox"/>	CUBE
<input type="checkbox"/>	HQ

[Add New](#)
[Select All](#)
[Clear All](#)
[Delete Selected](#)

The Regions should now be configured to use G.711 within a Region (e.g. a call from one HQ IP phone to another HQ IP phone) and to use iLBC between Regions (e.g. a call from an HQ IP phone to a BR1 IP phone).

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Region Configuration

Related Links: [Back To Find/List](#) [Go](#)

[Save](#)
[Delete](#)
[Reset](#)
[Apply Config](#)
[Add New](#)

Region Information
 Name*

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
BR2	Use System Default (Factory Default low loss)	16 kbps (iLBC, G.728)	384
CUBE	Use System Default (Factory Default low loss)	16 kbps (iLBC, G.728)	384
HQ	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	384

NOTE: Regions not displayed Use System Default Use System Default Use System Default

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Region Configuration

Related Links: [Back To Find/List](#) [Go](#)

[Save](#)
[Delete](#)
[Reset](#)
[Apply Config](#)
[Add New](#)

Status
i Update successful
i Click on the Reset button to have the changes take effect.

Region Information
 Name*

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
BR2	Use System Default (Factory Default low loss)	16 kbps (iLBC, G.728)	384
CUBE	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	384
HQ	Use System Default (Factory Default low loss)	16 kbps (iLBC, G.728)	384

NOTE: Regions not displayed Use System Default Use System Default Use System Default

System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help

Region Configuration

Related Links: [Back To Find/List](#) [Go](#)

Save Delete Reset Apply Config Add New

Status

- Update successful
- Click on the Reset button to have the changes take effect.

Region Information

Name*

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
BR2	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	384
CUBE	Use System Default (Factory Default low loss)	16 kbps (ILBC, G.728)	384
HQ	Use System Default (Factory Default low loss)	16 kbps (ILBC, G.728)	384

NOTE: Regions not displayed Use System Default Use System Default Use System Default

Create Locations for the CUBE and BR2 trunks

NOTE: The HQ site will use the **Hub_None** Location.

System > Location > Add New

Name: **CUBE**

Save

System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help

Location Configuration

Related Links: [Back To Find/List](#) [Go](#)

Save Delete Copy Add New

Status

- Add successful

Location Information

Name*

Links - Bandwidth Between CUBE and Adjacent Locations

Locations (1 - 1 of 1) Rows per Page 50

Find Locations where name begins with Find Clear Filter

Location	Weight	Audio Bandwidth	Video Bandwidth	Immersive Bandwidth
Hub_None	50	UNLIMITED	384	384

Add Select All Clear All Delete Selected

[Show Advanced](#)

Location RSVP Settings

Location	RSVP Setting
CUBE	No Reservation

NOTE: Location(s) not displayed Use System Default

Modify Setting(s) to Other Locations

Location	RSVP Setting
CUBE Hub_None Phantom Shadow	Use System Default

Create another Location for **BR2**.

Create a basic set of Partitions and Calling Search Spaces

Although subsequent tasks might require additional Partitions and Calling Search Spaces, we get started with a common set. First, the following Partitions are created:

- **INTERNAL** (for internal directory numbers)
- **HQ** (for Translation Patterns used by HQ IP phones)
- **CUBE** (for Translation Patterns used by the CUBE)
- **BR2** (for Translation Patterns used by the BR2 gateway)
- **PSTN** (for the \+! Route Pattern used by Translation Patterns)
- **STRIP** (for the Called Party Transformation Pattern that strips a leading + from a DNIS string)
- **HQ-DNIS-OUT** (for Called Party Transformation Patterns used by the HQ gateway)
- **HQ-ANI-IN** (for Calling Party Transformation Patterns used by HQ IP phones)

The following set of Calling Search Spaces should then be created and populated with the Partitions as follows:

Calling Search Space	Partition(s)
INTERNAL	INTERNAL
HQ	INTERNAL HQ
CUBE	INTERNAL CUBE
BR2	INTERNAL BR2
PSTN	PSTN
HQ-DNIS-OUT	STRIP HQ-DNIS-OUT
HQ-ANI-IN	HQ-ANI-IN

Call Routing > Class of Control > Partition

Add New

Name:

INTERNAL
HQ
CUBE
BR2
PSTN
STRIP
HQ-DNIS-OUT
HQ-ANI-IN

Save

Find

System

Call Routing

Media Resources

Advanced Features

Device

Application

User Management

Bulk Administration

Help

Find and List Partitions

Add New

Select All

Clear All

Delete Selected

Status

9 records found

Partition (1 - 9 of 9)

Rows per Page50

Find Partition where

Name

begins with

Find

Clear Filter

	Partition Name ^	Description
<input type="checkbox"/>	BR2	BR2
<input type="checkbox"/>	CUBE	CUBE
<input type="checkbox"/>	Directory URI	
<input type="checkbox"/>	HQ	HQ
<input type="checkbox"/>	HQ-ANI-IN	HQ-ANI-IN
<input type="checkbox"/>	HQ-DNIS-OUT	HQ-DNIS-OUT
<input type="checkbox"/>	INTERNAL	INTERNAL
<input type="checkbox"/>	PSTN	PSTN
<input type="checkbox"/>	STRIP	STRIP

Add New

Select All

Clear All

Delete Selected

Call Routing > Class of Control > Calling Search Space > Add New

Name: **INTERNAL**

Selected Partitions: **INTERNAL**

Save

Similarly, create the remaining partitions in the previous table.

Create a basic set of [Media Resource Groups \(MRGs\)](#) and [Media Resource Group Lists \(MRGLs\)](#)

Media Resources > Media Resource Group > Add New

Name: **HQ**

Selected Media Resources:

ANN_2

ANN_3

MTP_2

MTP_3

CFB_2

CFB_3


MOH_2

MOH_3


Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Media Resource Group Configuration

 Save

Status

 Status: Ready

Media Resource Group Status

Media Resource Group: New

Media Resource Group Information

Name*


Description


Devices for this Group


Available Media Resources**

Selected Media Resources*

☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

 Save

 *- indicates required item.

 **Includes Annunciators (ANN), Conference Bridges (CFB), Media Termination Points (MTP), Music On Hold Servers (MOH) and Transcoders (XCODE)


Copy


Name: **CUBE**

Save


System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Media Resource Group Configuration

Related Links: [Back To Find/List](#) 

 Save

Status

 Status: Ready

Media Resource Group Status

Media Resource Group: New (Copy of) HQ

Media Resource Group Information

Name*

Description

Devices for this Group

Available Media Resources**

Selected Media Resources*

☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

Copy

Name: **BR2**

Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Media Resource Group Configuration Related Links: [Back To Find/List](#) [Go](#)

Save Delete Copy Add New

Status

- Add successful
- Devices associated with this group must only be reset if the Name or Multi-cast option has changed. Use caution because multiple device resets may impact call processing.

Media Resource Group Status

Media Resource Group: BR2 (used by 0 devices)

Media Resource Group Information

Name*

Description

Devices for this Group

Available Media Resources**

Selected Media Resources*

☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

Media Resources > Media Resource Group List > Add New

Name: HQ

Selected Media Resource Groups: HQ

Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Media Resource Group List Configuration

Save

Status

Status: Ready

Media Resource Group List Status

Media Resource Group List: New

Media Resource Group List Information

Name*

Media Resource Groups for this List

Available Media Resource Groups

Selected Media Resource Groups

Copy

Name: CUBE

Selected Media Resource Groups: CUBE

Save

System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help

Media Resource Group List Configuration Related Links: [Back To Find/List](#) [Go](#)

Save Delete Copy Add New

Status
Add successful

Media Resource Group List Status
Media Resource Group List: CUBE (used by 0 devices)

Media Resource Group List Information
Name: CUBE

Media Resource Groups for this List
Available Media Resource Groups: BR2
HQ
Selected Media Resource Groups: CUBE

Copy

Name: **BR2**

Selected Media Resource Groups: **BR2**

Save

System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help

Media Resource Group List Configuration Related Links: [Back To Find/List](#) [Go](#)

Save Delete Copy Add New

Status
Add successful

Media Resource Group List Status
Media Resource Group List: BR2 (used by 0 devices)

Media Resource Group List Information
Name: BR2

Media Resource Groups for this List
Available Media Resource Groups: CUBE
HQ
Selected Media Resource Groups: BR2

Modify the default Device Pool (for use by HQ IP phones), and create two new ones (for the CUBE and BR2 gateways)

System > Device Pool > Find > Default

Device Pool Name: **HQ**

Date/Time Group: **HQ**

Region: **HQ**

Media Resource Group List: **HQ**

Location: **Hub_None**

Save

Apply Config

OK

System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management

Device Pool Configuration

Save
Delete
Copy
Reset
Apply Config
Add New

Status
i Status: Ready

Device Pool Information
Device Pool: Default (8 members**)

Device Pool Settings
Device Pool Name* HQ
Cisco Unified Communications Manager Group* SUB-PUB
Calling Search Space for Auto-registration < None >
Adjunct CSS < None >
Reverted Call Focus Priority Default
Intercompany Media Services Enrolled Group < None >

Roaming Sensitive Settings
Date/Time Group* HQ
Region* HQ
Media Resource Group List HQ
Location Hub_None
Network Locale < None >
SRST Reference* Disable

Copy

Device Pool Name: **CUBE**

Date/Time Group: **HQ**

Region: **CUBE**

Media Resource Group List: **CUBE**

Location: **CUBE**







Save

Apply Config



OK

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾

Device Pool Configuration

 Save
  Delete
  Copy
  Reset
  Apply Config
  Add New


Status

-  Add successful
-  Click on the Reset button to have the changes take effect.

Device Pool Information

Device Pool: CUBE (0 members**)

Device Pool Settings

Device Pool Name*	CUBE	
Cisco Unified Communications Manager Group*	SUB-PUB	⌵
Calling Search Space for Auto-registration	< None >	⌵
Adjunct CSS	< None >	⌵
Reverted Call Focus Priority	Default	⌵
Intercompany Media Services Enrolled Group	< None >	⌵

Roaming Sensitive Settings

Date/Time Group*	HQ	⌵
Region*	CUBE	⌵
Media Resource Group List	CUBE	⌵
Location	CUBE	⌵
Network Locale	< None >	⌵
SRST Reference*	Disable	⌵

Copy

Device Pool Name: **BR2**

Date/Time Group: **HQ**

Region: **BR2**

Media Resource Group List: **BR2**

Location: **BR2**

Save

Apply Config

OK

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾

Device Pool Configuration

Save
 Delete
 Copy
 Reset
 Apply Config
 Add New

Status

- Add successful
- Click on the Reset button to have the changes take effect.

Device Pool Information

Device Pool: BR2 (0 members**)

Device Pool Settings

Device Pool Name*

Cisco Unified Communications Manager Group*

Calling Search Space for Auto-registration

Adjunct CSS

Reverted Call Focus Priority

Intercompany Media Services Enrolled Group

Roaming Sensitive Settings

Date/Time Group*

Region*

Media Resource Group List

Location

Network Locale

SRST Reference*

Create a user for each IP phone

User Management > End User > Add New

User ID: **hqphone1**

Password: **cisco**

Confirm Password: **cisco**

PIN: **1234**

Confirm PIN: **1234**


Last name: **Phone 1**


First name: **HQ**

Save






System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

End User Configuration

 Save

Status
 Status: Ready




User Information

User Status	Active Local User
User ID *	hphone1 
Password 
Confirm Password 
PIN 
Confirm PIN 
Last name *	Phone 1
Middle name	
First name	HQ

Add to Access Control Group > Find
 (Check **Standard CCM End Users**)
 (Check **Standard CTI Enabled**)
Add Selected
Save


System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾


End User Configuration

 Save  Delete  Add New

Multilevel Precedence and Preemption Authorization

MLPP User Identification Number

MLPP Password 

Confirm MLPP Password 

MLPP Precedence Authorization Level

CAPF Information


Associated CAPF Profiles [View Details](#)

Permissions Information





Groups [View Details](#)

Roles [View Details](#)

[Add to Access Control Group](#) [Remove from Access Control Group](#)

 *- Indicates required item.

Find and List Access Control Groups

 Select All  Clear All  Add Selected  Close

- ☒ Standard CCM End Users
- ☐ Standard CCM Gateway Administration
- ☐ Standard CCM Phone Administration
- ☐ Standard CCM Read Only
- ☐ Standard CCM Server Maintenance
- ☐ Standard CCM Server Monitoring
- ☐ Standard CCM Super Users
- ☐ Standard CTI Allow Call Monitoring
- ☐ Standard CTI Allow Call Park Monitoring
- ☐ Standard CTI Allow Call Recording
- ☐ Standard CTI Allow Calling Number Modification
- ☐ Standard CTI Allow Control of All Devices
- ☐ Standard CTI Allow Control of Phones supporting Connected Xfer and conf
- ☐ Standard CTI Allow Control of Phones supporting Rollover Mode
- ☐ Standard CTI Allow Reception of SRTP Key Material
- ☒ Standard CTI Enabled

Repeat this step to create a user for **HQ Phone 2**.

Create a SIP Profile

Device > Device Settings > SIP Profile > Find > Standard SIP Profile > Copy

Name: **HQ**
Save

Add HQ Phone 1

Make sure there are not existing directory numbers (DNs) that match the DNs we are about to add to our new IP phones.

Call Routing > Directory Number > Find

In this case, we don't have any duplicate DNs. However, if we did, we should delete them.

Next, we need to get the MAC addresses for the SIP and SCCP IP phones we are about to add. To prevent mistyping, let's copy/paste them from switch SW1.

```
SW1#show cdp neigh
Capability Codes: R - Router, T - Trans Bridge, B - Source Route Bridge
                  S - Switch, H - Host, I - IGMP, r - Repeater, P - Phone,
                  D - Remote, C - CVTA, M - Two-port Mac Relay

Device ID         Local Intrfce   Holdtme    Capability Platform Port ID
HQ.kwtrain.com    Fas 1/0/24      137        R S I     CISCO2911 Gig 0/0
CUCM9-SUB1        Fas 1/0/1       141        H         VMware   eth0
CUCM9-PUB1        Fas 1/0/1       120        H         VMware   eth0
SEP0CD996912474   Fas 1/0/7       174        H P M     IP Phone  Port 1
SEP0CC8821098E9   Fas 1/0/8       175        H P M     IP Phone  Port 1
PSTN.yourdomain.com
                  Fas 1/0/2       137        R S I     CISCO2911 Gig 0/0
```

Device > Phone > Add New > Phone Type > Cisco 9971 > Next

MAC Address: **0CD996912474** (Pasted from switch SW1)

Description: **HQ Phone 1**

Device Pool: **HQ**

Phone Button Template: **Standard 9971 SIP**

Calling Search Space: **HQ**

Owner User ID: **hqphone1**

Device Security Profile: **Cisco 9971 – Standard SIP Non-Secure Profile**

SIP Profile: **Standard SIP Profile**

Cisco Camera: **Enabled**

Video Capabilities: **Enabled**

Save

OK

System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help

Phone Configuration

Save

Status

Status: Ready

Phone Type

Product Type: Cisco 9971
Device Protocol: SIP

Device Information

☒ Device is trusted

MAC Address*	0CD996912474	
Description	HQ Phone 1	
Device Pool*	HQ	View Details
Common Device Configuration	< None >	View Details
Phone Button Template*	Standard 9971 SIP	
Common Phone Profile*	Standard Common Phone Profile	
Calling Search Space	HQ	
AAR Calling Search Space	< None >	
Media Resource Group List	< None >	
User Hold MOH Audio Source	< None >	
Network Hold MOH Audio Source	< None >	
Location*	Hub_None	
AAR Group	< None >	
User Locale	< None >	
Network Locale	< None >	
Built In Bridge*	Default	
Privacy*	Default	
Device Mobility Mode*	Default	View Current Device Mobility Settings
Owner	<input checked="" type="radio"/> User <input type="radio"/> Anonymous (Public/Shared Space)	
Owner User ID*	hqphone1	
Phone Personalization*	Default	
Services Provisioning*	Default	
Phone Load Name		
Use Trusted Relay Point*	Default	
BLF Audible Alert Setting (Phone Idle)*	Default	
BLF Audible Alert Setting (Phone Busy)*	Default	
Always Use Prime Line*	Default	
Always Use Prime Line for Voice Message*		

Use Trusted Relay Point*	Default
BLF Audible Alert Setting (Phone Idle)*	Default
BLF Audible Alert Setting (Phone Busy)*	Default
Always Use Prime Line*	Default
Always Use Prime Line for Voice Message*	Default
Geolocation	< None >
Feature Control Policy	< None >
<input type="checkbox"/> Ignore Presentation Indicators (internal calls only) <input checked="" type="checkbox"/> Allow Control of Device from CTI <input checked="" type="checkbox"/> Logged Into Hunt Group <input type="checkbox"/> Remote Device <input type="checkbox"/> Protected Device**** <input type="checkbox"/> Require off-premise location	

Number Presentation Transformation

Caller ID For Calls From This Phone
 Calling Party Transformation CSS: < None >
☒ Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)

Remote Number
 Calling Party Transformation CSS: < None >
☒ Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

Protocol Specific Information

Packet Capture Mode*	None
Packet Capture Duration	0
BLF Presence Group*	Standard Presence group
SIP Dial Rules	< None >
MTP Preferred Originating Codec*	711ulaw
Device Security Profile*	Cisco 9971 - Standard SIP Non-Secure Profile
Rerouting Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile
Digest User	< None >

Cisco Camera*	Enabled	<input checked="" type="checkbox"/>
Video Capabilities*	Enabled	<input checked="" type="checkbox"/>

OK

Line [1] – Add a new DN

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Phone Configuration
Related Links: [Back To Find/List](#) Go

Save Delete Copy Reset Apply Config Add New

Status
 Add successful

Association Information
 Modify Button Items
 1 Line [1] - Add a new DN
 2 Line [2] - Add a new DN
 3 Add a new SD
 4 Add a new SD
 5 Add a new SD
 6 Add a new SD
 ----- Unassigned Associated Items -----

Phone Type
 Product Type: Cisco 9971
 Device Protocol: SIP

Device Information


Registration	Rejected
IP Address	10.10.120.19
Active Load ID	Unknown
Inactive Load ID	Unknown
Download Status	Unknown
<input checked="" type="checkbox"/> Device is Active	
<input checked="" type="checkbox"/> Device is trusted	

Directory Number: 2001


Route Partition: **INTERNAL**
Description: **HQ Phone 1**
Alerting Name: **HQ Phone 1**
ASCII Alerting Name: **HQ Phone 1**

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾


Directory Number Configuration

 Save

Status

 Directory Number Configuration has refreshed due to a directory number change

Directory Number Information

Directory Number* 

Route Partition

Description

Alerting Name

ASCII Alerting Name

☒ Active

Calling Search Space Activation Policy: **With Configured CSS**
Forward Busy Internal: **Voice Mail Check Box Checked with a CSS of HQ**
Forward Busy External: **Voice Mail Check Box Checked with a CSS of HQ**
Forward No Answer Internal: **Voice Mail Check Box Checked with a CSS of HQ**
Forward No Answer External: **Voice Mail Check Box Checked with a CSS of HQ**
Forward Unregistered Internal: **Voice Mail Check Box Checked with a CSS of HQ**
Forward Unregistered External: **Voice Mail Check Box Checked with a CSS of HQ**
No Answer Ring Duration: **10**

Call Forward and Call Pickup Settings

	Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy			
Forward All	<input type="checkbox"/> or		<input type="text" value="With Configured CSS"/>
Secondary Calling Search Space for Forward All			<input type="text" value="< None >"/>
Forward Busy Internal	<input checked="" type="checkbox"/> or		<input type="text" value="HQ"/>
Forward Busy External	<input checked="" type="checkbox"/> or		<input type="text" value="HQ"/>
Forward No Answer Internal	<input checked="" type="checkbox"/> or		<input type="text" value="HQ"/>
Forward No Answer External	<input checked="" type="checkbox"/> or		<input type="text" value="HQ"/>
Forward No Coverage Internal	<input type="checkbox"/> or		<input type="text" value="< None >"/>
Forward No Coverage External	<input type="checkbox"/> or		<input type="text" value="< None >"/>
Forward on CTI Failure	<input type="checkbox"/> or		<input type="text" value="< None >"/>
Forward Unregistered Internal	<input checked="" type="checkbox"/> or		<input type="text" value="HQ"/>
Forward Unregistered External	<input checked="" type="checkbox"/> or		<input type="text" value="HQ"/>
No Answer Ring Duration (seconds)	<input type="text" value="10"/>		
Call Pickup Group	<input type="text" value="< None >"/>		

Display (Caller ID): **HQ Phone 1**

ASCII Display (Caller ID): **HQ Phone 1**
External Phone Number Mask: +1408222XXXX
Busy Trigger: 1
Save

Line 1 on Device SEP0CD996912474

Display (Caller ID)	HQ Phone 1	Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Caller ID)	HQ Phone 1	
Line Text Label		
ASCII Line Text Label		
External Phone Number Mask	+1408222XXXX	
Visual Message Waiting Indicator Policy*	Use System Policy	
Audible Message Waiting Indicator Policy*	Default	
Ring Setting (Phone Idle)*	Use System Default	
Ring Setting (Phone Active)	Use System Default	Applies to this line when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting(Phone Idle)	Use System Default	
Call Pickup Group Audio Alert Setting(Phone Active)	Use System Default	
Recording Option*	Call Recording Disabled	
Recording Profile	< None >	
Monitoring Calling Search Space	< None >	
<input checked="" type="checkbox"/> Log Missed Calls		

Multiple Call/Call Waiting Settings on Device SEP0CD996912474

Note: The range to select the Max Number of calls is: 1-200

Maximum Number of Calls* 4

Busy Trigger* 1 (Less than or equal to Max. Calls)

Task 2.2 CUCM SCCP (Model 7965) IP Phones

Add HQ Phone 2

With all of the underlying configuration (e.g. Device Pools, Locations, Regions, etc.) already defined, we can quickly add HQ Phone 2, making modifications for the different IP phone model (i.e. Cisco 7965 IP Phone using SCCP).

Now that both the HQ Phone 1 and HQ Phone 2 phones have been added to the CUCM database for the HQ site, let's configure DHCP on the CUCM9-PUB1 server. This will allow the IP phones obtain IP address information and register using their newly configured configurations.

Navigation > Cisco Unified CM Administration > Go > (Login)

Configure CUCM9-PUB1 as a DHCP server

System > DHCP > DHCP Server > Add New

Host Server: **192.168.1.71**

Primary TFTP Server IPv4 Address (Option 150): **192.168.1.71**

Secondary TFTP Server IPv4 Address (Option 150): **192.168.1.72**

Save

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾

DHCP Server Configuration

Save

Status
Status: Ready

DHCP Server Information

Host Server *	192.168.1.71
Primary DNS IPv4 Address	
Secondary DNS IPv4 Address	
Primary TFTP Server IPv4 Address (Option 150)	192.168.1.71
Secondary TFTP Server IPv4 Address (Option 150)	192.168.1.72
Bootstrap Server IPv4 Address	
Domain Name	
TFTP Server Name (Option 66)	
ARP Cache Timeout (sec) *	0
IP Address Lease Time (sec) *	0
Renewal (T1) Time (sec) *	0
Rebinding (T2) Time (sec) *	0

Define the subnet for HQ IP phones

System > DHCP > DHCP Subnet > Add New

DHCP Server: **192.168.1.71**

Subnet IPv4 Address: **10.10.120.0**

Primary Start IPv4 Address: **10.10.120.10**

Primary End IPv4 Address: **10.10.120.20**

Primary Router IPv4 Address: **10.10.120.1**

IPv4 Subnet Mask: **255.255.255.0**

Save

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

DHCP Subnet Configuration

Save

Status
Status: Ready

DHCP Subnet Information

DHCP Server*	192.168.1.71
Subnet IPv4 Address*	10.10.120.0
Primary Start IPv4 Address*	10.10.120.10
Primary End IPv4 Address*	10.10.120.20
Secondary Start IPv4 Address	
Secondary End IPv4 Address	
Primary Router IPv4 Address	10.10.120.1
Secondary Router IPv4 Address	
IPv4 Subnet Mask*	255.255.255.0
Domain Name	

Start the DHCP service on CUCM9-PUB1

Navigation > Cisco Unified Serviceability > Go > (Login) > Tools > Service Activation > 192.168.1.71 > Go > (Check Cisco DHCP Monitor Service) > Save > OK

Cisco Unified Serviceability
For Cisco Unified Communications Solutions

Alarm ▾ Trace ▾ Tools ▾ Snmp ▾ CallHome ▾ Help ▾

Service Activation Related Links:

Save Set to Default Refresh

Status:
Ready

Select Server

Server* 192.168.1.71 Go

☐ Check All Services

CM Services		
	Service Name	Activation Status
<input checked="" type="checkbox"/>	Cisco CallManager	Activated
<input checked="" type="checkbox"/>	Cisco Messaging Interface	Activated
<input checked="" type="checkbox"/>	Cisco Unified Mobile Voice Access Service	Activated
<input checked="" type="checkbox"/>	Cisco IP Voice Media Streaming App	Activated
<input checked="" type="checkbox"/>	Cisco CTIManager	Activated
<input checked="" type="checkbox"/>	Cisco Extension Mobility	Activated
<input checked="" type="checkbox"/>	Cisco Extended Functions	Activated
<input checked="" type="checkbox"/>	Cisco DHCP Monitor Service	Deactivated
<input checked="" type="checkbox"/>	Cisco Intercluster Lookup Service	Activated
<input checked="" type="checkbox"/>	Cisco Location Bandwidth Manager	Activated
<input checked="" type="checkbox"/>	Cisco Dialed Number Analyzer Server	Activated

Next, repeat the previous procedure to register the BR1 Phone 1 and BR1 Phone 2 phones.

Task 2.3 CUCM IP Phone Customization

We've already set an Enterprise Parameter that disabled to advertisement of the G.722 codec. Also, the BR1 region (defined on the BR1 CUCM cluster) allows 64 kbps calls. At this point, the G.711ulaw codec will be used between BR1 IP phones.

To change this behavior, we can create a new Audio Codec Preference List. To do this, we'll modify the existing default Audio Codec Preference List of **Factory Default low loss**.

Create a new Audio Codec Preference List

System > Region Information > Audio Codec Preference List > Find > Factor Default low loss > Copy

Name: **CCIE**

(Reorder the codecs such that G.711 A-Law 64k is above G.711 U-Law 64k.)

Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Audio Codec Preference List Configuration

Save

Status

Status: Ready

Audio Codec Preference List Information

Name*

Description*

Codecs in List*

AMR-WB (7k-24k)	
AMR (5k-13k)	
MP4A-LATM 128k	
AAC-LD (MP4A Generic)	
MP4A-LATM 64k	
MP4A-LATM 56k	
L16 256k	
MP4A-LATM 48k	
G.722 64k	
ISAC 32k	
MP4A-LATM 32k	
G.722.1 32k	
G.722 56k	
G.722.1 24k	
G.722 48k	
MP4A-LATM 24k	
G.711 U-Law 64k	
G.711 A-Law 64k	
G.711 U-Law 56k	
G.711 A-Law 56k	
ILBC 16k	

Apply the new Audio Codec Preference List to the BR1 Region

**System > Region Information > Region > Find > BR1
(Select BR1)**

Audio Codec Preference List: **CCIE**

Save

Apply Config

OK

Modify Relationship to other Regions

Regions	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
BR1 BR2 CUBE	CCIE	Keep Current Setting	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text"/> kbps

Module 10: H.323 GW, MGCP GW, and SIP Trunk Configuration

Task 3.1 HQ Cisco IOS H.323 T1-PRI Gateway

Router HQ has already been configured as an H.323 gateway. However, this gateway needs to be added to CUCM.

Device > Gateway > Add New

Gateway Type: **H.323 Gateway**

Next

System > Call Routing > Media Resources > Advanced Features > Device

Add a new Gateway

Next

Select the type of gateway you would like to add:

Gateway Type* H.323 Gateway

Next

*- indicates required item.

Device Name: **10.10.120.1**

NOTE: This is the IP address of the Gig 0/0.200 subinterface.

Device Pool: **HQ**

Significant Digits: **4**

Calling Search Space: **HQ**

Redirecting Number IE Delivery – Inbound: **Checked**

Enable Inbound FastStart: **Checked**

NOTE: Be careful not to enable outbound FastStart (unless you're sure you want to).


Display IE Delivery: **Checked**

Redirecting Number IE Delivery – Outbound: **Checked**


Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾



Gateway Configuration

 Save

Status

 Status: Ready

Device Information

Product	H.323 Gateway
Device Protocol	H.225
 Device is not trusted	
Device Name*	10.10.120.1 
Description	10.10.120.1
Device Pool*	HQ
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Packet Capture Mode*	None
Packet Capture Duration	0
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Use Trusted Relay Point*	Default
Signaling Port*	1720

☐ Media Termination Point Required

☒ Retry Video Call As Audio

☒ Wait for Far End H.245 Terminal Capability Set

☐ Path Replacement Support

☐ Transmit UTF-8 for Calling Party Name

☐ SRTP Allowed - When this flag is checked, IPSec needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

☐ H.235 Pass Through Allowed

☒ PSTN Access

Multilevel Precedence and Preemption (MLPP) Information	
MLPP Domain	< None >
MLPP Indication	Not available on this device
MLPP Preemption	Not available on this device

Call Routing Information - Inbound Calls	
Significant Digits*	4
Calling Search Space	HQ
AAR Calling Search Space	< None >
Prefix DN	
<input checked="" type="checkbox"/> Redirecting Number IE Delivery - Inbound	
<input checked="" type="checkbox"/> Enable Inbound FastStart	

Call Routing Information - Outbound Calls	
Calling Party Selection*	Originator
Calling Party Presentation*	Default
Called party IE number type unknown*	Cisco CallManager
Calling party IE number type unknown*	Cisco CallManager
Called Numbering Plan*	Cisco CallManager
Calling Numbering Plan*	Cisco CallManager
Caller ID DN	
<input checked="" type="checkbox"/> Display IE Delivery	
<input checked="" type="checkbox"/> Redirecting Number IE Delivery - Outbound	
Redirecting Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Redirecting Party Transformation CSS	
<input type="checkbox"/> Enable Outbound FastStart	
Codec For Outbound FastStart	G711 u-law 64K
Called Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	

SUPER IMPORTANT POINT!

Even though you've configured everything perfectly on the H.323 gateway and on CUCM, it might not work unless you RESET the gateway in CUCM.

Reset

Reset

Close

Task 3.2 BR1 Cisco IOS MGCP T1-PRI Gateway

Router BR1 will be configured as an MGCP gateway using a hybrid approach, meaning that the gateway will first be configured in CUCM. Then, the router will be configured to download its configuration from CUCM. At this point, if we make any changes to the gateway configuration on the router, those changes will be overridden if we reset the gateway from CUCM. So, we'll break those ties that bind the CUCM MGCP gateway configuration with the Cisco IOS router's MGCP gateway configuration. Finally, we'll make required customizations to the gateway in Cisco IOS.

Add the MGCP gateway to CUCM.

Device > Gateway > Add New

Gateway Type: **Cisco 2911**

Next

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾

Add a new Gateway

Next

Select the type of gateway you would like to add:

Gateway Type* Cisco 2911

Next

*- indicates required item.

Protocol: **MGCP**

Next

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Add a new Gateway

Next

Select the type of gateway you would like to add:

Gateway Type Cisco 2911 Change Gateway type

Protocol* MGCP

Next

*- indicates required item.

On router BR1, we need to determine if there is a domain name assigned to the router. If there is, the domain name will be prefixed to the router's hostname and entered into the **Domain Name** field of the MGCP gateway configuration screen in CUCM. However, if the router does not have a domain name assigned, the router's hostname will be entered as the gateway's Domain Name.

Determine if a domain name is assigned to the router:

```
BR1#show run | i domain
no ip domain lookup
ip domain name kwtrain.com
BR1#
```

In this instance, the router has a domain name of kwtrain.com. As a result, the router's hostname of BR1 is prepended to the domain name (i.e. **BR1.kwtrain.com**) and entered into the Domain Name field.

Domain Name: **BR1.kwtrain.com**
Cisco Unified Communications Manager Group: **BR1**

The **show inventory** command can be used to determine the slot and subslot of the T1/E1 controller.

```
BR1#show inventory
NAME: "CISCO2911/K9", DESCR: "CISCO2911/K9 chassis, Hw Serial#:
FTX1726AL6K, Hw Revision: 1.0"
PID: CISCO2911/K9      , VID: V07 , SN: FTX1726AL6K

NAME: "VWIC2-2MFT-T1/E1 - 2-Port RJ-48 Multiflex Trunk - T1/E1 on Slot
0 SubSlot 0", DESCR: "VWIC2-2MFT-T1/E1 - 2-Port RJ-48 Multiflex Trunk
- T1/E1"
PID: VWIC2-2MFT-T1/E1  , VID: V01 , SN: FOC11160M6C

NAME: "WAN Interface Card - HWIC Serial 1T on Slot 0 SubSlot 1",
DESCR: "WAN Interface Card - HWIC Serial 1T"
PID: HWIC-1T           , VID: V02 , SN: FOC14092KZN

NAME: "4 Port FE Switch on Slot 0 SubSlot 2", DESCR: "4 Port FE
Switch"
PID: HWIC-4ESW         , VID: VN/A, SN: FOC10150TYW

NAME: "PVDM3 DSP DIMM with 16 Channels on Slot 0 SubSlot 4", DESCR:
"PVDM3 DSP DIMM with 16 Channels"
PID: PVDM3-16          , VID: V01 , SN: FOC17235E3M

NAME: "PVDM3 DSP DIMM with 16 Channels on Slot 0 SubSlot 5", DESCR:
"PVDM3 DSP DIMM with 16 Channels"
PID: PVDM3-16          , VID: V01 , SN: FOC17244F0V

NAME: "C2911 AC Power Supply", DESCR: "C2911 AC Power Supply"
PID: PWR-2911-AC        , VID: V05 , SN: DCA1708R0N6
```

The output of the **show inventory** command reveals that the T1/E1 voice interface is installed in **Slot 0, SubSlot 0**. Slot 0 represents the router's motherboard. Also, this is a two-port T1/E1 voice interface, meaning that **the first port number is 0**. Therefore, the controller identifier is **0/0/0** (i.e. **slot 0, subslot 0, port 0**).

Module in Slot 0: **NM-4VWIC-MBRD**

Global ISDN Switch Type: **NI2**

Save

The output of the **show inventory** command indicated the product ID of the T1 voice interface is **VWIC2-2MFT-T1/E1**.

Subunit 0: **VWIC2-2MFT-T1/E1-T1**

Save

CUCM has now determined the T1 controller identifier to be **0/0/0**. The question mark icon on the T1 controller's icon indicates that the controller is not configured. So, we click that icon to start configuring the T1 controller as an MGCP endpoint.

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Gateway Configuration

Save Delete Reset Apply Config Add New

Status

Status: Ready

Gateway Details

Product	Cisco 2911
Gateway	BR1.kwtrain.com
Protocol	MGCP
Device is not trusted	
Domain Name*	BR1.kwtrain.com
Description	BR1.kwtrain.com
Cisco Unified Communications Manager Group*	BR1

Configured Slots, VICs and Endpoints

Module in Slot 0 **NM-4VWIC-MBRD**

Subunit 0 **VWIC2-2MFT-T1E1-T1** 0/0/0 0/0/1

Subunit 1 **< None >**

Subunit 2 **< None >**

Subunit 3 **< None >**

Module in Slot 1 **< None >**

Product Specific Configuration Layout


Global ISDN Switch Type **NI2**

Device Protocol: **Digital Access PRI**

Next

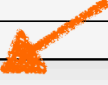
System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾


Find and List Gateway


 Next

Select Protocol for this Gateway

Device Protocol*



 Next

 *- indicates required item.

Device Pool: **BR1**

Channel Selection Order: **Top down**

Significant Digits: **4**

Calling Search Space: **BR1**

Display IE Delivery: **Checked**

Redirecting Number IE Delivery – Outbound: **Checked**

Redirecting Number IE Delivery – Inbound: **Checked**

Save

Gateway Configuration





Save

Status



Status: Ready

Device Information

Product	Cisco MGCP T1 Port
Gateway	BR1.kwtrain.com
Device Protocol	Digital Access PRI
⚠ Device is not trusted	
End-Point Name *	S0/SU0/DS1-0@BR1.kwtrain.com
Description	S0/SU0/DS1-0@BR1.kwtrain.com 
Device Pool*	BR1 
Common Device Configuration	< None >
Call Classification*	Use System Default
NetworkLocale	< None >
Packet Capture Mode*	None
Packet Capture Duration	0
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Load Information	
Use Trusted Relay Point*	Default
Route Class Signaling Enabled*	Off
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> V150 (subset)	
<input checked="" type="checkbox"/> PSTN Access	

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain	< None >
MLPP Indication	Default
MLPP Preemption	Default

Interface Information

PRI Protocol Type*	PRI NI2
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Protocol Side*	User
Channel Selection Order*	Top Down
Channel IE Type*	Use Number when 1B
PCM Type*	μ-law
Delay for first restart (1/8 sec ticks)*	32
Delay between restarts (1/8 sec ticks)*	4
<input checked="" type="checkbox"/> Inhibit restarts at PRI initialization	
<input type="checkbox"/> Enable status poll	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> Enable G.Clear	

Call Routing Information - Inbound Calls

Significant Digits*	4
Calling Search Space	BR1
AAR Calling Search Space	< None >
Prefix DN	

Call Routing Information - Outbound Calls

Calling Party Presentation*	Default
Calling Party Selection*	Originator
Called party IE number type unknown*	Cisco CallManager
Calling party IE number type unknown*	Cisco CallManager
Called Numbering Plan*	Cisco CallManager
Calling Numbering Plan*	Cisco CallManager
Number of digits to strip*	0
Caller ID DN	
SMDI Base Port*	0
Called Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	

PRI Protocol Type Specific Information

☒ Display IE Delivery

☒ Redirecting Number IE Delivery - Outbound

Redirecting Party Transformation CSS < None >

☒ Use Device Pool Redirecting Party Transformation CSS

☒ Redirecting Number IE Delivery - Inbound

☒ Send Extra Leading Character in Display IE***

☐ Setup non-ISDN Progress Indicator IE Enable****

☐ MCDN Channel Number Extension Bit Set to Zero**

☐ Send Calling Name In Facility IE

☐ Interface Identifier Present**

Interface Identifier Value** 0

Connected Line ID Presentation (QSIG Inbound Call)* Default

Reset

Reset

Close

Configure the BR1 router to download a base MGCP configuration from CUCM.

```
BR1(config)#card type t1 0 0
BR1(config)#ccm-manager config server 10.10.150.71
BR1(config)#ccm-manager config
BR1(config)#
```

This package type is not the default package type.

Reset the Gateway on CCM if this Gateway registered with Cisco Call Manager (CCM)

...OUTPUT OMITTED...

Router BR1 has now been configured as an MGCP gateway. However, we need to break the linkage between CUCM9-PUB2 and BR1, such that resetting the MGCP gateway in CUCM will not cause the BR1 MGCP gateway to reset.

```
BR1(config)#no ccm-manager config
```

Customize the MGCP gateway configuration on router BR1.

Router BR1 should use only three B channels. Also, MGCP signaling and media should be bound to the VLAN 400 SVI.

```
BR1(config)#voice-port 0/0/0:23
```

```

BR1(config-voiceport)#shutdown
BR1(config-voiceport)#
...OUTPUT OMITTED...
BR1(config-voiceport)#int s 0/0/0:23
BR1(config-if)#no isdn bind-l3 ccm-manager
BR1(config)#controller t1 0/0/0
BR1(config-controller)#shutdown
BR1(config-controller)#
Jul 29 16:29:30.394: %CONTROLLER-5-UPDOWN: Controller T1 0/0/0,
changed state to administratively down
BR1(config-controller)#
Jul 29 16:29:32.394: %LINK-3-UPDOWN: Interface Serial0/0/0:23,
changed state to down
BR1(config-controller)#no pri-group
BR1(config-controller)#pri-group timeslots 1-3 service mgcp
BR1(config-controller)#
Jul 29 16:29:52.734: %LINEPROTO-5-UPDOWN: Line protocol on
Interface Serial0/0/0:0, changed state to down
Jul 29 16:29:52.734: %LINEPROTO-5-UPDOWN: Line protocol on
Interface Serial0/0/0:1, changed state to down
Jul 29 16:29:52.734: %LINEPROTO-5-UPDOWN: Line protocol on
Interface Serial0/0/0:2, changed state to down
Jul 29 16:29:52.734: %LINEPROTO-5-UPDOWN: Line protocol on
Interface Serial0/0/0:23, changed state to down
BR1(config-controller)#no shutdown
BR1(config-controller)#
Jul 29 16:30:03.142: %CONTROLLER-5-UPDOWN: Controller T1 0/0/0,
changed state to up
Jul 29 16:30:05.142: %LINK-3-UPDOWN: Interface Serial0/0/0:23,
changed state to up
BR1(config-controller)#interface serial 0/0/0:23
BR1(config-if)#isdn bind-l3 ccm-manager
BR1(config-if)#exit
BR1(config)#mgcp bind control source-interface vlan400
BR1(config)#mgcp bind media source-interface vlan400
BR1(config)#no mgcp
WARNING: no mgcp: Teardown MGCP application may take a while to
clean up resources
BR1(config)#
Jul 29 16:31:02.355: %MGCP_APP-6-MGCP_SHUTDOWN_COMPLETE: MGCP
Shutdown has completed

BR1(config)#mgcp
BR1(config)#end
BR1#
Jul 29 16:31:16.447: %SYS-5-CONFIG_I: Configured from console by
cisco on console

```

BR1#

Task 3.4 SIP Trunk

Create Trunk on CUCM9-PUB1

Device > Trunk > Add New

Trunk Type: **SIP Trunk**

Device Protocol: **SIP**

Next

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾

Trunk Configuration

Next

Status

Status: Ready

Trunk Information

Trunk Type* SIP Trunk

Device Protocol* SIP

Trunk Service Type* None(Default)

Device Name: **HQ-TO-CUBE**

Device Pool: **CUBE**

Calling Search Space: **CUBE**

Destination Address: **10.10.120.1** (NOTE: This is the IP address to which SIP is bound in the CUBE configuration.)

SIP Trunk Security Profile: **Non Secure SIP Trunk Profile**

SIP Profile: **Standard SIP Profile**

Save

Reset

Reset

Close

System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help

Trunk Configuration
Save
Delete
Reset
Add New

Product:
Device Protocol:
Trunk Service Type
Device Name*
Description
Device Pool*
Common Device Configuration
Call Classification*
Media Resource Group List
Location*
AAR Group
Tunneled Protocol*
QSIG Variant*
ASN.1 ROSE OID Encoding*
Packet Capture Mode*
Packet Capture Duration
☐ Media Termination Point Required
☒ Retry Video Call as Audio
☐ Path Replacement Support
☐ Transmit UTF-8 for Calling Party Name
☐ Transmit UTF-8 Names in QSIG APDU
☐ Unattended Port
☐ SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will cause the call to fail.
Consider Traffic on This Trunk Secure*
Route Class Signaling Enabled*
Use Trusted Relay Point*
☒ PSTN Access
☐ Run On All Active Unified CM Nodes

SIP Trunk
SIP
None(Default)
HQ-TO-CUBE
CUBE
< None >
Use System Default
< None >
CUBE
< None >
None
No Changes
No Changes
None
0
When using both sRTP and TLS
Default
Default

Intercompany Media Engine (IME)
E.164 Transformation Profile
< None >

Multilevel Precedence and Preemption (MLPP) Information
MLPP Domain
< None >

Call Routing Information
☒ Remote-Party-Id
☒ Asserted-Identity
Asserted-Type*
Default
SIP Privacy*
Default

Inbound Calls
Significant Digits*
All
Connected Line ID Presentation*
Default
Connected Name Presentation*
Default
Calling Search Space
CUBE
AAR Calling Search Space
< None >
Prefix DN
☐ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured in the DevicePool/Service Parameter will be used.
Clear Prefix Settings
Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
Incoming Number	Default	0	< None >

Connected Party Settings
Connected Party Transformation CSS
< None >
☒ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection*

Calling Line ID Presentation*

Calling Name Presentation*

Calling and Connected Party Info Format*

☐ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS

☒ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

☐ Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	10.10.120.1		5060

MTP Preferred Originating Codec*

BLF Presence Group*

SIP Trunk Security Profile*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile*

DTMF Signaling Method*

Create Route Group on CUCM9-PUB1

Call Routing > Route/Hunt > Route Group > Add New

Route Group Name: **SIP-TRUNK-TO-CUBE**




Available Devices: **HQ-TO-CUBE**


Add to Route Group


Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Group Configuration

 Save  Delete  Add New

Status
 Update successful



Route Group Information
 Route Group Name* 
 Distribution Algorithm*

Route Group Member Information

Find Devices to Add to Route Group
 Device Name contains
 Available Devices**



10.10.120.1

HQ-TO-CUBE


 Port(s)
 

Current Route Group Members
 Selected Devices (ordered by priority)*

HQ-TO-CUBE (All Ports)


 Removed Devices*** 

Create Route List on CUCM9-PUB1

Call Routing > Route/Hunt > Route List > Add New


Name: **HQ-TO-BR1**


Cisco Unified Communications Manager Group: **SUB-PUB**



Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Route List Configuration

 Save

Status
 Status: Ready

Route List Information
☒ Device is trusted
 Name* 
 Description
 Cisco Unified Communications Manager Group* 

Add Route Group

Route List Member Information

Selected Groups**

Add Route Group

Route Group: SIP-TRUNK-TO-CUBE

Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾

Route List Detail Configuration

Save

Status

Status: Ready

Route List Member Information

Route Group* SIP-TRUNK-TO-CUBE-[NON-QSIG]

Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Route List Configuration

Save Delete Copy Reset Apply Config Add New

Status

Status: Ready

Route List Information

Registration Registered with Cisco Unified Communications Manager 192.168.1.72

IP Address 192.168.1.72

☒ Device is trusted

Name* HQ-TO-BR1

Description

Cisco Unified Communications Manager Group* SUB-PUB

☒ Enable this Route List (change effective on Save; no reset required)

☐ Run On All Active Unified CM Nodes

Route List Member Information

Selected Groups** SIP-TRUNK-TO-CUBE

Add Route Group

Create Route Pattern on CUCM9-PUB1

Call Routing > Route/Hunt > Route Pattern > Add New

Route Pattern: **3XXX**
Route Partition: **INTERNAL**
Gateway/Route List: **HQ-TO-BR1**
Provide Outside Dial Tone: **(Uncheck)**
Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Route Pattern Configuration

Save Delete Copy Add New

Status

Status: Ready

Pattern Definition

Route Pattern*	3XXX
Route Partition	INTERNAL
Description	
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	HQ-TO-BR1 (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern
Call Classification*	OnNet
<input type="checkbox"/> Allow Device Override	<input type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level*	0
<input type="checkbox"/> Require Client Matter Code	

Repeat the previous procedure (i.e. adding a SIP Trunk, Route Group, Route List, and Route Pattern) on CUCM9-PUB2 (i.e. the BR1 CUCM server).

Task 3.5 Troubleshooting Video Call Between HQ and BR1

One router HQ (i.e. the router acting as the CUBE), we enter the **debug ccsip messages** command, make a call from HQ Phone 1 to BR1 Phone 1, and copy the **debug** output to a

Microsoft Notepad document. (**NOTE:** We should then turn off debugging with the **u all** command.) The output of the **debug** command is given below. Observe the following:

- The first set of highlights shows the SIP **INVITE** message sent from the CUBE (10.10.120.1) to the CUCM9-PUB2 (i.e. the BR1 CUCM) server (10.10.150.71).
- The SIP **INVITE** message does not contain Session Description Protocol (SDP) information (e.g. RTP payload type information for audio and video).
- Therefore, this call is using SIP Delayed Offer (as opposed to Early Offer, which contains SDP information in the INVITE message). This means that we should see SDP information in a SIP **200 OK** message sent from CUCM9-PUB2 (i.e. 10.10.150.71) when BR1 Phone 1 answers the call.
- If you look further down in the output, the SIP **200 OK** message received by the CUBE from CUCM9-PUB2 highlighted.
- The **m=video 31634 RTP/AVP 126 97** entry in the SIP **200 OK** message indicates that BR1 Phone 1 does indeed wish to set up a video call with HQ Phone 1.
- The H.264 video codec uses a **dynamic payload type**, meaning that the payload type value could be any number in the range 96 – 127.
- In this example CUCM9-PUB2 (i.e. the BR1 CUCM server) has offered to use a payload type of either **126** or **97**.
- However, notice the next **SIP 200 OK** message, sent from the CUBE to the CUCM9-SUB1 server. Its SDP information contains the entry **m=video 16426 RTP/AVP 119 119**. This means that the CUBE is trying to negotiate an RTP payload type of 119 for the video.
- Interestingly, the SIP **ACK** message, sent from the CUCM9-SUB1 server, in response to the previous SIP 200 OK message, contains the entry **m=video 31604 RTP/AVP 97**. This means that HQ Phone 1 offered to **receive** video using an RTP payload type 97, rather than the proposed RTP payload type of 119. The reason this happens is that Cisco 9971 IP Phones are only configured to **receive** RTP video using payload types of 97 and 126 when the H.264 codec is being used. Therefore, the HQ Phone 1 will not receive video using the proposed RTP payload type of 119 for video. However, HQ Phone 1 will **transmit** video using the proposed RTP payload type of 119.
- The **ACK** message received by the CUBE (from the CUCM9-SUB1 server) is then forwarded from the CUBE to the CUCM9-PUB2 server, containing the entry **m=video 16424 RTP/AVP 97**.
- The issue we now have is that **HQ Phone 1** is **transmitting** video using **RTP payload type 119** and **receiving** video using **RTP payload type 97**. However, **BR1 Phone 1** has no knowledge of RTP payload type 119 being used. It thinks that it's going to **transmit and receive** video using an **RTP payload type of 97**. However, since HQ Phone 1 is transmitting video using an RTP payload type of 119, BR1 Phone 1 will not receive the video stream.

HQ#debug ccsip messages

SIP Call messages tracing is enabled

HQ#

Aug 4 13:37:13.759: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:

Received:

INVITE sip:3001@10.10.120.1:5060 SIP/2.0
Via: SIP/2.0/TCP 192.168.1.72:5060;branch=z9hG4bK247f30e04e
From: "HQ Phone 1" <sip:2001@192.168.1.72>;tag=69~cdc2033a-490d-4759-83cf-da2aa2f1a2f6-33745340
To: <sip:3001@10.10.120.1>
Date: Tue, 04 Aug 2015 13:37:13 GMT
Call-ID: e8869000-5c01c009-a-4801a8c0@192.168.1.72
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM9.1
Allow: INVITE, OPTI
HQ#ONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback,X-cisco-original-called
Call-Info: <sip:192.168.1.72:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 3901132800-0000065536-0000000004-1208068288
Session-Expires: 1800
P-Asserted-Identity: "HQ Phone 1" <sip:2001@192.168.1.72>
Remote-Party-ID: "HQ Phone 1" <sip:2001@192.168.1.72>;party=calling;screen=yes;privac
HQ#y=off
Contact: <sip:2001@192.168.1.72:5060;transport=tcp>;video;audio
Max-Forwards: 69
Content-Length: 0

Aug 4 13:37:13.763: //21/E88690000000/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 100 Trying
Via: SIP/2.0/TCP 192.168.1.72:5060;branch=z9hG4bK247f30e04e
From: "HQ Phone 1" <sip:2001@192.168.1.72>;tag=69~cdc2033a-490d-4759-83cf-da2aa2f1a2f6-33745340
To: <sip:3001@10.10.120.1>
Date: Tue, 04 Aug 2015 13:37:13 GMT
Call-ID: e8869000-5c01c009-a-4801a8c0@192.168.1.72
CSeq: 101 INVITE

HQ#Allow-Events: kpml, telephone-event
Server: Cisco-SIPGateway/IOS-15.2.4.M3
Content-Length: 0

Aug 4 13:37:13.767: //22/E88690000000/SIP/Msg/ccsipDisplayMsg:

*** THE FOLLOWING IS A SIP INVITE MESSAGE SENT FROM THE CUBE TO THE CUCM9-PUB2 SERVER, CORRESPONDING TO THE CALL LEG BETWEEN THE CUBE AND THE CUCM9-PUB2 SERVER. ***

Sent:

INVITE sip:3001@10.10.150.71:5060 SIP/2.0

Via: SIP/2.0/UDP 10.10.120.1:5060;branch=z9hG4bK101C9

Remote-Party-ID: "HQ Phone 1" <sip:2001@10.10.120.1>;party=calling;screen=yes;privacy=off

From: "HQ Phone 1" <sip:2001@10.10.120.1>;tag=25C9D0-2660

To: <sip:3001@10.10.150.71>

Date: Tue, 04 Aug 2015 13:37:13 GMT

Call-ID: BFC9E

HQ#1E9-39E411E5-80299979-57357498@10.10.120.1

Supported: timer,resource-priority,replaces,sdp-anat

Min-SE: 1800

Cisco-Guid: 3901132800-0000065536-0000000004-1208068288

User-Agent: Cisco-SIPGateway/IOS-15.2.4.M3

Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER

CSeq: 101 INVITE

Timestamp: 1438695433

Contact: <sip:2001@10.10.120.1:5060>

Call-Info: <sip:10.10.120.1:5060>;method="NOTIFY;Event=telephone-event;Duration=2000"

Expires: 180

Allow-E

HQ#vents: kpml, telephone-event

Max-Forwards: 68

Session-Expires: 1800

Content-Length: 0

Aug 4 13:37:13.771: //22/E88690000000/SIP/Msg/ccsipDisplayMsg:

Received:

SIP/2.0 100 Trying

Via: SIP/2.0/UDP 10.10.120.1:5060;branch=z9hG4bK101C9

From: "HQ Phone 1" <sip:2001@10.10.120.1>;tag=25C9D0-2660

To: <sip:3001@10.10.150.71>

Date: Tue, 04 Aug 2015 13:37:13 GMT

Call-ID: BFC9E1E9-39E411E5-80299979-57357498@10.10.120.1

CSeq: 101 INVITE

Allow-Events: presence

Content-Length: 0

A

HQ#ug 4 13:37:13.831: //22/E88690000000/SIP/Msg/ccsipDisplayMsg:

Received:

SIP/2.0 180 Ringing

Via: SIP/2.0/UDP 10.10.120.1:5060;branch=z9hG4bK101C9

From: "HQ Phone 1" <sip:2001@10.10.120.1>;tag=25C9D0-2660

To: <sip:3001@10.10.150.71>;tag=73~3f45eec0-372d-429f-8b76-70ec7e16beb2-17085947

Date: Tue, 04 Aug 2015 13:37:13 GMT

Call-ID: BFC9E1E9-39E411E5-80299979-57357498@10.10.120.1

CSeq: 101 INVITE

Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY

Allow-Ev

HQ#uents: presence

Supported: X-cisco-srtp-fallback

Supported: Geolocation

P-Asserted-Identity: "BR1 Phone 1" <sip:3001@10.10.150.71>

Remote-Party-ID: "BR1 Phone 1"

<sip:3001@10.10.150.71>;party=called;screen=yes;privacy=off

Contact: <sip:3001@10.10.150.71:5060>;video

Content-Length: 0

Aug 4 13:37:13.831: //21/E88690000000/SIP/Msg/ccsipDisplayMsg:

Sent:

SIP/2.0 180 Ringing

Via: SIP/2.0/TCP 192.168.1.72:5060;branch=z9hG4bK247f30e04e

From: "HQ Phone 1" <sip:2001@192.168.1.72>;tag=69~cd

HQ#c2033a-490d-4759-83cf-da2aa2f1a2f6-33745340

To: <sip:3001@10.10.120.1>;tag=25CA10-1CC8

Date: Tue, 04 Aug 2015 13:37:13 GMT

Call-ID: e8869000-5c01c009-a-4801a8c0@192.168.1.72

CSeq: 101 INVITE

Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER

Allow-Events: kpml, telephone-event

Remote-Party-ID: "BR1 Phone 1" <sip:3001@10.10.120.1>;party=called;screen=yes;privacy=off

Contact: <sip:3001@10.10.120.1:5060;transport=tcp>

Server: Cisco-SIPGateway/IO

HQ#S-15.2.4.M3

Content-Length: 0

Aug 4 13:37:15.771: //22/E88690000000/SIP/Msg/ccsipDisplayMsg:

*** THE FOLLOWING IS A SIP 200 OK MESSAGE RECEIVED BY THE CUBE FROM THE CUCM9-PUB2 SERVER, CORRESPONDING TO THE CALL LEG BETWEEN THE CUBE AND THE CUCM9-PUB2 SERVER. ***

Received:

SIP/2.0 200 OK

Via: SIP/2.0/UDP 10.10.120.1:5060;branch=z9hG4bK101C9

From: "HQ Phone 1" <sip:2001@10.10.120.1>;tag=25C9D0-2660

To: <sip:3001@10.10.150.71>;tag=73~3f45eec0-372d-429f-8b76-70ec7e16beb2-17085947

Date: Tue, 04 Aug 2015 13:37:13 GMT

Call-ID: BFC9E1E9-39E411E5-80299979-57357498@10.10.120.1

CSeq: 101 INVITE

Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, HQ#REFER, SUBSCRIBE, NOTIFY

Allow-Events: presence, kpml

Supported: replaces

Supported: X-cisco-srtp-fallback

Supported: Geolocation

Session-Expires: 1800;refresher=uas

Require: timer

P-Asserted-Identity: "BR1 Phone 1" <sip:3001@10.10.150.71>

Remote-Party-ID: "BR1 Phone 1"

<sip:3001@10.10.150.71>;party=called;screen=yes;privacy=off

Contact: <sip:3001@10.10.150.71:5060>;video

Content-Type: application/sdp

Content-Length: 649

v=0

o=CiscoSystemsCCM-SIP 73 1 IN IP4 10.10.150.71

s=

HQ#SIP Call

c=IN IP4 10.10.140.10

b=TIAS:384000

b=AS:384

t=0 0

m=audio 23782 RTP/AVP 116 18 101

a=rtpmap:116 iLBC/8000

a=ptime:20

a=maxptime:20

a=fmtp:116 mode=20

a=rtpmap:18 G729/8000

a=ptime:20

a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 31634 RTP/AVP 126 97
b=TIAS:368000
a=rtpmap:126 H264/90000
a=fmtp:126 profile-level-id=42801E;packetization-mode=1;level-asymmetry-allowed=1
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42801E;pa
HQ#cketization-mode=0;level-asymmetry-allowed=1
a=imageattr:* recv [x=640,y=480,q=0.50]
a=content:main
imageattr parse payload numtok not foundimageattr payload found, specific is 255
Aug 4 13:37:15.779: //21/E88690000000/SIP/Msg/ccsipDisplayMsg:

*** THE FOLLOWING IS A SIP 200 OK MESSAGE SENT FROM THE CUBE TO THE CUCM9-SUB1 SERVER, CORRESPONDING TO THE CALL LEG BETWEEN THE CUCM9-SUB1 SERVER AND THE CUBE. ***

Sent:

SIP/2.0 200 OK

Via: SIP/2.0/TCP 192.168.1.72:5060;branch=z9hG4bK247f30e04e

From: "HQ Phone 1" <sip:2001@192.168.1.72>;tag=69~cdc2033a-490d-4759-83cf-da2aa2f1a2f6-33745340

To: <sip:3001@10.10.120.1>;tag=25CA10-1CC8

Date: Tue, 04 Aug 2015 13:37

HQ#:13 GMT

Call-ID: e8869000-5c01c009-a-4801a8c0@192.168.1.72

CSeq: 101 INVITE

Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER

Allow-Events: kpml, telephone-event

Remote-Party-ID: "BR1 Phone 1" <sip:3001@10.10.120.1>;party=called;screen=yes;privacy=off

Contact: <sip:3001@10.10.120.1:5060;transport=tcp>

Supported: replaces

Call-Info: <sip:10.10.120.1:5060>;method="NOTIFY;Event=telephone-event;Duration=500"

Supported: sdp-anat

Server: Cisco-S

HQ#IPGateway/IOS-15.2.4.M3

Session-Expires: 1800;refresher=uas

Require: timer

Supported: timer

Content-Type: application/sdp

Content-Disposition: session;handling=required

Content-Length: 527

v=0
o=CiscoSystemsSIP-GW-UserAgent 811 8425 IN IP4 10.10.120.1
s=SIP Call
c=IN IP4 10.10.120.1
t=0 0
m=audio 16420 RTP/AVP 116 18 101
c=IN IP4 10.10.120.1
a=rtpmap:116 iLBC/8000
a=fmtp:116
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-1
HQ#5
m=video 16426 RTP/AVP 119 119
c=IN IP4 10.10.120.1
b=TIAS:368000
a=rtpmap:119 H264/90000
a=fmtp:119 profile-level-id=42801E;packetization-mode=1
a=rtpmap:119 H264/90000
a=fmtp:119 profile-level-id=42801E;packetization-mode=0

Aug 4 13:37:15.791: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:

***** THE FOLLOWING IS A SIP ACK MESSAGE SENT FROM THE CUCM9-SUB1 SERVER TO THE CUBE, IN RESPONSE TO THE PREVIOUS SIP 200 OK MESSAGE. *****

Received:

ACK sip:3001@10.10.120.1:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 192.168.1.72:5060;branch=z9hG4bK256409f4ee
From: "HQ Phone 1" <sip:2001@192.168.1.72>;tag=69~cdc2033a-490d-4759-83cf
HQ#-da2aa2f1a2f6-33745340
To: <sip:3001@10.10.120.1>;tag=25CA10-1CC8
Date: Tue, 04 Aug 2015 13:37:13 GMT
Call-ID: e8869000-5c01c009-a-4801a8c0@192.168.1.72
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 437

v=0

o=CiscoSystemsCCM-SIP 69 1 IN IP4 192.168.1.72
s=SIP Call
c=IN IP4 10.10.120.20
b=TIAS:384000
b=AS:384
t=0 0
m=audio 31490 RTP/AVP 116 101
a=rtpmap:116 iLBC/8000
a=ptime:30
a=maxptime:30
a=fmtp:116 mod
HQ#e=30
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 31604 RTP/AVP 97
b=TIAS:368000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42801E;packetization-mode=0;level-asymmetry-allowed=1
a=content:main

Aug 4 13:37:15.799: //22/E88690000000/SIP/Msg/ccsipDisplayMsg:

***** THE FOLLOWING IS THE PREVIOUS SIP ACK MESSAGE (RECEIVED FROM THE CUCM9-SUB1 SERVER) BEING FORWARDED BY THE CUBE TO THE CUCM9-PUB2 SERVER. *****

Sent:

ACK sip:3001@10.10.150.71:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.120.1:5060;branch=z9hG4bK11214D
From: "HQ Phone 1" <sip:2001@10.10.120.1>;tag=25C9D0-2660
To: <sip:3001@10.10.150.71>;tag=73~3f45eec0-372d-
HQ#429f-8b76-70ec7e16beb2-17085947
Date: Tue, 04 Aug 2015 13:37:13 GMT
Call-ID: BFC9E1E9-39E411E5-80299979-57357498@10.10.120.1
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: kpml, telephone-event
Content-Type: application/sdp
Content-Length: 426

v=0
o=CiscoSystemsSIP-GW-UserAgent 9442 4939 IN IP4 10.10.120.1
s=SIP Call
c=IN IP4 10.10.120.1
t=0 0

m=audio 16422 RTP/AVP 116 101
c=IN IP4 10.10.120.1
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=30
a=rtpmap:101 telephone-event/8000
a
HQ#=fmp:101 0-15
a=ptime:30
a=maxptime:30
m=video 16424 RTP/AVP 97
c=IN IP4 10.10.120.1
b=TIAS:368000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42801E;packetization-mode=0

HQ#u all
All possible debugging has been turned off
HQ#

The contents of the Microsoft Notepad document that you submit for this task could look like the following:

```
HQ#debug ccsip messages
SIP Call messages tracing is enabled
HQ#
Aug  4 13:37:13.759: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
INVITE sip:3001@10.10.120.1:5060 SIP/2.0
Via: SIP/2.0/TCP 192.168.1.72:5060;branch=z9hG4bK247f30e04e
From: "HQ Phone 1" <sip:2001@192.168.1.72>;tag=69~cdc2033a-490d-4759-
83cf-da2aa2f1a2f6-33745340
To: <sip:3001@10.10.120.1>
Date: Tue, 04 Aug 2015 13:37:13 GMT
Call-ID: e8869000-5c01c009-a-4801a8c0@192.168.1.72
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM9.1
Allow: INVITE, OPTI
HQ#ONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE,
NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback,X-cisco-original-called
Call-Info: <sip:192.168.1.72:5060>;method="NOTIFY;Event=telephone-
event;Duration=500"
Cisco-Guid: 3901132800-0000065536-0000000004-1208068288
Session-Expires: 1800
```

P-Asserted-Identity: "HQ Phone 1" <sip:2001@192.168.1.72>
Remote-Party-ID: "HQ Phone 1"
<sip:2001@192.168.1.72>;party=calling;screen=yes;privacy=off
Contact: <sip:2001@192.168.1.72:5060;transport=tcp>;video;audio
Max-Forwards: 69
Content-Length: 0

Aug 4 13:37:13.763: //21/E88690000000/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 100 Trying
Via: SIP/2.0/TCP 192.168.1.72:5060;branch=z9hG4bK247f30e04e
From: "HQ Phone 1" <sip:2001@192.168.1.72>;tag=69~cdc2033a-490d-4759-83cf-da2aa2f1a2f6-33745340
To: <sip:3001@10.10.120.1>
Date: Tue, 04 Aug 2015 13:37:13 GMT
Call-ID: e8869000-5c01c009-a-4801a8c0@192.168.1.72
CSeq: 101 INVITE

HQ#Allow-Events: kpml, telephone-event
Server: Cisco-SIPGateway/IOS-15.2.4.M3
Content-Length: 0

Aug 4 13:37:13.767: //22/E88690000000/SIP/Msg/ccsipDisplayMsg:
Sent:
INVITE sip:3001@10.10.150.71:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.120.1:5060;branch=z9hG4bK101C9
Remote-Party-ID: "HQ Phone 1"
<sip:2001@10.10.120.1>;party=calling;screen=yes;privacy=off
From: "HQ Phone 1" <sip:2001@10.10.120.1>;tag=25C9D0-2660
To: <sip:3001@10.10.150.71>
Date: Tue, 04 Aug 2015 13:37:13 GMT
Call-ID: BFC9E
HQ#1E9-39E411E5-80299979-57357498@10.10.120.1
Supported: timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 3901132800-0000065536-0000000004-1208068288
User-Agent: Cisco-SIPGateway/IOS-15.2.4.M3
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1438695433
Contact: <sip:2001@10.10.120.1:5060>
Call-Info: <sip:10.10.120.1:5060>;method="NOTIFY;Event=telephone-event;Duration=2000"
Expires: 180
Allow-E
HQ#vents: kpml, telephone-event
Max-Forwards: 68
Session-Expires: 1800

Content-Length: 0

Aug 4 13:37:13.771: //22/E88690000000/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.10.120.1:5060;branch=z9hG4bK101C9
From: "HQ Phone 1" <sip:2001@10.10.120.1>;tag=25C9D0-2660
To: <sip:3001@10.10.150.71>
Date: Tue, 04 Aug 2015 13:37:13 GMT
Call-ID: BFC9E1E9-39E411E5-80299979-57357498@10.10.120.1
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0

A
HQ#ug 4 13:37:13.831: //22/E88690000000/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.10.120.1:5060;branch=z9hG4bK101C9
From: "HQ Phone 1" <sip:2001@10.10.120.1>;tag=25C9D0-2660
To: <sip:3001@10.10.150.71>;tag=73~3f45eec0-372d-429f-8b76-70ec7e16beb2-17085947
Date: Tue, 04 Aug 2015 13:37:13 GMT
Call-ID: BFC9E1E9-39E411E5-80299979-57357498@10.10.120.1
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Ev
HQ#uents: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: "BR1 Phone 1" <sip:3001@10.10.150.71>
Remote-Party-ID: "BR1 Phone 1" <sip:3001@10.10.150.71>;party=called;screen=yes;privacy=off
Contact: <sip:3001@10.10.150.71:5060>;video
Content-Length: 0

Aug 4 13:37:13.831: //21/E88690000000/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 192.168.1.72:5060;branch=z9hG4bK247f30e04e
From: "HQ Phone 1" <sip:2001@192.168.1.72>;tag=69~cd
HQ#u all
All possible debugging has been turned off
HQ#c2033a-490d-4759-83cf-da2aa2f1a2f6-33745340
To: <sip:3001@10.10.120.1>;tag=25CA10-1CC8
Date: Tue, 04 Aug 2015 13:37:13 GMT
Call-ID: e8869000-5c01c009-a-4801a8c0@192.168.1.72
CSeq: 101 INVITE

Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER,
SUBSCRIBE, NOTIFY, INFO, REGISTER
Allow-Events: kpml, telephone-event
Remote-Party-ID: "BR1 Phone 1"
<sip:3001@10.10.120.1>;party=called;screen=yes;privacy=off
Contact: <sip:3001@10.10.120.1:5060;transport=tcp>
Server: Cisco-SIPGateway/IO
Hq#S-15.2.4.M3
Content-Length: 0

Aug 4 13:37:15.771: //22/E88690000000/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.10.120.1:5060;branch=z9hG4bK101C9
From: "HQ Phone 1" <sip:2001@10.10.120.1>;tag=25C9D0-2660
To: <sip:3001@10.10.150.71>;tag=73~3f45eec0-372d-429f-8b76-
70ec7e16beb2-17085947
Date: Tue, 04 Aug 2015 13:37:13 GMT
Call-ID: BFC9E1E9-39E411E5-80299979-57357498@10.10.120.1
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE,
Hq#REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence, kpml
Supported: replaces
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Session-Expires: 1800;refresher=uas
Require: timer
P-Asserted-Identity: "BR1 Phone 1" <sip:3001@10.10.150.71>
Remote-Party-ID: "BR1 Phone 1"
<sip:3001@10.10.150.71>;party=called;screen=yes;privacy=off
Contact: <sip:3001@10.10.150.71:5060>;video
Content-Type: application/sdp
Content-Length: 649

v=0
o=CiscoSystemsCCM-SIP 73 1 IN IP4 10.10.150.71
s=
Hq#SIP Call
c=IN IP4 10.10.140.10
b=TIAS:384000
b=AS:384
t=0 0
m=audio 23782 RTP/AVP 116 18 101
a=rtpmap:116 iLBC/8000
a=ptime:20
a=maxptime:20
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=ptime:20
a=fmtp:18 annexb=no

a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

BR1 Phone 1 offers to the CUBE to use an RTP payload type of 126 or 97

m=video 31634 RTP/AVP 126 97
b=TIAS:368000
a=rtpmap:126 H264/90000
a=fmtp:126 profile-level-id=42801E;packetization-mode=1;level-
asymmetry-allowed=1
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42801E;pa
HQ#cketization-mode=0;level-asymmetry-allowed=1
a=imageattr:* recv [x=640,y=480,q=0.50]
a=content:main
imageattr parse payload numtok not foundimageattr payload found,
specific is 255
Aug 4 13:37:15.779: //21/E88690000000/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 200 OK
Via: SIP/2.0/TCP 192.168.1.72:5060;branch=z9hG4bK247f30e04e
From: "HQ Phone 1" <sip:2001@192.168.1.72>;tag=69~cdc2033a-490d-4759-
83cf-da2aa2f1a2f6-33745340
To: <sip:3001@10.10.120.1>;tag=25CA10-1CC8
Date: Tue, 04 Aug 2015 13:37
HQ#:13 GMT
Call-ID: e8869000-5c01c009-a-4801a8c0@192.168.1.72
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER,
SUBSCRIBE, NOTIFY, INFO, REGISTER
Allow-Events: kpml, telephone-event
Remote-Party-ID: "BR1 Phone 1"
<sip:3001@10.10.120.1>;party=called;screen=yes;privacy=off
Contact: <sip:3001@10.10.120.1:5060;transport=tcp>
Supported: replaces
Call-Info: <sip:10.10.120.1:5060>;method="NOTIFY;Event=telephone-
event;Duration=500"
Supported: sdp-anat
Server: Cisco-S
HQ#IPGateway/IOS-15.2.4.M3
Session-Expires: 1800;refresher=uas
Require: timer
Supported: timer
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 527

v=0
o=CiscoSystemsSIP-GW-UserAgent 811 8425 IN IP4 10.10.120.1
s=SIP Call

c=IN IP4 10.10.120.1
t=0 0
m=audio 16420 RTP/AVP 116 18 101
c=IN IP4 10.10.120.1
a=rtpmap:116 iLBC/8000
a=fmtp:116
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-1
HQ#5

CUBE offers to HQ to use an RTP payload type of 119

m=video 16426 RTP/AVP 119 119
c=IN IP4 10.10.120.1
b=TIAS:368000
a=rtpmap:119 H264/90000
a=fmtp:119 profile-level-id=42801E;packetization-mode=1
a=rtpmap:119 H264/90000
a=fmtp:119 profile-level-id=42801E;packetization-mode=0

Aug 4 13:37:15.791: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
ACK sip:3001@10.10.120.1:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 192.168.1.72:5060;branch=z9hG4bK256409f4ee
From: "HQ Phone 1" <sip:2001@192.168.1.72>;tag=69~cdc2033a-490d-4759-83cf
HQ#-da2aa2f1a2f6-33745340
To: <sip:3001@10.10.120.1>;tag=25CA10-1CC8
Date: Tue, 04 Aug 2015 13:37:13 GMT
Call-ID: e8869000-5c01c009-a-4801a8c0@192.168.1.72
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 437

v=0
o=CiscoSystemsCCM-SIP 69 1 IN IP4 192.168.1.72
s=SIP Call
c=IN IP4 10.10.120.20
b=TIAS:384000
b=AS:384
t=0 0
m=audio 31490 RTP/AVP 116 101
a=rtpmap:116 iLBC/8000
a=ptime:30
a=maxptime:30
a=fmtp:116 mod

HQ#e=30
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

While HQ will agree to transmit using an RTP payload type of 119, a Cisco 9971 IP Phone cannot receive using that RTP payload type and instead says it will receive video using an RTP payload type of 97

m=video 31604 RTP/AVP 97
b=TIAS:368000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42801E;packetization-mode=0;level-
asymmetry-allowed=1
a=content:main

Aug 4 13:37:15.799: //22/E88690000000/SIP/Msg/ccsipDisplayMsg:
Sent:
ACK sip:3001@10.10.150.71:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.120.1:5060;branch=z9hG4bK11214D
From: "HQ Phone 1" <sip:2001@10.10.120.1>;tag=25C9D0-2660
To: <sip:3001@10.10.150.71>;tag=73~3f45eec0-372d-
HQ#429f-8b76-70ec7e16beb2-17085947
Date: Tue, 04 Aug 2015 13:37:13 GMT
Call-ID: BFC9E1E9-39E411E5-80299979-57357498@10.10.120.1
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: kpml, telephone-event
Content-Type: application/sdp
Content-Length: 426

v=0
o=CiscoSystemsSIP-GW-UserAgent 9442 4939 IN IP4 10.10.120.1
s=SIP Call
c=IN IP4 10.10.120.1
t=0 0
m=audio 16422 RTP/AVP 116 101
c=IN IP4 10.10.120.1
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=30
a=rtpmap:101 telephone-event/8000
a
HQ#=fmtp:101 0-15
a=ptime:30
a=maxptime:30

The CUBE forwards the ACK message to BR1. Since BR1 is unaware of the negotiation between the CUBE and HQ for HQ to transmit using an RTP payload type of 119, BR1 expects to receive video using an RTP payload type of 97, resulting in no video being displayed on BR1 Phone 1.

```
m=video 16424 RTP/AVP 97
c=IN IP4 10.10.120.1
b=TIAS:368000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42801E;packetization-mode=0
```

```
HQ#u all
All possible debugging has been turned off
HQ#
```

To fix the issue, we can tell the CUBE not get involved with the session negotiation (i.e. the SDP negotiation) between the endpoints.

```
HQ#conf term
Enter configuration commands, one per line. End with CNTL/Z.
HQ(config)#voice service voip
HQ(config-voi-serv)#sip
HQ(config-serv-sip)#pass-thru content sdp
HQ(config-serv-sip)#end
HQ#
```

Now, when you call between HQ Phone 1 and BR1 Phone 1, you should have two-way video.

Task 3.6 SIP Gateway

Create a SIP trunk on CUCM-PUB1

Device > Trunk > Add New

Trunk Type: **SIP Trunk**

Next

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾

Trunk Configuration

Next

Status

Status: Ready

Trunk Information

Trunk Type*

Device Protocol*

Trunk Service Type*

Next

Device Name: **HQ-TO-BR2**

Device Pool: **BR2**

Significant Digits: **4**

Calling Search Space: **BR2**

Destination Address: **10.10.32.3** (**NOTE:** This is the IP address of BR2's Lo 0 interface, to which SIP signaling is bound in BR2's configuration.)

SIP Trunk Security Profile: **Non Secure SIP Trunk Profile**

SIP Profile: **Standard SIP Profile**

Save

OK

Reset

Reset

Close

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Trunk Configuration

Save

Device Information

Product: SIP Trunk
Device Protocol: SIP
Trunk Service Type: None(Default)
Device Name*: HQ-TO-BR2
Description:
Device Pool*: BR2
Common Device Configuration: < None >
Call Classification*: Use System Default
Media Resource Group List: < None >
Location*: Hub_None
AAR Group: < None >
Tunneled Protocol*: None
QSIG Variant*: No Changes
ASN.1 ROSE OID Encoding*: No Changes
Packet Capture Mode*: None
Packet Capture Duration: 0
☐ Media Termination Point Required
☒ Retry Video Call as Audio
☐ Path Replacement Support
☐ Transmit UTF-8 for Calling Party Name
☐ Transmit UTF-8 Names in QSIG APDU
☐ Unattended Port
☐ SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.
Consider Traffic on This Trunk Secure*: When using both sRTP and TLS
Route Class Signalling Enabled*: Default
Use Trusted Relay Point*: Default
☒ PSTN Access
☐ Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain < None >

Call Routing Information

☒ Remote-Party-Id
☒ Asserted-Identity
 Asserted-Type* Default
 SIP Privacy* Default

Inbound Calls

Significant Digits* 4
 Connected Line ID Presentation* Default
 Connected Name Presentation* Default
 Calling Search Space BR2
 AAR Calling Search Space < None >
 Prefix DN
☐ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix assigned.

[Clear Prefix Settings](#) [Default Prefix Settings](#)

Number Type	Prefix	Strip Digits	Calling Search Space
Incoming Number	Default		< None >

Connected Party Settings

Connected Party Transformation CSS < None >
☒ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS < None >
☒ Use Device Pool Called Party Transformation CSS
 Calling Party Transformation CSS < None >
☒ Use Device Pool Calling Party Transformation CSS
 Calling Party Selection* Originator
 Calling Line ID Presentation* Default
 Calling Name Presentation* Default
 Calling and Connected Party Info Format* Deliver DN only in connected party
☐ Redirecting Diversion Header Delivery - Outbound
 Redirecting Party Transformation CSS < None >
☒ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN
 Caller Name
☐ Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Information

Destination

☐ Destination Address is an SRV

Destination Address	Destination Address IPv6	Destination Port
1* 10.10.32.3		5060

MTP Preferred Originating Codec* 711ulaw
 BLF Presence Group* Standard Presence group
 SIP Trunk Security Profile* Non Secure SIP Trunk Profile
 Rerouting Calling Search Space < None >
 Out-Of-Dialog Refer Calling Search Space < None >
 SUBSCRIBE Calling Search Space < None >
 SIP Profile* Standard SIP Profile
 DTMF Signaling Method* No Preference

Assign the SIP trunk to a new Route Group

Call Routing > Route/Hunt > Route Group > Add New

Route Group Name: **HQ-TO-BR2**


Available Devices: **HQ-TO-BR2**

Add to Route Group


Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

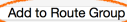
Route Group Configuration

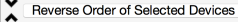
 Save


Status
i Status: Ready

Route Group Information
 Route Group Name* 
 Distribution Algorithm*

Route Group Member Information

Find Devices to Add to Route Group
 Device Name contains Find
 Available Devices**
 10.10.120.1
 10.10.160.1
 HQ-TO-BR2
 HQ-TO-CUBE
 Port(s)


Current Route Group Members
 Selected Devices (ordered by priority)* HQ-TO-BR2 (All Ports) 
 Removed Devices***

 Save

Assign the Route Group to a new Route List

Call Routing > Route/Hunt > Route List > Add New


Name: **SIP-TRUNK-TO-BR2**

Cisco Unified Communications Manager Group: **SUB-PUB**



Save


System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Route List Configuration

 Save

Status
i Status: Ready

Route List Information
☒ Device is trusted
 Name* 
 Description
 Cisco Unified Communications Manager Group* 

 Save

Add Route Group

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Route List Configuration

Save
 Delete
 Copy
 Reset
 Apply Config
 Add New

Status

Add successful

Route List Information

Registration: Unknown
 IP Address: Unknown
☒ Device is trusted
 Name*: SIP-TRUNK-TO-BR2
 Description:
 Cisco Unified Communications Manager Group*: SUB-PUB ▾
☒ Enable this Route List (change effective on Save; no reset required)
☐ Run On All Active Unified CM Nodes

Route List Member Information

Selected Groups**
 Removed Groups***

Add Route Group

Save
 Delete
 Copy
 Reset
 Apply Config
 Add New

Route Group: HQ-TO-BR2

Save
OK

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Route List Detail Configuration

Save

Status

Status: Ready

Route List Member Information

Route Group*: HQ-TO-BR2-[NON-QSIG] ▾

Calling Party Transformations

Use Calling Party's External Phone Number Mask*: Default ▾
 Calling Party Transform Mask:
 Prefix Digits (Outgoing Calls):
 Calling Party Number Type*: Cisco CallManager ▾
 Calling Party Numbering Plan*: Cisco CallManager ▾

Called Party Transformations

Discard Digits: < None > ▾
 Called Party Transform Mask:
 Prefix Digits (Outgoing Calls):
 Called Party Number Type*: Cisco CallManager ▾
 Called Party Numbering Plan*: Cisco CallManager ▾

Save

Save
Reset
Reset

Close

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Route List Configuration

Save Delete Copy Reset Apply Config Add New

Status
Add successful

Route List Information
Registration: Unknown
IP Address: Unknown
☒ Device is trusted
Name*: SIP-TRUNK-TO-BR2
Description:
Cisco Unified Communications Manager Group*: SUB-PUB
☒ Enable this Route List (change effective on Save; no reset required)
☐ Run On All Active Unified CM Nodes

Route List Member Information
Selected Groups**: HQ-TO-BR2
Removed Groups***

Route List Details
 HQ-TO-BR2

Save Delete Copy Reset Apply Config Add New

Create a Route Pattern (that references the new Route List) pointing to BR2 IP phones

Call Routing > Route/Hunt > Route Pattern > Add New

Route Pattern: **4XXX**

Route Partition: **INTERNAL**

Gateway/Route List: **SIP-TRUNK-TO-BR2**

Provide Outside Dial Tone: **(Unchecked)**

Save

OK

OK

Repeat the above procedure (i.e. creating a SIP Trunk, Route Group, Route List, and Route Pattern) on the BR2 CUCM9-PUB2 server.

Configure COR for Incoming/Outgoing Call Legs on BR2

At this point, in order to support both a SIP and an SCCP IP phone at BR2 and to support bidirectional video between BR2 Phone 1 and HQ Phone 1 or BR1 Phone 1, we need to configure one set of dial peers used by the SIP IP phone to reach HQ and BR1 and another set of dial peers used by the SCCP IP phone to reach HQ and BR1. Since there is quite a bit of configuration involved, this is a good opportunity to use Microsoft Notepad. Specifically, we can type out our configuration in Microsoft Notepad and then copy/paste the configuration into BR2. (**NOTE:** If the **pass-thru content sdp** command has previously been issued on BR2, under the Voice Services VoIP > SIP subconfiguration, then it should be removed.) Following is the text that we could enter into our Microsoft Notepad document.

```
dial-peer cor custom
name SIP
name SCCP

dial-peer cor list SIP
member SIP

dial-peer cor list SCCP
member SCCP

dial-peer voice 5 voip
corlist outgoing SCCP

dial-peer voice 6 voip
corlist outgoing SCCP
```



```
dial-peer voice 7 voip
corlist outgoing SCCP
```

```
ephone-dn 1
corlist outgoing SCCP
```

```
voice register pool 1
cor outgoing SIP default
```

```
dial-peer voice 8 voip
destination-pattern 2...$
session protocol sipv2
session target ipv4:192.168.1.72
codec transparent
rtp payload-type cisco-codec-fax-ack 98
rtp payload-type cisco-codec-video-h264 97
corlist outgoing SIP
dtmf-relay sip-kpml
no vad
```

```
dial-peer voice 9 voip
preference 1
destination-pattern 2...$
session protocol sipv2
session target ipv4:192.168.1.71
codec transparent
rtp payload-type cisco-codec-fax-ack 98
rtp payload-type cisco-codec-video-h264 97
corlist outgoing SIP
dtmf-relay sip-kpml
no vad
```

```
dial-peer voice 10 voip
destination-pattern 3...$
session protocol sipv2
session target ipv4:10.10.150.71
codec transparent
rtp payload-type cisco-codec-fax-ack 98
rtp payload-type cisco-codec-video-h264 97
corlist outgoing SIP
dtmf-relay sip-kpml
no vad
```

```
dial-peer voice 11 voip
incoming called-number 4001
session protocol sipv2
codec transparent
rtp payload-type cisco-codec-fax-ack 98
rtp payload-type cisco-codec-video-h264 97
corlist incoming SIP
dtmf-relay sip-kpml
```

```
no vad
```

```
dial-peer voice 12 voip  
incoming called-number 4002  
session protocol sipv2  
voice-class codec 1  
corlist outgoing SCCP  
dtmf-relay sip-kpml  
no vad
```

Module 11: HQ and BR1 Call Routing

Task 4.1 HQ Call Routing

Create Route Groups

Now that the HQ gateway has been configured, it can be added to a Route Group, which can then be assigned as a Local Route Group to the HQ Device Pool.

Call Routing > Route/Hunt > Route Group

Add New

Route Group Name: **HQ**


Select **10.10.120.1**

Add to Route Group


Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Group Configuration

 Save

Status

 Status: Ready

Route Group Information

Route Group Name*

Distribution Algorithm*

Route Group Member Information

Find Devices to Add to Route Group

Device Name contains

Available Devices**

10.10.120.1

Port(s)

Current Route Group Members

Selected Devices (ordered by priority)*

▼ Reverse Order of Selected Devices ▲

Removed Devices***

▼ ▲

Assign Local Route Groups

System > Device Pool > Find > HQ

Local Route Group: **HQ**

Save







Reset

Reset


Close

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾

Device Pool Configuration

 Save
  Delete
  Copy
  Reset
  Apply Config
  Add New


Status

 Status: Ready

Device Pool Information

Device Pool: HQ (11 members**)

Device Pool Settings

Device Pool Name* 

Cisco Unified Communications Manager Group*

Calling Search Space for Auto-registration

Adjunct CSS

Reverted Call Focus Priority

Intercompany Media Services Enrolled Group

Roaming Sensitive Settings

Date/Time Group*

Region*

Media Resource Group List

Location

Network Locale

SRST Reference*

Connection Monitor Duration***

Single Button Barge*

Join Across Lines*

Physical Location

Device Mobility Group

Local Route Group

Create a Route List that contains the Standard Local Route Group

Call Routing > Route/Hunt > Route List

Add New

Name: **SLRG**


Cisco Unified Communications Manager Group: **SUB-PUB**

Save

Route List Configuration



Status

 Status: Ready

Route List Information

☒ Device is trusted

Name *


SLRG


Description


Cisco Unified Communications Manager Group *

SUB-PUB

Save

 *- indicates required item.

 **Ordered by highest priority

 ***Will be removed from Route List when you click Save

Add Route Group

Add Route Group


Route Group: **Standard Local Route Group**


Save

OK

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route List Detail Configuration


 Save

Status
 Status: Ready

Route List Member Information
 Route Group* Standard Local Route Group ▾

Calling Party Transformations
 Use Calling Party's External Phone Number Mask* Default ▾
 Calling Party Transform Mask
 Prefix Digits (Outgoing Calls)
 Calling Party Number Type* Cisco CallManager ▾
 Calling Party Numbering Plan* Cisco CallManager ▾

Called Party Transformations
 Discard Digits < None > ▾
 Called Party Transform Mask
 Prefix Digits (Outgoing Calls)
 Called Party Number Type* Cisco CallManager ▾
 Called Party Numbering Plan* Cisco CallManager ▾

 Save

Save

OK

Create a generic Route Pattern to be used by multiple Translation Patterns

Call Routing > Route/Hunt > Route Pattern

Add New

Route Pattern: \+!

Route Partition: **PSTN**

Gateway/Route List: **SLRG**

Urgent Priority: **Checked**

Save

OK

OK

System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help

Route Pattern Configuration

Save

Status

i
Status: Ready

Pattern Definition

Route Pattern*

\+!

Route Partition

PSTN

Description

Numbering Plan

-- Not Selected --

Route Filter

< None >

MLPP Precedence*

Default

☐
Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

< None >

Route Class*

Default

Gateway/Route List*

SLRG

(Edit)

Route Option

☒
Route this pattern

☐
Block this pattern

No Error

Call Classification*

OffNet

☐
Allow Device Override
☒
Provide Outside Dial Tone
☐
Allow Overlap Sending
☒
Urgent Priority

☐
Require Forced Authorization Code

Authorization Level*

0

☐
Require Client Matter Code

Document a Route Plan for HQ

Similar to how we documented a BR2 route plan earlier (based on task requirements), we next need to document a route plan for HQ.

HQ Route Plan

Pattern	Route List	Route Group	ANI/TON	DNIS/TON	DM
911	SLRG	LRG	10/N	911/U	None
9.[2-9]XXXXXX	SLRG	LRG	7/S	7/S	PD
9.1[2-9]XX[2-9]XXXXXX	SLRG	LRG	10/N	11/N	PD
9.011!	SLRG	LRG	E.164/I	Var./I	PD
9.011!#	SLRG	LRG	E.164/I	Var./I	PD, Trailing #

Create Translation Patterns conforming to the previously documented HQ Route Plan

Using the HQ route plan, we start creating a series of Translation Patterns, which will match dialed digits, perform required digit manipulation, and extend calls to the \+! Route Pattern.

HQ Emergency

Call Routing > Translation Pattern > Add New

Translation Pattern: **911**

Partition: **HQ**

Calling Search Space: **PSTN**

Use Calling Party's External Phone Number Mask: **Checked**

Calling Party Transform Mask: **XXXXXXXXXX**

Calling Party Number Type: **National**

Calling Party Numbering Plan: **ISDN**

Prefix Digits (Outgoing Calls): **+**

Called Party Number Type: **Unknown**

Called Party Numbering Plan: **ISDN**

Save

The screenshot displays the 'Translation Pattern Configuration' page. At the top, there is a navigation bar with links: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. Below the navigation bar, the page title is 'Translation Pattern Configuration' and there are 'Related Links: Back To Find/List' and 'Go'. A 'Save' button is visible on the left. The main content area is divided into two sections: 'Pattern Definition' and 'Calling Party Transformations'.

Pattern Definition

Translation Pattern	911
Partition	HQ
Description	
Numbering Plan	< None >
Route Filter	< None >
MLPP Precedence *	Default
Resource Priority Namespace Network Domain	< None >
Calling Search Space	PSTN
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error
<input checked="" type="checkbox"/> Provide Outside Dial Tone	
<input checked="" type="checkbox"/> Urgent Priority	

Calling Party Transformations

<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	XXXXXXXXXX
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation *	Default
Calling Name Presentation *	Default
Calling Party Number Type *	National
Calling Party Numbering Plan *	ISDN

Called Party Transformations	
Discard Digits	< None >
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	+
Called Party Number Type*	Unknown
Called Party Numbering Plan*	ISDN
<input type="button" value="Save"/>	

HQ Local

Copy

Translation Pattern: **9.[2-9]XXXXXX**

Partition: **HQ**

Calling Search Space: **PSTN**

Urgent Priority: **Unchecked**

Use Calling Party's External Phone Number Mask: **Checked**

Calling Party Transform Mask: **XXXXXXX**

Calling Party Number Type: **Subscriber**

Calling Party Numbering Plan: **ISDN**

Discard Digits: **PreDot**

Prefix Digits (Outgoing Calls): **+**

Called Party Number Type: **Subscriber**

Called Party Numbering Plan: **ISDN**

Save

Translation Pattern Configuration		Related Links: Back To Find/List <input type="button" value="Go"/>
<input type="button" value="Save"/>		
Pattern Definition		
Translation Pattern	9.[2-9]XXXXXX	
Partition	HQ	
Description		
Numbering Plan	< None >	
Route Filter	< None >	
MLPP Precedence*	Default	
Resource Priority Namespace Network Domain	< None >	
Calling Search Space	PSTN	
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error	
<input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Urgent Priority		
Calling Party Transformations		
<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask		
Calling Party Transform Mask	XXXXXXX	
Prefix Digits (Outgoing Calls)		
Calling Line ID Presentation*	Default	
Calling Name Presentation*	Default	
Calling Party Number Type*	Subscriber	
Calling Party Numbering Plan*	ISDN	

Called Party Transformations	
Discard Digits	PreDot
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	+
Called Party Number Type*	Subscriber
Called Party Numbering Plan*	ISDN

Save

HQ Long Distance

Copy

Translation Pattern: **9.1[2-9]XX[2-9]XXXXXX**

Partition: **HQ**

Calling Search Space: **PSTN**

Urgent Priority: **Unchecked**

Use Calling Party's External Phone Number Mask: **Checked**

Calling Party Transform Mask: **XXXXXXXXXX**

Calling Party Number Type: **National**

Calling Party Numbering Plan: **ISDN**

Discard Digits: **PreDot**

Prefix Digits (Outgoing Calls): **+**

Called Party Number Type: **National**

Called Party Numbering Plan: **ISDN**

Save

Translation Pattern Configuration		Related Links: Back To Find/List Go
<p>Save</p>		
Pattern Definition		
Translation Pattern	9.1[2-9]XX[2-9]XXXXXX	
Partition	HQ	
Description		
Numbering Plan	< None >	
Route Filter	< None >	
MLPP Precedence*	Default	
Resource Priority Namespace Network Domain	< None >	
Calling Search Space	PSTN	
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error	
<input checked="" type="checkbox"/> Provide Outside Dial Tone		
<input type="checkbox"/> Urgent Priority		
Calling Party Transformations		
<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask		
Calling Party Transform Mask	XXXXXXXXXX	
Prefix Digits (Outgoing Calls)		
Calling Line ID Presentation*	Default	
Calling Name Presentation*	Default	
Calling Party Number Type*	National	
Calling Party Numbering Plan*	ISDN	

Called Party Transformations	
Discard Digits	PreDot
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	+
Called Party Number Type*	National
Called Party Numbering Plan*	ISDN

HQ International (without trailing #)

Copy

Translation Pattern: **9.011!**

Partition: **HQ**

Calling Search Space: **PSTN**

Urgent Priority: **Unchecked**

Use Calling Party's External Phone Number Mask: **Checked**

Calling Party Transform Mask: *blank*

Calling Party Number Type: **International**

Calling Party Numbering Plan: **ISDN**

Discard Digits: **PreDot**

Prefix Digits (Outgoing Calls): **+**

Called Party Number Type: **International**

Called Party Numbering Plan: **ISDN**

Save

Translation Pattern Configuration		Related Links: Back To Find/List Go
<div>Save</div>		
Pattern Definition		
Translation Pattern	9.011!	
Partition	HQ	
Description		
Numbering Plan	< None >	
Route Filter	< None >	
MLPP Precedence *	Default	
Resource Priority Namespace Network Domain	< None >	
Calling Search Space	PSTN	
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error	
<input checked="" type="checkbox"/> Provide Outside Dial Tone		
<input type="checkbox"/> Urgent Priority		
Calling Party Transformations		
<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask		
Calling Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Calling Line ID Presentation *	Default	
Calling Name Presentation *	Default	
Calling Party Number Type *	International	
Calling Party Numbering Plan *	ISDN	
Called Party Transformations		
Discard Digits	PreDot	
Called Party Transform Mask		
Prefix Digits (Outgoing Calls)	+	
Called Party Number Type *	National	
Called Party Numbering Plan *	ISDN	

HQ International (with trailing #)

Copy

Translation Pattern: **9.011!#**

Partition: **HQ**

Calling Search Space: **PSTN**

Urgent Priority: **Unchecked**

Use Calling Party's External Phone Number Mask: **Checked**

Calling Party Transform Mask: *blank*

Calling Party Number Type: **International**

Calling Party Numbering Plan: **ISDN**

Discard Digits: **PreDot Trailing-#**

Prefix Digits (Outgoing Calls): **+**

Called Party Number Type: **International**

Called Party Numbering Plan: **ISDN**

Save

Translation Pattern Configuration

Related Links: [Back To Find/List](#) [Go](#)

Save

Pattern Definition

Translation Pattern

9.011!#

Partition

HQ

Description

Numbering Plan

< None >

Route Filter

< None >

MLPP Precedence *

Default

Resource Priority Namespace Network Domain

< None >

Calling Search Space

PSTN

Route Option

☒ Route this pattern

☐ Block this pattern No Error

☒ Provide Outside Dial Tone

☒ Urgent Priority

Calling Party Transformations

☒ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation *

Default

Calling Name Presentation *

Default

Calling Party Number Type *

International

Calling Party Numbering Plan *

ISDN

Called Party Transformations

Discard Digits

PreDot Trailing-#

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

+

Called Party Number Type *

National

Called Party Numbering Plan *

ISDN

Configure the HQ gateway to strip the leading + from the DNIS.

At this point in our configuration a generic Route Pattern of \+! is sending a call to the HQ gateway. However, the leading + should be stripped off the DNIS before sending the call out to the PSTN. This can be accomplished with a **Called Party Transformation Pattern**.

Call Routing > Transformation Pattern > Called Party Transformation Pattern > Add New Pattern: \+.
Partition: STRIP
Discard Digits: PreDot
Save

The screenshot shows the 'Called Party Transformation Pattern Configuration' page. At the top, there is a navigation bar with menus: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. Below the navigation bar, the page title is 'Called Party Transformation Pattern Configuration'. On the right, there are 'Related Links: Back To Find/List' and a 'Go' button. The main content area has a 'Save' button and a 'Pattern Definition' section. In the 'Pattern Definition' section, the 'Pattern*' field is set to '\+.', the 'Partition' dropdown is set to 'STRIP', and the 'Urgent Priority' checkbox is checked. Below this is the 'Called Party Transformations' section, where 'Discard Digits' is set to 'PreDot', 'Called Party Transformation Mask' is empty, 'Prefix Digits' is empty, 'Called Party Number Type*' is set to 'Cisco CallManager', and 'Called Party Numbering Plan*' is set to 'Cisco CallManager'.

This **Called Party Transformation Pattern** belongs to the **STRIP Partition**, which belongs to the **STRIP Calling Search Space**. This **Calling Search Space** now needs to be assigned as the **HQ gateway's Called Party Transformation Pattern Calling Search Space**.

Device > Gateway > Find > 10.10.20.1
Called Party Transformation CSS: HQ-DNIS-OUT
Use Device Pool Called Party Transformation CSS: Unchecked
Save
OK
Reset
Reset
Close

The screenshot shows the 'Gateway Configuration' page. At the top, there is a navigation bar with menus: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. Below the navigation bar, the page title is 'Gateway Configuration'. On the right, there are 'Related Links: Back To Find/List' and a 'Go' button. The main content area has a toolbar with buttons: Save, Delete, Copy, Reset, and Add New. Below the toolbar, the 'Called Party Transformation CSS' dropdown is set to 'HQ-DNIS-OUT'. There is a checkbox for 'Use Device Pool Called Party Transformation CSS' which is unchecked. Below this, the 'Calling Party Transformation CSS' dropdown is set to '< None >'. At the bottom, there is a checkbox for 'Use Device Pool Calling Party Transformation CSS' which is checked.

Verification

Although most lab verification tasks are postponed until all lab tasks are completed, call routing is an exception. The reason is, if call routing doesn't work as required many other points could be lost. Therefore, we now take time to verify the ANI and DNIS sent to the PSTN for a series of calls from an HQ phone.

This verification is done by examining the output of the **debug isdn q931** command issued on the HQ gateway.

HQ Emergency Test Call

HQ#**debug isdn q931**

```
Jul 30 19:34:11.231: ISDN Se0/0/0:23 Q931: pak_private_number: Invalid
type/plan 0x0 0x1 may be overridden; sw-type 13
Jul 30 19:34:11.231: ISDN Se0/0/0:23 Q931: Sending SETUP callref = 0x0080
callID = 0x8001 switch = primary-ni interface = User
Jul 30 19:34:11.231: ISDN Se0/0/0:23 Q931: TX -> SETUP pd = 8 callref =
0x0080
    Bearer Capability i = 0x8090A2
        Standard = CCITT
        Transfer Capability = Speech
        Transfer Mode = Circuit
        Transfer Rate = 64 kbit/s
    Channel ID i = 0xA98383
        Exclusive, Channel 3
    Display i = 'HQ Phone 2'
    Calling Party Number i = 0x2181, '4082222002'
        Plan:ISDN, Type:National
    Called Party Number i = 0x80, '911'
        Plan:Unknown, Type:Unknown
Jul 30 19:34:11.255: ISDN Se0/0/0:23 Q931: RX <- CALL_PROC pd = 8 callref =
0x8080
    Channel ID i = 0xA98383
        Exclusive, Channel 3
Jul 30 19:34:11.255: ISDN Se0/0/0:23 Q931: RX <- ALERTING pd = 8 callref =
0x8080
    Progress Ind i = 0x8188- In-band info or appropriate now available
HQ#
```

HQ Local Test Call

HQ#

```
Jul 30 19:39:40.090: ISDN Se0/0/0:23 Q931: Applying typeplan for sw-type 0xD
is 0x4 0x1, Calling num 2222002
Jul 30 19:39:40.090: ISDN Se0/0/0:23 Q931: Sending SETUP callref = 0x0082
callID = 0x8003 switch = primary-ni interface = User
Jul 30 19:39:40.090: ISDN Se0/0/0:23 Q931: TX -> SETUP pd = 8 callref =
0x0082
    Bearer Capability i = 0x8090A2
        Standard = CCITT
        Transfer Capability = Speech
```

```

        Transfer Mode = Circuit
        Transfer Rate = 64 kbit/s
    Channel ID i = 0xA98383
        Exclusive, Channel 3
    Display i = 'HQ Phone 2'
    Calling Party Number i = 0x4181, '2222002'
        Plan:ISDN, Type:Subscriber(local)
    Called Party Number i = 0xC1, '2222020'
        Plan:ISDN, Type:Subscriber(local)
Jul 30 19:39:40.110: ISDN Se0/0/0:23 Q931: RX <- CALL_PROC pd = 8  callref =
0x8082
    Channel ID i = 0xA98383
        Exclusive, Channel 3
Jul 30 19:39:40.114: ISDN Se0/0/0:23 Q931: RX <- ALERTING pd = 8  callref =
0x8082
    Progress Ind i = 0x8188 - In-band info or appropriate now available
Jul 30 19:39:40.882: ISDN Se0/0/0:23 Q931: TX -> DISCONNECT pd = 8  callref =
0x0082
    Cause i = 0x8090 - Normal call clearing
Jul 30 19:39:40.890: ISDN Se0/0/0:23 Q931: RX <- RELEASE pd = 8  callref =
0x8082
Jul 30 19:39:40.890: ISDN Se0/0/0:23 Q931: TX -> RELEASE_COMP pd = 8  callref
= 0x0082
HQ#

```

HQ Long Distance Test Call

```

HQ#
Jul 30 19:46:31.270: ISDN Se0/0/0:23 Q931: Applying typeplan for sw-type 0xD
is 0x2 0x1, Calling num 4082222002
Jul 30 19:46:31.270: ISDN Se0/0/0:23 Q931: Sending SETUP  callref = 0x0084
callID = 0x8005 switch = primary-ni interface = User
Jul 30 19:46:31.270: ISDN Se0/0/0:23 Q931: TX -> SETUP pd = 8  callref =
0x0084
    Bearer Capability i = 0x8090A2
        Standard = CCITT
        Transfer Capability = Speech
        Transfer Mode = Circuit
        Transfer Rate = 64 kbit/s
    Channel ID i = 0xA98383
        Exclusive, Channel 3
    Display i = 'HQ Phone 2'
    Calling Party Number i = 0x2181, '4082222002'
        Plan:ISDN, Type:National
    Called Party Number i = 0xA1, '18593333030'
        Plan:ISDN, Type:National
Jul 30 19:46:31.290: ISDN Se0/0/0:23 Q931: RX <- CALL_PROC pd = 8  callref =
0x8084
    Channel ID i = 0xA98383
        Exclusive, Channel 3
Jul 30 19:46:31.294: ISDN Se0/0/0:23 Q931: RX <- ALERTING pd = 8  callref =
0x8084
    Progress Ind i = 0x8188 - In-band info or appropriate now available
Jul 30 19:46:31.854: ISDN Se0/0/0:23 Q931: TX -> DISCONNECT pd = 8  callref =
0x0084
    Cause i = 0x8090 - Normal call clearing

```

```

Jul 30 19:46:31.862: ISDN Se0/0/0:23 Q931: RX <- RELEASE pd = 8   callref =
0x8084
Jul 30 19:46:31.862: ISDN Se0/0/0:23 Q931: TX -> RELEASE_COMP pd = 8   callref
= 0x0084
HQ#

```

HQ International Test Call (without using # to terminate the interdigit timeout)

```

HQ#
Jul 30 19:49:59.496: ISDN Se0/0/0:23 Q931: Applying typeplan for sw-type 0xD
is 0x1 0x1, Calling num 14082222002
Jul 30 19:49:59.500: ISDN Se0/0/0:23 Q931: Sending SETUP   callref = 0x0085
callID = 0x8006 switch = primary-ni interface = User
Jul 30 19:49:59.500: ISDN Se0/0/0:23 Q931: TX -> SETUP pd = 8   callref =
0x0085
    Bearer Capability i = 0x8090A2
        Standard = CCITT
        Transfer Capability = Speech
        Transfer Mode = Circuit
        Transfer Rate = 64 kbit/s
    Channel ID i = 0xA98383
        Exclusive, Channel 3
    Display i = 'HQ Phone 2'
    Calling Party Number i = 0x1181, '14082222002'
        Plan:ISDN, Type:International
    Called Party Number i = 0x91, '0118144444040'
        Plan:ISDN, Type:International
Jul 30 19:49:59.520: ISDN Se0/0/0:23 Q931: RX <- CALL_PROC pd = 8   callref =
0x8085
    Channel ID i = 0xA98383
        Exclusive, Channel 3
Jul 30 19:49:59.524: ISDN Se0/0/0:23 Q931: RX <- ALERTING pd = 8   callref =
0x8085
    Progress Ind i = 0x8188 - In-band info or appropriate now available
Jul 30 19:50:01.432: ISDN Se0/0/0:23 Q931: TX -> DISCONNECT pd = 8   callref =
0x0085
    Cause i = 0x8090 - Normal call clearing
Jul 30 19:50:01.440: ISDN Se0/0/0:23 Q931: RX <- RELEASE pd = 8   callref =
0x8085
Jul 30 19:50:01.440: ISDN Se0/0/0:23 Q931: TX -> RELEASE_COMP pd = 8   callref
= 0x0085
HQ#

```

TROUBLESHOOTING ISSUE:

Note that the ANI is missing a leading +, which is what it needs to be in E.164 format. This is an issue with H.323 gateways. To fix the issue, we can create a voice translation rule to prepend the + for outgoing international calls.

HQ#**conf term**

Enter configuration commands, one per line. End with CNTL/Z.

HQ(config)#**voice translation-rule 2**


```

HQ(cfg-translation-rule)#rule 1 // /\0/ type international
international
HQ(cfg-translation-rule)#exit
HQ(config)#voice translation-profile PREFIX_PLUS
HQ(cfg-translation-profile)#translate calling 2
HQ(cfg-translation-profile)#exit
HQ(config)#dial-peer voice 4
HQ(config-dial-peer)#translation-profile out PREFIX_PLUS
HQ(config-dial-peer)#end
HQ#

```

Now, we place the test call again. This time the ANI does have a leading +.

```

HQ#
Jul 30 19:57:51.776: ISDN Se0/0/0:23 Q931: Applying typeplan for sw-type 0xD
is 0x1 0x1, Calling num +14082222002
Jul 30 19:57:51.776: ISDN Se0/0/0:23 Q931: Sending SETUP callref = 0x0086
callID = 0x8007 switch = primary-ni interface = User
Jul 30 19:57:51.776: ISDN Se0/0/0:23 Q931: TX -> SETUP pd = 8 callref =
0x0086
    Bearer Capability i = 0x8090A2
        Standard = CCITT
        Transfer Capability = Speech
        Transfer Mode = Circuit
        Transfer Rate = 64 kbit/s
    Channel ID i = 0xA98383
        Exclusive, Channel 3
    Display i = 'HQ Phone 2'
    Calling Party Number i = 0x1181, '+14082222002'
        Plan:ISDN, Type:International
    Called Party Number i = 0x91, '0118144444040'
        Plan:ISDN, Type:International
Jul 30 19:57:51.800: ISDN Se0/0/0:23 Q931: RX <- CALL_PROC pd = 8 callref =
0x8086
    Channel ID i = 0xA98383
        Exclusive, Channel 3
Jul 30 19:57:51.800: ISDN Se0/0/0:23 Q931: RX <- ALERTING pd = 8 callref =
0x8086
    Progress Ind i = 0x8188- In-band info or appropriate now available
HQ#

```

HQ International Test Call (using # to terminate the interdigit timeout)

```

HQ#
Jul 30 20:03:47.860: ISDN Se0/0/0:23 Q931: Applying typeplan for sw-type 0xD
is 0x1 0x1, Calling num +14082222002
Jul 30 20:03:47.860: ISDN Se0/0/0:23 Q931: Sending SETUP callref = 0x0088
callID = 0x8009 switch = primary-ni interface = User
Jul 30 20:03:47.860: ISDN Se0/0/0:23 Q931: TX -> SETUP pd = 8 callref =
0x0088
    Bearer Capability i = 0x8090A2
        Standard = CCITT
        Transfer Capability = Speech

```

```

        Transfer Mode = Circuit
        Transfer Rate = 64 kbit/s
    Channel ID i = 0xA98383
        Exclusive, Channel 3
    Display i = 'HQ Phone 2'
    Calling Party Number i = 0x1181, '+14082222002'
        Plan:ISDN, Type:International
    Called Party Number i = 0x91, '0118144444040'
        Plan:ISDN, Type:International
Jul 30 20:03:47.884: ISDN Se0/0/0:23 Q931: RX <- CALL_PROC pd = 8  callref =
0x8088
    Channel ID i = 0xA98383
        Exclusive, Channel 3
Jul 30 20:03:47.884: ISDN Se0/0/0:23 Q931: RX <- ALERTING pd = 8  callref =
0x8088
    Progress Ind i = 0x8188 - In-band info or appropriate now available
Jul 30 20:03:49.152: ISDN Se0/0/0:23 Q931: TX -> DISCONNECT pd = 8  callref =
0x0088
    Cause i = 0x8090 - Normal call clearing
Jul 30 20:03:49.160: ISDN Se0/0/0:23 Q931: RX <- RELEASE pd = 8  callref =
0x8088
Jul 30 20:03:49.160: ISDN Se0/0/0:23 Q931: TX -> RELEASE_COMP pd = 8  callref
= 0x0088
HQ#

```

Also, place a call from a PSTN phone to an HQ IP phone, in order to confirm that the HQ gateway can receive incoming calls from the PSTN.

Task 4.2 BR1 Call Routing

Create Route Groups


Similar to what we did at the HQ site, we have configured the BR1 gateway, and it can be added to a Route Group, which can then be assigned as a Local Route Group to the BR1 Device Pool.


Call Routing > Route/Hunt > Route Group Add New

Route Group Name: **BR1**
 Select **S0/SU0/DS1-0@BR1.kwtrain.com**
 Add to Route Group
 Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Group Configuration

 Save

Status
 Status: Ready

Route Group Information
Route Group Name*
Distribution Algorithm*

Route Group Member Information
Find Devices to Add to Route Group
Device Name contains Find
Available Devices**
Port(s)

Current Route Group Members
Selected Devices (ordered by priority)*

Removed Devices***

Assign Local Route Groups

System > Device Pool > Find > BR1

Local Route Group: **BR1**

Save







Reset

Reset


Close

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Device Pool Configuration

 Save
  Delete
  Copy
  Reset
  Apply Config
  Add New


Status

 Status: Ready

Device Pool Information

Device Pool: BR1 (7 members**)

Device Pool Settings

Device Pool Name*	BR1 
Cisco Unified Communications Manager Group*	BR1 ▾
Calling Search Space for Auto-registration	< None > ▾
Adjunct CSS	< None > ▾
Reverted Call Focus Priority	Default ▾
Intercompany Media Services Enrolled Group	< None > ▾

Roaming Sensitive Settings

Date/Time Group*	BR1 ▾
Region*	BR1 ▾
Media Resource Group List	BR1 ▾
Location	Hub_None ▾
Network Locale	< None > ▾
SRST Reference*	Disable ▾
Connection Monitor Duration***	
Single Button Barge*	Default ▾
Join Across Lines*	Default ▾
Physical Location	< None > ▾
Device Mobility Group	< None > ▾
Local Route Group	BR1 ▾

Create a Route List that contains the Standard Local Route Group

Call Routing > Route/Hunt > Route List

Add New


Name: **SLRG**

Cisco Unified Communications Manager Group: **SUB-PUB**


Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Route List Configuration

 Save

Status

 Status: Ready


Route List Information

☒ Device is trusted

Name*

Description

Cisco Unified Communications Manager Group*

 Save

Add Route Group

Add Route Group


Route Group: **Standard Local Route Group**

Save


OK

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Route List Detail Configuration

 Save

Status


 Status: Ready

Route List Member Information

Route Group* Standard Local Route Group ▾

Calling Party Transformations

Use Calling Party's External Phone Number Mask* Default ▾

Calling Party Transform Mask 

Prefix Digits (Outgoing Calls)

Calling Party Number Type* Cisco CallManager ▾

Calling Party Numbering Plan* Cisco CallManager ▾

Called Party Transformations

Discard Digits < None > ▾

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type* Cisco CallManager ▾

Called Party Numbering Plan* Cisco CallManager ▾

Save

Save
OK

Create a generic Route Pattern to be used by multiple Translation Patterns

Call Routing > Route/Hunt > Route Pattern
Add New

Route Pattern: \+!
Route Partition: PSTN
Gateway/Route List: SLRG
Urgent Priority: Checked
Save
OK
OK

System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help

Route Pattern Configuration

Save

Status
i Status: Ready

Pattern Definition
Route Pattern* \+!
Route Partition PSTN
Description
Numbering Plan -- Not Selected --
Route Filter < None >
MLPP Precedence* Default
☐ Apply Call Blocking Percentage
Resource Priority Namespace Network Domain < None >
Route Class* Default
Gateway/Route List* SLRG (Edit)
Route Option
☒ Route this pattern
☐ Block this pattern No Error
Call Classification* OffNet
☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☒ Urgent Priority
☐ Require Forced Authorization Code
Authorization Level* 0
☐ Require Client Matter Code

Just as we documented a route plan for the HQ and BR2 sites, we now need to document the route plan for BR1.

BR1 Route Plan

Pattern	Route List	Route Group	ANI/TON	DNIS/TON	DM
911	SLRG	LRG	10/N	911/U	None
9.[2-9]XXXXXX	SLRG	LRG	7/S	7/S	PD
91.[2-9]XX[2-9]XXXXXX	SLRG	LRG	10/N	10/N	PD
9011.!	SLRG	LRG	E.164/I	Var./I	PD
9011.!#	SLRG	LRG	E.164/I	Var./I	PD, Trailing #

BR1 Emergency

Call Routing > Translation Pattern > Add New

Translation Pattern: **911**

Partition: **BR1**

Calling Search Space: **PSTN**

Use Calling Party's External Phone Number Mask: **Checked**

Calling Party Transform Mask: **XXXXXXXXXX**

Calling Party Number Type: **National**

Calling Party Numbering Plan: **ISDN**

Prefix Digits (Outgoing Calls): **+**

Called Party Number Type: **Unknown**

Called Party Numbering Plan: **ISDN**

Save

System

Call Routing

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Advanced Features

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User Management

Bulk Administration

Translation Pattern Configuration

Save

Delete

Copy

Add New

Pattern Definition

Translation Pattern

911

Partition

BR1

Description

Numbering Plan

< None >

Route Filter

< None >

MLPP Precedence*

Default

Resource Priority Namespace Network Domain

< None >

Route Class*

Default

Calling Search Space

PSTN

External Call Control Profile

< None >

Route Option

Route this pattern

Block this pattern

No Error

☒ Provide Outside Dial Tone

☒ Urgent Priority

☐ Route Next Hop By Calling Party Number

Calling Party Transformations

☒ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

XXXXXXXXXX

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Default

Calling Name Presentation*

Default

Calling Party Number Type*

National

Calling Party Numbering Plan*

ISDN

Connected Party Transformations

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Called Party Transformations

Discard Digits < None >

Called Party Transform Mask

Prefix Digits (Outgoing Calls) +

Called Party Number Type* Unknown

Called Party Numbering Plan* ISDN

Save Delete Copy Add New

BR1 Local

Copy

Translation Pattern: 9.[2-9]XXXXXX
Partition: BR1
Calling Search Space: PSTN
Urgent Priority: Unchecked
Use Calling Party’s External Phone Number Mask: Checked
Calling Party Transform Mask: XXXXXXXX
Calling Party Number Type: Subscriber
Calling Party Numbering Plan: ISDN
Discard Digits: PreDot
Prefix Digits (Outgoing Calls): +
Called Party Number Type: Subscriber
Called Party Numbering Plan: ISDN
Save

Translation Pattern Configuration

 Save
  Delete
  Copy
  Add New

Pattern Definition

Translation Pattern	9.[2-9]XXXXXX
Partition	BR1
Description	
Numbering Plan	< None >
Route Filter	< None >
MLPP Precedence*	Default
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Calling Search Space	PSTN
External Call Control Profile	< None >
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern
	No Error

☒ Provide Outside Dial Tone
☐ Urgent Priority
☐ Route Next Hop By Calling Party Number

Calling Party Transformations

☒ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask	XXXXXX
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	Subscriber
Calling Party Numbering Plan*	ISDN

Connected Party Transformations

Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

Called Party Transformations

Discard Digits	PreDot
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	+
Called Party Number Type*	Subscriber
Called Party Numbering Plan*	ISDN

- Save Delete Copy Add New -

BR1 Long Distance

Copy

Translation Pattern: **91.[2-9]XX[2-9]XXXXXX**

Partition: **BR1**

Calling Search Space: **PSTN**

Urgent Priority: **Unchecked**

Use Calling Party's External Phone Number Mask: **Checked**

Calling Party Transform Mask: **XXXXXXXXXX**

Calling Party Number Type: **National**

Calling Party Numbering Plan: **ISDN**

Discard Digits: **PreDot**

Prefix Digits (Outgoing Calls): **+**





Called Party Number Type: **National**

Called Party Numbering Plan: **ISDN**

Save

System
Call Routing
Media Resources
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Translation Pattern Configuration

 Save
 Delete
 Copy
 Add New

Pattern Definition

Translation Pattern	91.[2-9]XX[2-9]XXXXXX
Partition	BR1
Description	
Numbering Plan	< None >
Route Filter	< None >
MLPP Precedence*	Default
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Calling Search Space	PSTN
External Call Control Profile	< None >
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <div>No Error</div>

☒ Provide Outside Dial Tone
☐ Urgent Priority
☐ Route Next Hop By Calling Party Number

Calling Party Transformations

☒ Use Calling Party's External Phone Number Mask
 Calling Party Transform Mask: XXXXXXXXXX
 Prefix Digits (Outgoing Calls):
 Calling Line ID Presentation*: Default
 Calling Name Presentation*: Default
 Calling Party Number Type*: National
 Calling Party Numbering Plan*: ISDN

Connected Party Transformations

Connected Line ID Presentation*: Default
 Connected Name Presentation*: Default

Called Party Transformations

Discard Digits: PreDot
 Called Party Transform Mask:
 Prefix Digits (Outgoing Calls): +
 Called Party Number Type*: National
 Called Party Numbering Plan*: ISDN

Save
Delete
Copy
Add New

BR1 International (without trailing #)

Copy

Translation Pattern: **9011.!**

Partition: **HQ**

Calling Search Space: **PSTN**

Urgent Priority: **Unchecked**

Use Calling Party's External Phone Number Mask: **Checked**

Calling Party Transform Mask: *blank*

Calling Party Number Type: **International**

Calling Party Numbering Plan: **ISDN**

Discard Digits: **PreDot**

Prefix Digits (Outgoing Calls): **+**





Called Party Number Type: **International**

Called Party Numbering Plan: **ISDN**

Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾

Translation Pattern Configuration

 Save  Delete  Copy  Add New

Pattern Definition

Translation Pattern	9011.!
Partition	BR1
Description	
Numbering Plan	< None >
Route Filter	< None >
MLPP Precedence*	Default
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Calling Search Space	PSTN
External Call Control Profile	< None >
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error

☒ Provide Outside Dial Tone
☐ Urgent Priority
☐ Route Next Hop By Calling Party Number

Calling Party Transformations

☒ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	International
Calling Party Numbering Plan*	ISDN

Connected Party Transformations

Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

Called Party Transformations

Discard Digits	PreDot
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	+
Called Party Number Type*	International
Called Party Numbering Plan*	ISDN

– Save Delete Copy Add New

BR1 International (with trailing #)

Copy

Translation Pattern: **9.011!#**

Partition: **BR1**

Calling Search Space: **PSTN**

Urgent Priority: **Unchecked**

Use Calling Party's External Phone Number Mask: **Checked**

Calling Party Transform Mask: *blank*

Calling Party Number Type: **International**

Calling Party Numbering Plan: **ISDN**

Discard Digits: **PreDot Trailing-#**

Prefix Digits (Outgoing Calls): **+**





Called Party Number Type: **International**

Called Party Numbering Plan: **ISDN**

Save

System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration

Translation Pattern Configuration

 Save
 Delete
 Copy
 Add New

Pattern Definition

Translation Pattern	9011.1#
Partition	BR1
Description	
Numbering Plan	< None >
Route Filter	< None >
MLPP Precedence*	Default
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Calling Search Space	PSTN
External Call Control Profile	< None >
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern

☒ Provide Outside Dial Tone
☐ Urgent Priority
☐ Route Next Hop By Calling Party Number

Calling Party Transformations

☒ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	International
Calling Party Numbering Plan*	ISDN

Connected Party Transformations





Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

Connected Party Transformations

Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

Called Party Transformations

Discard Digits	PreDot Trailing-#
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	+
Called Party Number Type*	International
Called Party Numbering Plan*	ISDN

 Save
 Delete
 Copy
 Add New

Configure the BR1 gateway to strip the leading + from the DNIS.

At this point in our configuration a generic Route Pattern of \+! is sending a call to the BR1 gateway. However, the leading + should be stripped off the DNIS before sending the call out to the PSTN. This can be accomplished with a **Called Party Transformation Pattern**.

Call Routing > Transformation > Transformation Pattern > Called Party Transformation Pattern > Add New

Pattern: \+!

Partition: **STRIP**

Discard Digits: **PreDot**

Save

System ▾

Call Routing ▾

Media Resources ▾


Advanced Features ▾

Device ▾


Application ▾

User Management ▾

Called Party Transformation Pattern Configuration

 Save


Status

 Status: Ready

Pattern Definition


Pattern*

\+!



Partition


STRIP



Description


Numbering Plan

< None >



Route Filter

< None >




☒ Urgent Priority

Called Party Transformations

Discard Digits

PreDot




Called Party Transformation Mask

Prefix Digits


Called Party Number Type*

Cisco CallManager



Called Party Numbering Plan*

Cisco CallManager



Save

This **Called Party Transformation Pattern** belongs to the **STRIP Partition**, which belongs to the **BR1-DNIS-OUT Calling Search Space**. This **Calling Search Space** now needs to be assigned as the **BR1 gateway's Called Party Transformation Pattern Calling Search Space**.

Device > Gateway > Find > BR1.kwtrain.com > 0/0/0

Called Party Transformation CSS: **HQ-DNIS-OUT**

Use Device Pool Called Party Transformation CSS: **Unchecked**

Save

OK

Reset

Reset

Close

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾

Gateway Configuration

Save Delete Reset Apply Config Add New

Call Routing Information - Outbound Calls

Calling Party Presentation *	Default ▾
Calling Party Selection *	Originator ▾
Called party IE number type unknown *	Cisco CallManager ▾
Calling party IE number type unknown *	Cisco CallManager ▾
Called Numbering Plan *	Cisco CallManager ▾
Calling Numbering Plan *	Cisco CallManager ▾
Number of digits to strip *	0 ▾
Caller ID DN	
SMDI Base Port *	0
Called Party Transformation CSS	BR1-DNIS-OUT ▾
<input type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None > ▾
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	

Verification

Although most lab verification tasks are postponed until all lab tasks are completed, call routing is an exception. The reason is, if call routing doesn't work as required many other points could be lost. Therefore, we now take time to verify the ANI and DNIS sent to the PSTN for a series of calls from an HQ phone.

This verification is done by examining the output of the **debug isdn q931** command issued on the BR1 gateway.

BR1 Emergency Test Call:

```
BR1# debug isdn q931
debug isdn q931 is ON.
BR1#
Jul 29 21:43:49.482: ISDN Se0/0/0:23 Q931: TX -> SETUP pd = 8 callref =
0x0002
    Bearer Capability i = 0x8090A2
```

```

        Standard = CCITT
        Transfer Capability = Speech
        Transfer Mode = Circuit
        Transfer Rate = 64 kbit/s
    Channel ID i = 0xA98383
        Exclusive, Channel 3
    Display i = 'BR1 Phone 2'
    Calling Party Number i = 0x2181, '8595553002'
        Plan:ISDN, Type:National
    Called Party Number i = 0x81, '911'
        Plan:ISDN, Type:Unknown
Jul 29 21:43:49.502: ISDN Se0/0/0:23 Q931: RX <- CALL_P
BR1#ROC pd = 8 callref = 0x8002
    Channel ID i = 0xA98383
        Exclusive, Channel 3
Jul 29 21:43:49.506: ISDN Se0/0/0:23 Q931: RX <- ALERTING pd = 8 callref =
0x8002
    Progress Ind i = 0x8188 - In-band info or appropriate now available
BR1#
Jul 29 21:43:51.338: ISDN Se0/0/0:23 Q931: TX -> DISCONNECT pd = 8 callref =
0x0002
    Cause i = 0x8090 - Normal call clearing
Jul 29 21:43:51.342: ISDN Se0/0/0:23 Q931: RX <- RELEASE pd = 8 callref =
0x8002
Jul 29 21:43:51.366: ISDN Se0/0/0:23 Q931: TX -> RELEASE_COMP pd = 8 callref
= 0x0002

```

BR1 Local Test Call

```

BR1#
Jul 29 21:50:27.443: ISDN Se0/0/0:23 Q931: TX -> SETUP pd = 8 callref =
0x0008
    Bearer Capability i = 0x8090A2
        Standard = CCITT
        Transfer Capability = Speech
        Transfer Mode = Circuit
        Transfer Rate = 64 kbit/s
    Channel ID i = 0xA98383
        Exclusive, Channel 3
    Display i = 'BR1 Phone 2'
    Calling Party Number i = 0x4181, '3333002'
        Plan:ISDN, Type:Subscriber(local)
    Called Party Number i = 0xC1, '3333030'
        Plan:ISDN, Type:Subscriber(local)
Jul 29 21:50:27.463: ISDN Se0/0/0:2
BR1#3 Q931: RX <- CALL_PROC pd = 8 callref = 0x8008
    Channel ID i = 0xA98383
        Exclusive, Channel 3
Jul 29 21:50:27.463: ISDN Se0/0/0:23 Q931: RX <- ALERTING pd = 8 callref =
0x8008
    Progress Ind i = 0x8188 - In-band info or appropriate now available
BR1#
Jul 29 21:50:28.663: ISDN Se0/0/0:23 Q931: TX -> DISCONNECT pd = 8 callref =
0x0008
    Cause i = 0x8090 - Normal call clearing

```

```
Jul 29 21:50:28.671: ISDN Se0/0/0:23 Q931: RX <- RELEASE pd = 8 callref = 0x8008
Jul 29 21:50:28.687: ISDN Se0/0/0:23 Q931: TX -> RELEASE_COMP pd = 8 callref = 0x0008
```

BR1 Long Distance Test Call

```
BR1#
Jul 29 21:55:01.327: ISDN Se0/0/0:23 Q931: TX -> SETUP pd = 8 callref = 0x0009
    Bearer Capability i = 0x8090A2
        Standard = CCITT
        Transfer Capability = Speech
        Transfer Mode = Circuit
        Transfer Rate = 64 kbit/s
    Channel ID i = 0xA98383
        Exclusive, Channel 3
    Display i = 'BR1 Phone 2'
    Calling Party Number i = 0x2181, '8593333002'
        Plan:ISDN, Type:National
    Called Party Number i = 0xA1, '4082222020'
        Plan:ISDN, Type:National
Jul 29 21:55:01.351: ISDN Se0/0/0:23 Q931: RX <
BR1#- CALL_PROC pd = 8 callref = 0x8009
    Channel ID i = 0xA98383
        Exclusive, Channel 3
Jul 29 21:55:01.351: ISDN Se0/0/0:23 Q931: RX <- ALERTING pd = 8 callref = 0x8009
    Progress Ind i = 0x8188 - In-band info or appropriate now available
BR1#
Jul 29 21:55:05.179: ISDN Se0/0/0:23 Q931: TX -> DISCONNECT pd = 8 callref = 0x0009
    Cause i = 0x8090 - Normal call clearing
Jul 29 21:55:05.187: ISDN Se0/0/0:23 Q931: RX <- RELEASE pd = 8 callref = 0x8009
Jul 29 21:55:05.207: ISDN Se0/0/0:23 Q931: TX -> RELEASE_COMP pd = 8 callref = 0x0009
BR1#
```

BR1 International Test Call (using # to terminate the interdigit timeout):

```
BR1#
Jul 29 21:57:20.875: ISDN Se0/0/0:23 Q931: TX -> SETUP pd = 8 callref = 0x000A
    Bearer Capability i = 0x8090A2
        Standard = CCITT
        Transfer Capability = Speech
        Transfer Mode = Circuit
        Transfer Rate = 64 kbit/s
    Channel ID i = 0xA98383
        Exclusive, Channel 3
    Display i = 'BR1 Phone 2'
    Calling Party Number i = 0x1181, '+18593333002'
        Plan:ISDN, Type:International
    Called Party Number i = 0x91, '8144444040'
```

```

Plan:ISDN, Type:International
Jul 29 21:57:20.895: ISDN Se0/0/0:2
BR1#3 Q931: RX <- CALL_PROC pd = 8 callref = 0x800A
Channel ID i = 0xA98383
Exclusive, Channel 3
Jul 29 21:57:20.899: ISDN Se0/0/0:23 Q931: RX <- ALERTING pd = 8 callref =
0x800A
Progress Ind i = 0x8188 - In-band info or appropriate now available
BR1#
Jul 29 21:57:22.719: ISDN Se0/0/0:23 Q931: TX -> DISCONNECT pd = 8 callref =
0x000A
Cause i = 0x8090 - Normal call clearing
Jul 29 21:57:22.727: ISDN Se0/0/0:23 Q931: RX <- RELEASE pd = 8 callref =
0x800A
Jul 29 21:57:22.751: ISDN Se0/0/0:23 Q931: TX -> RELEASE_COMP pd = 8 callref
= 0x000A
BR1#

```

BR1 International Test Call (without using # to terminate the interdigit timeout):

```

BR1#
Jul 29 22:00:50.316: ISDN Se0/0/0:23 Q931: TX -> SETUP pd = 8 callref =
0x000B
Bearer Capability i = 0x8090A2
Standard = CCITT
Transfer Capability = Speech
Transfer Mode = Circuit
Transfer Rate = 64 kbit/s
Channel ID i = 0xA98383
Exclusive, Channel 3
Display i = 'BR1 Phone 2'
Calling Party Number i = 0x1181, '+18593333002'
Plan:ISDN, Type:International
Called Party Number i = 0x91, '8144444040'
Plan:ISDN, Type:International
Jul 29 22:00:50.336: ISDN Se0/0/0:2
BR1#3 Q931: RX <- CALL_PROC pd = 8 callref = 0x800B
Channel ID i = 0xA98383
Exclusive, Channel 3
Jul 29 22:00:50.340: ISDN Se0/0/0:23 Q931: RX <- ALERTING pd = 8 callref =
0x800B
Progress Ind i = 0x8188 - In-band info or appropriate now available
BR1#
Jul 29 22:00:51.412: ISDN Se0/0/0:23 Q931: TX -> DISCONNECT pd = 8 callref =
0x000B
Cause i = 0x8090 - Normal call clearing
Jul 29 22:00:51.416: ISDN Se0/0/0:23 Q931: RX <- RELEASE pd = 8 callref =
0x800B
Jul 29 22:00:51.448: ISDN Se0/0/0:23 Q931: TX -> RELEASE_COMP pd = 8 callref
= 0x000B
BR1#

```

Make a call from a PSTN phone to a BR1 IP phone, to confirm incoming calls are functioning correctly.

Module 12: Number Globalization/Localization, URI Dialing, and Video Conferencing

Task 4.4 Number Globalization and Localization

Determine the current incoming ANI

```
HQ#debug isdn q931
debug isdn q931 is          ON.
HQ#
Aug  5 13:41:27.761: ISDN Se0/0/0:23 Q931: RX <- SETUP pd = 8  callref =
0x0080
    Bearer Capability i = 0x8090A2
        Standard = CCITT
        Transfer Capability = Speech
        Transfer Mode = Circuit
        Transfer Rate = 64 kbit/s
    Channel ID i = 0xA98381
        Exclusive, Channel 1
    Progress Ind i = 0x8583 - Origination address is non-ISDN
    Calling Party Number i = 0x4180, '2222020'
        Plan:ISDN, Type:Subscriber(local)
    Called Party Number i = 0xC1, '2222002'
        Plan:ISDN, Type:Subscriber(local)
Au
HQ#g  5 13:41:27.761: ISDN Se0/0/0:23 Q931: Received SETUP  callref = 0x8080
callID = 0x0001 switch = primary-ni interface = User
Aug  5 13:41:27.773: ISDN Se0/0/0:23 Q931: TX -> CALL_PROC pd = 8  callref =
0x8080
    Channel ID i = 0xA98381
        Exclusive, Channel 1
Aug  5 13:41:27.909: ISDN Se0/0/0:23 Q931: TX -> ALERTING pd = 8  callref =
0x8080
HQ#
Aug  5 13:41:31.193: ISDN Se0/0/0:23 Q931: RX <- DISCONNECT pd = 8  callref =
0x0080
    Cause i = 0x8290 - Normal call clearing
Aug  5 13:41:31.193: ISDN Se0/0/0:23 Q931: TX -> RELEASE pd = 8  callref =
0x8080
Aug  5 13:41:31.197: ISDN Se0/0/0:23 Q931: RX <- RELEASE_COMP pd = 8  callref
= 0x0080
HQ#
```

The call coming in from the PSTN has an ANI of 2222020 with a TON of Subscriber. However, we need to globalize (i.e. put in E.164 format) the ANI at the incoming gateway (i.e. the HQ gateway).

Globalize the ANI at the gateway

Device > Gateway > Find > 10.10.120.1

Incoming Calling Party Settings – Subscriber Number – Prefix: **+1408**

Save
Reset
Reset
Close

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Gateway Configuration Related Links: [Back To Find/List](#) [Go](#)

Save Delete Copy Reset Apply Config Add New

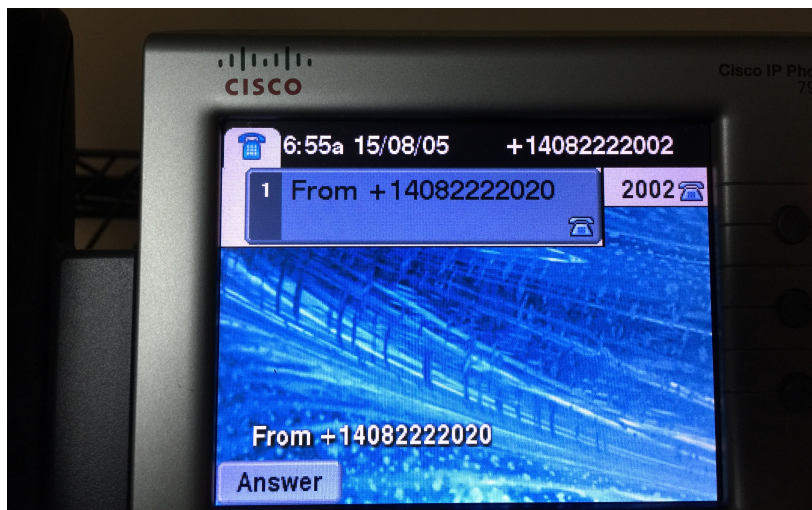
Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

☐ Clear Prefix Settings ☐ Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
National Number	Default	0	< None >	<input checked="" type="checkbox"/>
International Number	Default	0	< None >	<input checked="" type="checkbox"/>
Unknown Number	Default	0	< None >	<input checked="" type="checkbox"/>
Subscriber Number	+1408	0	< None >	<input checked="" type="checkbox"/>

At this point, when the PSTN phone with a phone number of 2222020 calls HQ Phone 2 (with a phone number of 2222002), the ANI appears on HQ Phone 2 in a fully globalized format (i.e. +14082222020) as shown below. While we want that fully globalized number stored in the CUCM database, we want a localized number to appear on the display of HQ Phone 2. This can be accomplished with a Calling Party Transformation Pattern.





Globalize the ANI at the gateway

Call Routing > Transformation > Transformation Pattern > Calling Party Transformation Pattern > Add New
 Pattern: \+1408.[2-9]XXXXXX
 Partition: HQ-ANI-IN
 Discard Digit Instructions: **PreDot**
Save


System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾


Calling Party Transformation Pattern Configuration

 Save


Status
 Status: Ready


Pattern Definition

Pattern* \+1408.[2-9]XXXXXX 

Partition HQ-ANI-IN 

Description


Numbering Plan < None > 

Route Filter < None > 

☒ Urgent Priority


Calling Party Transformations


☐ Use Calling Party's External Phone Number Mask


Discard Digits PreDot 


Calling Party Transformation Mask

Prefix Digits

Calling Line ID Presentation* Default 

Calling Party Number Type* Cisco CallManager 

Calling Party Numbering Plan* Cisco CallManager 

 Save

The **Partition** of **HQ-ANI-IN** belongs to the **Calling Search Space** of **HQ-ANI-IN**. Next, we can apply this Calling Search Space to all HQ IP phones by configuring the **Calling Party Transformation CSS** for the **HQ Device Pool** to **HQ-ANI-IN**.

System > Device Pool > HQ

Calling Party Transformation CSS: **HQ-ANI-IN**

Save







Reset

Reset


Close


System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾


Device Pool Configuration


 Save  Delete  Copy  Reset  Apply Config  Add New


Device Mobility Related Information****

Device Mobility Calling Search Space < None > 

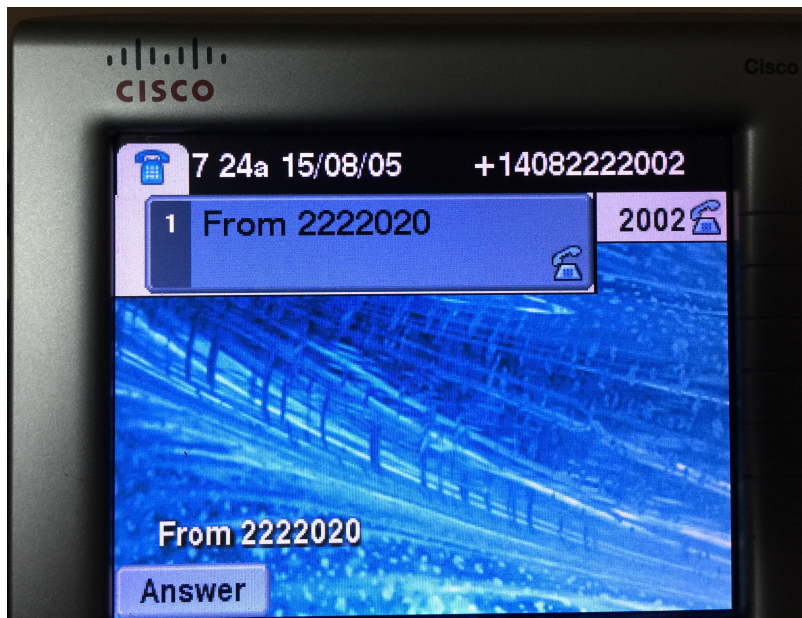
AAR Calling Search Space < None > 

AAR Group < None > 

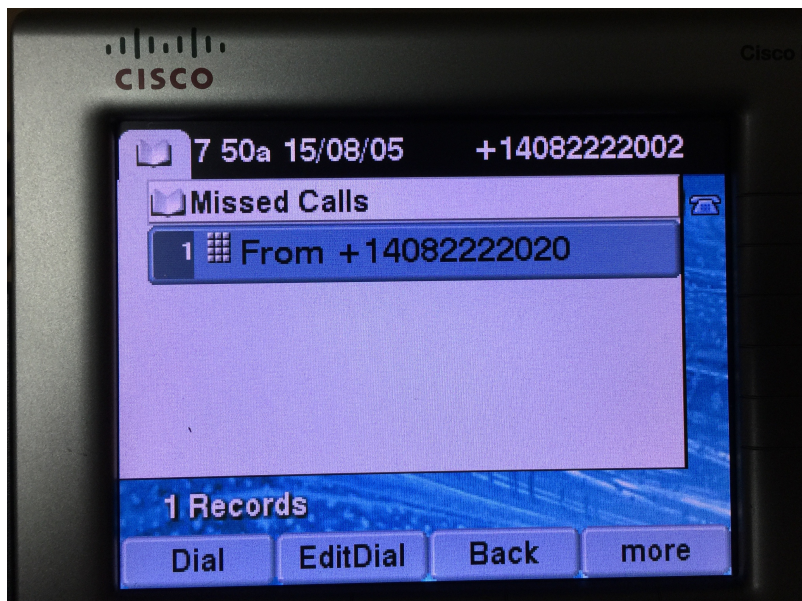
Calling Party Transformation CSS HQ-ANI-IN 

Called Party Transformation CSS < None > 

At this point, **HQ Phone 1** shows a **localized number** of **2222020** on its display when receiving a call from 2222020.



However, the **ANI** is displayed in a globalized format (i.e. **+14082222020**) under the Missed Calls directory.



Create a Route List and Route Pattern

We want a user to be able to select this globalized number from the **Missed Calls** directory, press the **Dial** softkey, and have the call placed via the HQ gateway, or via the BR1 gateway if the HQ gateway is not available. This redundancy scenario requires the configuration of an additional **Route List** and **Route Pattern**.

Call Routing > Route/Hunt > Route List > Add New

Name: **HQ-TO-LOCAL**

Cisco Unified Communications Manager Group: **SUB-PUB**

Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Route List Configuration

Save

Status

Status: Ready

Route List Information

☒ Device is trusted

Name*

Description

Cisco Unified Communications Manager Group*

Add Route Group

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Route List Configuration

Save Delete Copy Reset Apply Config Add New

Status

Add successful

Route List Information

Registration Unknown

IP Address Unknown

☒ Device is trusted

Name*

Description

Cisco Unified Communications Manager Group*

☒ Enable this Route List (change effective on Save; no reset required)

☐ Run On All Active Unified CM Nodes

Route List Member Information

Selected Groups**

Add Route Group

Route Group: **Standard Local Route Group**

Use Calling Party's External Phone Number Mask: **On**

Calling Party Transformation Mask: **XXXXXXX**

Calling Party Number Type: **Subscriber**

Calling Party Numbering Plan: **ISDN**

Save

OK

System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management

Route List Detail Configuration

Save

Status

i
Status: Ready

Route List Member Information

Route Group * Standard Local Route Group

Calling Party Transformations

Use Calling Party's External Phone Number Mask* On

Calling Party Transform Mask XXXXXXXX

Prefix Digits (Outgoing Calls)

Calling Party Number Type* Subscriber

Calling Party Numbering Plan* ISDN

Called Party Transformations

Discard Digits < None >

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type* Cisco CallManager

Called Party Numbering Plan* Cisco CallManager

Save

NOTE: While we do perform digit manipulation at the Route List/Route Group Details level for ANI, we don't do any digit manipulation here for DNIS information. Instead, DNIS digit manipulation will be performed by a **Called Party Transformation Pattern** at the **gateway** level.

Add Route Group

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Route List Configuration

Save
 Delete
 Copy
 Reset
 Apply Config
 Add New

Status

Add successful

Route List Information

Registration: Unregistered
 IP Address: 192.168.1.72
☒ Device is trusted
 Name*: HQ-TO-LOCAL
 Description:
 Cisco Unified Communications Manager Group*: SUB-PUB

☒ Enable this Route List (change effective on Save; no reset required)
☐ Run On All Active Unified CM Nodes

Route List Member Information

Selected Groups**
 Standard Local Route Group


Removed Groups***

Route List Details


Route Group: **SIP-TRUNK-TO-CUBE**
 Use Calling Party Transform Mask: **On**
 Calling Party Transform Mask: **XXXXXXXXXX**
 Calling Party Number Type: **National**
 Calling Party Numbering Plan: **ISDN**
Save
OK

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Route List Detail Configuration

 Save

Status

 Status: Ready

Route List Member Information

Route Group * SIP-TRUNK-TO-CUBE-[NON-QSIG] ⌵

Calling Party Transformations

Use Calling Party's External Phone Number Mask * On ⌵

Calling Party Transform Mask XXXXXXXXXX ⌵

Prefix Digits (Outgoing Calls)

Calling Party Number Type * National ⌵

Calling Party Numbering Plan * ISDN ⌵

Called Party Transformations


Discard Digits < None > ⌵

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type * Cisco CallManager ⌵







Called Party Numbering Plan * Cisco CallManager ⌵

 Save


Save
Reset
Reset
Close

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾

Route List Configuration

 Save  Delete  Copy  Reset  Apply Config  Add New

Status

 Add successful

Route List Information

Registration IP Address Registered with Cisco Unified Communications Manager 192.168.1.72
192.168.1.72

☒ Device is trusted

Name * HQ-TO-LOCAL


Description

Cisco Unified Communications Manager Group * SUB-PUB ⌵

☒ Enable this Route List (change effective on Save; no reset required)


☐ Run On All Active Unified CM Nodes


Route List Member Information

Selected Groups** Standard Local Route Group
SIP-TRUNK-TO-CUBE  Add Route Group

Removed Groups***

Route List Details

 Standard Local Route Group

 SIP-TRUNK-TO-CUBE

We now create a Route Pattern that matches the numbered called from HQ's call history and points to our newly created Route List.

Call Routing > Route/Hunt > Route Pattern > Add New

Route Pattern: \+1408[2-9]XXXXXX

Route Partition: HQ

Gateway/Route List: HQ-TO-LOCAL

Save

OK

OK

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾

Route Pattern Configuration

Save Delete Copy Add New

Pattern Definition

Route Pattern * \+1408[2-9]XXXXXX

Route Partition HQ

Description

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence * Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain < None >

Route Class * Default

Gateway/Route List * HQ-TO-LOCAL (Edit)

Route Option

☒ Route this pattern

☐ Block this pattern No Error

Call Classification * OffNet

☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level * 0

☐ Require Client Matter Code

Create Called Party Transformation Patterns

Now that the ANI has been formatted properly, and the Route List and Route Pattern have been created, we need to tell the HQ gateway (and the SIP trunk connecting to BR1 via the CUBE) how to manipulate the fully-globalized DNIS before sending the call out to the PSTN. This is done through the use of Called Party Transformation Patterns.

Call Routing > Transformation > Transformation Pattern > Called Party Transformation Pattern > Add New

Pattern: \+1408.[2-9]XXXXXX

Partition: HQ-DNIS-OUT

Save (NOTE: the configuration must be saved at this point before the Discard Digit Instruction (DDI) can be specified.)

Discard Digits: PreDot

Called Party Number Type: Subscriber

Called Party Numbering Plan: ISDN

Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Called Party Transformation Pattern Configuration

Save Delete Copy Add New

Status

Add successful

Pattern Definition

Pattern*

Partition

Description

Numbering Plan

Route Filter

☒ Urgent Priority

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type*

Called Party Numbering Plan*

Create Dial Peer on CUBE

If HQ Phone 2 dials call from its Missed Calls Directory menu, and the HQ gateway is not available, the call will fail over to the SIP trunk, which goes to the CUBE (i.e. router HQ). Therefore, we need to briefly go back to router HQ and add a dial peer to point calls the call to the CUCM9-PUB2 server at BR1.

```
HQ#conf term
Enter configuration commands, one per line. End with CNTL/Z.
HQ(config)#dial-peer voice 9 voip
HQ(config-dial-peer)#destination-pattern +1408[2-9].....$
HQ(config-dial-peer)#session protocol sipv2
HQ(config-dial-peer)#session target ipv4:10.10.150.71
HQ(config-dial-peer)#dtmf-relay sip-kpml
HQ(config-dial-peer)#codec transparent
HQ(config-dial-peer)#no vad
HQ(config-dial-peer)#end
HQ#
```

Create a Translation Pattern on CUCM9-PUB2

If the call does fail over to the BR1 site, the call flows from the CUCM cluster at HQ over a SIP trunk into the CUBE. The CUBE then uses the dial peer we just configured to forward the call over a SIP trunk to the BR1 CUCM server.

The format of the incoming DNIS is +1408[2-9]XXXXXX, and the call is being placed by the SIP trunk, which has a **CSS** of **CUBE** and a **Device Pool** of **CUBE**. We need something, a Translation

Pattern or a Route Pattern, to match the incoming dial string. We could make it work using either approach. However, we'll stay consistent with our approach of using Translation Patterns.

Call Routing > Translation Pattern > Add New

Translation Pattern: **\+1.408[2-9]XXXXXX**

Partition: **CUBE** (NOTE: This makes the Translation Pattern reachable by the SIP trunk.)

Calling Search Space: **PSTN** (NOTE: This lets the Translation Pattern match the **\+!** Route Pattern.)

Calling Party Number Type: **National**

Calling Party Numbering Plan: **ISDN**

Discard Digits: **PreDot** (NOTE: This gets rid of the **+1.**)

Prefix Digits (Outgoing Calls): **+** (NOTE: This lets the Translation Pattern match the **\+!** Route Pattern.)





Called Party Number Type: **National**

Called Party Numbering Plan: **ISDN**


Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Translation Pattern Configuration

 Save
  Delete
  Copy
  Add New

Status

 Status: Ready

Pattern Definition

Translation Pattern: \+1.408[2-9]XXXXXX

Partition: CUBE

Description:

Numbering Plan: < None >

Route Filter: < None >

MLPP Precedence*: Default

Resource Priority Namespace Network Domain: < None >

Route Class*: Default

Calling Search Space: PSTN

External Call Control Profile: < None >

Route Option:

☒ Route this pattern

☐ Block this pattern (No Error)

☒ Provide Outside Dial Tone

☒ Urgent Priority

☐ Route Next Hop By Calling Party Number

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask:

Prefix Digits (Outgoing Calls):

Calling Line ID Presentation*: Default

Calling Name Presentation*: Default

Calling Party Number Type*: National

Calling Party Numbering Plan*: ISDN

Connected Party Transformations

Connected Line ID Presentation*: Default

Connected Name Presentation*: Default

Called Party Transformations

Discard Digits: PreDot

Called Party Transform Mask:

Prefix Digits (Outgoing Calls): +

Called Party Number Type*: National

Called Party Numbering Plan*: ISDN

- Save

Delete

Copy

Add New

There's one other issue we need to address. The \+! Route Pattern references a Route List of **SLRG**. This Route List uses the Route Group defined by the **Local Route Group** parameter in the calling device's Device Pool. In this case, the calling party is the SIP trunk, which belongs to the **Device Pool** of **CUBE**. However, we've not yet specified a **Local Route Group** for the **CUBE Device Pool**. Let's do that now.

System > Device Pool > CUBE

Local Route Group: **BR1** (NOTE: This Route Group contains the BR1 gateway.)

Save







Reset

Reset


Close

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Device Pool Configuration

 Save
  Delete
  Copy
  Reset
  Apply Config
  Add New


Status

 Status: Ready

Device Pool Information

Device Pool: CUBE (1 members**)

Device Pool Settings

Device Pool Name* CUBE 

Cisco Unified Communications Manager Group* BR1 ▾

Calling Search Space for Auto-registration < None > ▾

Adjunct CSS < None > ▾

Reverted Call Focus Priority Default ▾

Intercompany Media Services Enrolled Group < None > ▾

Roaming Sensitive Settings

Date/Time Group* BR1 ▾

Region* CUBE ▾

Media Resource Group List CUBE ▾

Location CUBE ▾

Network Locale < None > ▾

SRST Reference* Disable ▾


Connection Monitor Duration***

Single Button Barge* Default ▾

Join Across Lines* Default ▾

Physical Location < None > ▾

Device Mobility Group < None > ▾

Local Route Group BR1  ▾

Verification

Like the other CUCM call routing tasks, let's take the time to verify our plus dialing configuration is behaving as expected. First, from the **Missed Calls** directory on **HQ Phone 2**, we select the fully-globalized number of **+14082222020**, and press the **Dial** softkey. We can monitor the ANI and the DNIS on the HQ gateway by issuing the **debug isdn q931** command.

```
HQ#debug isdn q931
debug isdn q931 is ON.
HQ#
Aug 5 18:52:54.532: ISDN Se0/0/0:23 Q931: Applying typeplan for sw-type 0xD
is 0x4 0x1, Calling num 2222002
Aug 5 18:52:54.532: ISDN Se0/0/0:23 Q931: Sending SETUP callref = 0x0082
callID = 0x8003 switch = primary-ni interface = User
Aug 5 18:52:54.532: ISDN Se0/0/0:23 Q931: TX -> SETUP pd = 8 callref =
0x0082
    Bearer Capability i = 0x8090A2
        Standard = CCITT
        Transfer Capability = Speech
        Transfer Mode = Circuit
        Transfer Rate = 64 kbit/s
    Channel ID i = 0xA98383
```

```

        Exclusive,
HQ#Channel 3
    Display i = 'HQ Phone 2'
    Calling Party Number i = 0x4181, '2222002'
        Plan:ISDN, Type:Subscriber(local)
    Called Party Number i = 0xC1, '2222020'
        Plan:ISDN, Type:Subscriber(local)
Aug  5 18:52:54.552: ISDN Se0/0/0:23 Q931: RX <- CALL_PROC pd = 8  callref =
0x8082
    Channel ID i = 0xA98383
        Exclusive, Channel 3
Aug  5 18:52:54.552: ISDN Se0/0/0:23 Q931: RX <- ALERTING pd = 8  callref =
0x8082
    Progress Ind i = 0x8188 - In-band info or appropriate now available
HQ#
Aug  5 18:52:57.724: ISDN Se0/0/0:23 Q931: TX -> DISCONNECT pd = 8  callref =
0x0082
    Cause i = 0x8090 - Normal call clearing
Aug  5 18:52:57.732: ISDN Se0/0/0:23 Q931: RX <- RELEASE pd = 8  callref =
0x8082
Aug  5 18:52:57.732: ISDN Se0/0/0:23 Q931: TX -> RELEASE_COMP pd = 8  callref
= 0x0082
HQ#

```

To test the redundancy component of our configuration, we **temporarily swap the Route Groups** in the newly created Route List.

Call Routing > Route/Hunt > Route List > Find > HQ-TO-LOCAL

Selected Groups: **Standard Local Route Group** (Click down arrow on the right to move this Route Group to the bottom of the list.)

Save







Reset

Reset


Close

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Route List Configuration

 Save
  Delete
  Copy
  Reset
  Apply Config
  Add New

Status

 Status: Ready

Route List Information

Registration: Registered with Cisco Unified Communications Manager 192.168.1.72

IP Address: 192.168.1.72

☒ Device is trusted

Name*: HQ-TO-LOCAL

Description:

Cisco Unified Communications Manager Group*: SUB-PUB

☒ Enable this Route List (change effective on Save; no reset required)


☐ Run On All Active Unified CM Nodes

Route List Member Information



Selected Groups**

- Standard Local Route Group
- SIP-TRUNK-TO-CUBE

Removed Groups***



Route List Details

-  Standard Local Route Group
-  SIP-TRUNK-TO-CUBE

Let's place the call again (from HQ Phone 2's Missed Calls Directory) and observe what DNIS router BR1 sends to the PSTN.

```
BR1#debug isdn q931
debug isdn q931 is ON.
BR1#
Aug 6 15:51:37.482: ISDN Se0/0/0:23 Q931: TX -> SETUP pd = 8 callref = 0x0007
    Bearer Capability i = 0x8090A2
        Standard = CCITT
        Transfer Capability = Speech
        Transfer Mode = Circuit
        Transfer Rate = 64 kbit/s
    Channel ID i = 0xA98383
        Exclusive, Channel 3
    Display i = 'HQ Phone 2'
    Calling Party Number i = 0x2183, '4082222002'
        Plan:ISDN, Type:National
    Called Party Number i = 0xA1, '4082222020'
        Plan:ISDN, Type:National
Aug 6 15:51:37.502: ISDN Se0/0/0:23 Q931: RX <-
BR1# CALL_PROC pd = 8 callref = 0x8007
    Channel ID i = 0xA98383
        Exclusive, Channel 3
Aug 6 15:51:37.506: ISDN Se0/0/0:23 Q931: RX <- ALERTING pd = 8 callref = 0x8007
    Progress Ind i = 0x8188 - In-band info or appropriate now available
Aug 6 15:51:38.210: ISDN Se0/0/0:23 Q931: TX -> DISCONNECT pd = 8 callref = 0x0007
    Cause i = 0x8090 - Normal call clearing
```

```

Aug  6 15:51:38.214: ISDN Se0/0/0:23 Q931: RX <- RELEASE pd = 8  callref = 0x8007
Aug  6 15:51:38.242: ISDN Se0/0/0:23 Q931: TX -> RELEA
BR1#SE_COMP pd = 8  callref = 0x0007
BR1#

```

The ANI, DNIS, and TON information has been verified, using both the HQ gateway and the BR1 gateway. Now, we need to put the Route Groups in the HQ-TO-LOCAL Route List back in the correct order.

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Route List Configuration

Save Delete Copy Reset Apply Config Add New

Status
Update successful

Route List Information
 Registration: Registered with Cisco Unified Communications Manager 192.168.1.72
 IP Address: 192.168.1.72
☒ Device is trusted
 Name*: HQ-TO-LOCAL
 Description:
 Cisco Unified Communications Manager Group*: SUB-PUB
☒ Enable this Route List (change effective on Save; no reset required)
☐ Run On All Active Unified CM Nodes

Route List Member Information
 Selected Groups**: SIP-TRUNK-TO-CUBE, Standard Local Route Group
 Removed Groups***:

Route List Details
☒ SIP-TRUNK-TO-CUBE
☒ Standard Local Route Group

Save Delete Copy Reset Apply Config Add New

Task 4.5 URI Dialing

Assign a URI to the DN on HQ Phone 1

Device > Phone > Find > 9971 > Line[1] – 2001 in INTERNAL

URI: **hquser1@cisco.local**

Partition: **INTERNAL**

Save

Reset

Reset

Close

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Directory Number Configuration Related Links: [Configure Device \(SEP0CD996912474\)](#) Go

Save Delete Reset Apply Config Add New

Directory URIs

Primary	URI	Partition	Remove
<input checked="" type="radio"/>	hquser1@cisco.local	INTERNAL	

Add Row

Add the Speed Dial to HQ Phone 2

Device > Phone > Find > 7965 > Add a new SD

Number: **hquser1@cisco.local**

Label: **HQ Phone 1**

ASCII Label: **HQ Phone 1**

Save

Close

Speed Dial Configuration for SEPECC8821098E9

Save Close Help

Status

Status: Ready

Speed Dial (Button) Settings

Number	Label	ASCII Label
1		
2		
3		
4	hquser1@cisco.local	HQ Phone 1

Task 5.1 Video Conference Bridge

We already configured router HQ to act as a video conferencing bridge. However, we now need to configure the HQ CUCM cluster to register that DSP resource.

Define a Conference Bridge on the HQ CUCM cluster

Media Resources > Conference Bridge > Add New

Conference Bridge Type: **Cisco IOS Homogeneous Video Conference Bridge**

Conference Bridge Name: **HQ-VID-CFB**


Device Pool: **HQ**


Location: **Hub_None**

Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Conference Bridge Configuration

 Save

Status
 Status: Ready

Conference Bridge Information
 Conference Bridge : New

IOS Conference Bridge Info

Conference Bridge Type* Cisco IOS Homogeneous Video Conference Bridge ←

☒ Device is trusted

Conference Bridge Name* HQ-VID-CFB ←


Description

Device Pool* HQ ←

Common Device Configuration < None > ←

Location* Hub_None ←

Use Trusted Relay Point* Default ←

 Save







The video conference bridge shown now be registered.


Conference Bridge Configuration

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Conference Bridge Configuration

Related Links:

 Save  Delete  Copy  Reset  Apply Config  Add New

Status
 Status: Ready

Conference Bridge Information

Conference Bridge : HQ-VID-CFB (HQ-VID-CFB)

Registration Registered with Cisco Unified Communications Manager 192.168.1.72

IP Address 10.10.120.1

Module 13: Doing EVERYTHING on Cisco Unity Connection (CUC)

Task 7.1 Cisco Unity Connection (CUC) SCCP Integration Configuration

Associate Users with IP Phone and DN

In preparation for the CUCM-CUC integration, we should associate each HQ user with an IP phone and a DN on that IP phone.

User Management > End User > Find > hqphone1 > Device Association

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

End User Configuration

Save ☒ Delete ☒ Add New

Device Information

Controlled Devices

Available Profiles

CTI Controlled Device Profiles

Device Association
Line Appearance Association for Presence

Find > (Check box next to HQ Phone 1) > Save Selected/Changes > Go (Next to Related Links)

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

User Device Association

Related Links: Back to User

Select All ☐ Clear All ☐ Select All In Search ☐ Clear All In Search ☐ Save Selected/Changes Remove All Associated Devices ☒

User Device Association (1 - 2 of 2) Rows per Page 50

Find User Device Association where (Name) (begins with) Find Clear Filter

☒ Show the devices already associated with user

	Device Name	Directory Number	Description
<input checked="" type="checkbox"/>	SEP0CD996912474	2001	HQ Phone 1
<input type="checkbox"/>	SEP0CC8821098E9	2002	HQ Phone 2

Select All ☐ Clear All ☐ Select All In Search ☐ Clear All In Search ☐ Save Selected/Changes Remove All Associated Devices ☒

Primary Extension > 2001 > Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾

End User Configuration

Save ☒ Delete ☒ Add New

☒ Allow Control of Device from CTI
☐ Enable Extension Mobility Cross Cluster

Directory Number Associations

Primary Extension

Mobility Information

☐ Enable Mobility
☐ Enable Mobile Voice Access

Maximum Wait Time for Desk Pickup* 10000

Remote Destination Limit* 4

Remote Destination Profiles

[View Details](#)

Repeat for hqphone2 user.

Run the Cisco Voice Mail Port Wizard

Advanced Features > Voice Mail > Cisco Voice Mail Port Wizard

You're asked for the name that will act as the prefix to voice mail port names. This, by the way, has to match on the CUC server. We can leave it at the default, and click **Next**.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration 1 Go
administrator Search Documentation About Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Steps

- Add a new voice mail server
- Add ports**
- Configure device information for ports
- Configure Directory Numbers
- Configure Line Group
- Confirmation
- Summary

Cisco Voice Mail Server

Add ports to a new Cisco Voice Mail Server using this name:

Back < **Next >** Cancel

We'll now say that we want to add **2** ports, based on the task indicating we need to configure two voice mail port DNS. Then, click **Next**.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration 1 Go
administrator Search Documentation About Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Steps

- Add a new voice mail server
- Add ports**
- Configure device information for ports
- Configure Directory Numbers
- Configure Line Group
- Confirmation
- Summary

Cisco Voice Mail Ports

CiscoUM1 currently has 0 ports configured.

VM Port names have the format <servername>-<n> where n is the port number. The number of ports which can be configured for a VM server varies from 1 to 250 depending on the name of the server. If you wish to configure more ports, please reduce the length of the server name.

How many ports do you want to add:

Back < **Next >** Cancel

Description: **Voice Mail Port**
 Device Pool: **HQ**
 Calling Search Space: **HQ**
 AAR Calling Search Space: **HQ**
 Location: **Hub_None**
 Device Security Mode: **Non Secure Voice Mail Port**
Next

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help

Steps:

- Add a new voice mail server
- Add ports
- Configure device information for ports
- Configure Directory Numbers
- Configure Line Group
- Confirmation
- Summary

Cisco Voice Mail Device Information

Enter the device information for ports. A Device Pool selection is required. The Wizard applies these settings to all new ports.

Device Information

Description: Voice Mail Port

Device Pool: HQ

Calling Search Space: HQ

AAR Calling Search Space: HQ

Location: Hub_None

Device Security Mode: Non Secure Voice Mail Port

Use Trusted Relay Port: Default

Back Next Cancel

We now enter the beginning voice mail port DN (NOT the voicemail pilot number).

Beginning Directory Number: **2501**

Partition: **INTERNAL**

Calling Search Space: **HQ**

External Number Mask: **+1408222XXXX**

Next

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help

Steps:

- Add a new voice mail server
- Add ports
- Configure device information for ports
- Configure Directory Numbers
- Configure Line Group
- Confirmation
- Summary

Cisco Voice Mail Directory Numbers

Enter the directory number settings for the new Cisco Voice Mail Server if a Partition is selected, you must select a Calling Search Space that includes the selected Partition.

Beginning Directory Number: 2501 (each new port receives the next available directory number)

Partition: INTERNAL

Calling Search Space: HQ

AAR Group: < None >

Internal Caller ID Display: VoiceMail

Internal Caller ID Display (ASCII format): VoiceMail

External Number Mask: +1408222XXXX

Back Next Cancel

Select the option **Yes. Add directory numbers to a new Line Group.**

Next

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Steps:
 Add a new voice mail server
 Add ports
 Configure device information for ports
 Configure Directory Numbers
Configure Line Group
 Confirmation
 Summary

Do you want to add these directory numbers to a Line Group?
 For using these ports, you need to add corresponding directory numbers to a line group. You can add them to an existing line group or to a new line group. If you decide to add it later, you can do so by using Line Group configuration page.

☒ Yes: Add directory numbers to a new Line Group
☐ Yes: Add directory numbers to an existing Line Group
☐ No: I will add them later.

Back < **Next >** Cancel

Accept the default Line Group name of **CiscoUM1**.
Next

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Steps:
 Add a new voice mail server
 Add ports
 Configure device information for ports
 Configure Directory Numbers
Configure Line Group
 Confirmation
 Summary

Line Group:
 Enter the Line Group settings for Cisco Voice Mail Server

Line Group Name: CiscoUM1

Back < **Next >** Cancel

We now get a screen of summary information. If everything looks good, click **Finish**.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Steps

- Add a new voice mail server
- Add ports
- Configure device information for ports
- Configure Directory Numbers
- Configure Line Group
- Optimization**
- Summary

Ready to Add Cisco Voice Mail Ports

The information shown below will be applied to the Cisco Voice Mail Ports being created. If this information is correct, click Finish to add the new ports. If the information shown is not correct, click the Back button to edit the information, or Cancel to quit without adding any ports.

Cisco Voice Mail Device Information (apply to all ports)

Number of Ports to Add	2 (adding ports 1 - 2)
Cisco Voice Mail Server Name	CiscoUM1
Description	Voice Mail Port
Device Pool	HQ
Calling Search Space	HQ
AAR Calling Search Space	HQ
Location	Hub_None
Device Security Mode	Non Secure Voice Mail Port
Use Trusted Relay Point	Default

Directory Number Information

New Directory Numbers	2501 - 2502
Partition	INTERNAL
Calling Search Space	HQ
AAR Group	< None >
Internal Caller ID Display	VoiceMail
Internal Caller ID Display (ASCII format)	VoiceMail
External Number Mask	+1408222XXXX
Line Group	CiscoUM1

Back < **Finish** > Cancel

We're then reminded that we need to add the new Line Group to a Hunt List, and that Hunt List needs to be assigned to a Hunt Pilot.

Create a Hunt List

Call Routing > Route/Hunt > Hunt List > Add New

Name: **CiscoVM**

Cisco Unified Communications Manager Group: **SUB-PUB**

Enable this Hunt List: *(Checked)*

For Voice Mail Usage: *(Checked)*

Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Hunt List Configuration

Save

Status

Status: Ready

Hunt List Information

☒ Device is trusted

Name*

Description

Cisco Unified Communications Manager Group*

☒ Enable this Hunt List (change effective on Save; no reset required)







☒ For Voice Mail Usage

Save


Add Line Group

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Hunt List Configuration

 Save  Delete  Copy  Reset  Apply Config  Add New

Status

 Add successful

Hunt List Information

☒ Device is trusted

Name*

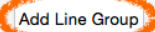
Description

Cisco Unified Communications Manager Group*

☒ Enable this Hunt List (change effective on Save; no reset required)

☒ For Voice Mail Usage

Hunt List Member Information

 Add Line Group

Selected Groups**

Removed Groups***


Line Group: **CiscoUM1**

Save


OK

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾

Hunt List Detail Configuration


 Save

Status

 Status: Ready

Hunt List Member Information

Line Group*

 Save

Save

OK

Reset
Reset
Close

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾

Hunt List Configuration

Save Delete Copy Reset Apply Config Add New

Status
 Add successful

Hunt List Information

☒ Device is trusted
Name* CiscoVM
Description
Cisco Unified Communications Manager Group* SUB-PUB
☒ Enable this Hunt List (change effective on Save; no reset required)
☒ For Voice Mail Usage

Hunt List Member Information

Add Line Group
Selected Groups** CiscoUM1
Removed Groups***

Hunt List Details
 CiscoUM1

Save Delete Copy Reset Apply Config Add New

Now, it's time to point a Hunt Pilot to the Hunt List.

[Create a Hunt Pilot](#)

Call Routing > Route/Hunt > Hunt Pilot > Add New

Hunt Pilot: **2500**

Route Partition: **INTERNAL**

Hunt List: **CiscoVM**

Provide Outside Dial Tone: *(unchecked)*

Save

[Create a MWI DNs](#)

Next, let's create a couple of message waiting indicator (MWI) DNs. First, let's create the MWI ON DN.

Advanced Features > Voice Mail > Message Waiting > Add New

Message Waiting Number: **2510**

Partition: **INTERNAL**


Message Waiting Indicator: **On**

Calling Search Space: **HQ**

Save

System > Call Routing > Media Resources > Advanced Features > Device > Application

Message Waiting Configuration

 Save

Message Waiting Information

Message Waiting Number* 2510

Partition INTERNAL

Description

Message Waiting Indicator* ☒ On ☐ Off

Calling Search Space HQ

Save

Now, let's create the MWI OFF DN.

Advanced Features > Voice Mail > Message Waiting > Add New

Message Waiting Number: **5211**

Partition: **INTERNAL**


Message Waiting Indicator: **Off**

Calling Search Space: **HQ**


Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾

Message Waiting Configuration

 Save

Status

 Status: Ready

Message Waiting Information


Message Waiting Number* 2511

Partition INTERNAL

Description

Message Waiting Indicator* On ☒ Off ☐

Calling Search Space HQ

 Save

Next, we need to create a voice mail pilot number that points to the Hunt Pilot we just created. Or, we can just modify the existing default voice mail pilot number (which is what we'll do).

Modify the Default Voice Mail Pilot

Advanced Features > Voice Mail > Voice Mail Pilot > Find > Default




Voice Mail Pilot Number: **2500**

Calling Search Space: **HQ**


Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾

Voice Mail Pilot Configuration

 Save  Delete  Add New

Status

 Status: Ready




Voice Mail Pilot Information

Voice Mail Pilot Number 2500

Calling Search Space HQ

Description Default

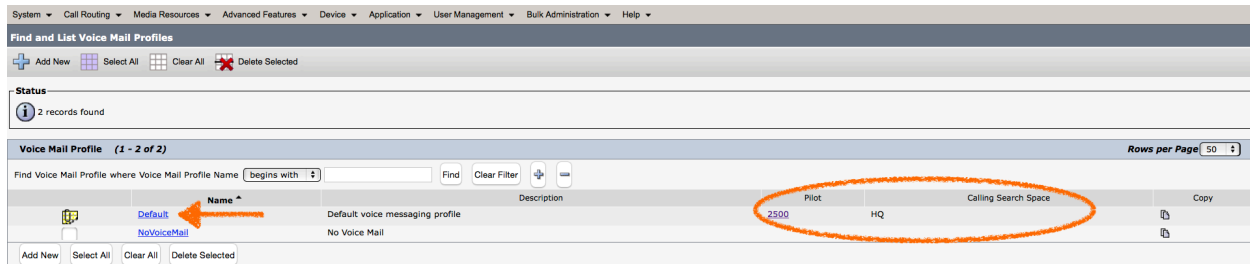
☒ Make this the default Voice Mail Pilot for the system

 Save  Delete  Add New

Finally, on the CUCM server we need to reference the voice mail pilot with a voice mail profile. However, since we modified the default voice mail pilot, our voice mail profile should already reference the appropriate voice mail pilot. However, we should verify that.

Verify the Default Voice Mail Profile

Advanced Features > Voice Mail > Voice Mail Profile > Find

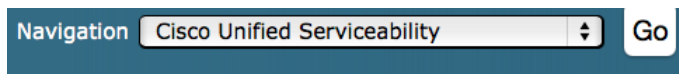


Name	Description	Pilot	Calling Search Space	Copy
Default	Default voice messaging profile	2500	HQ	
NoVoiceMail	No Voice Mail			

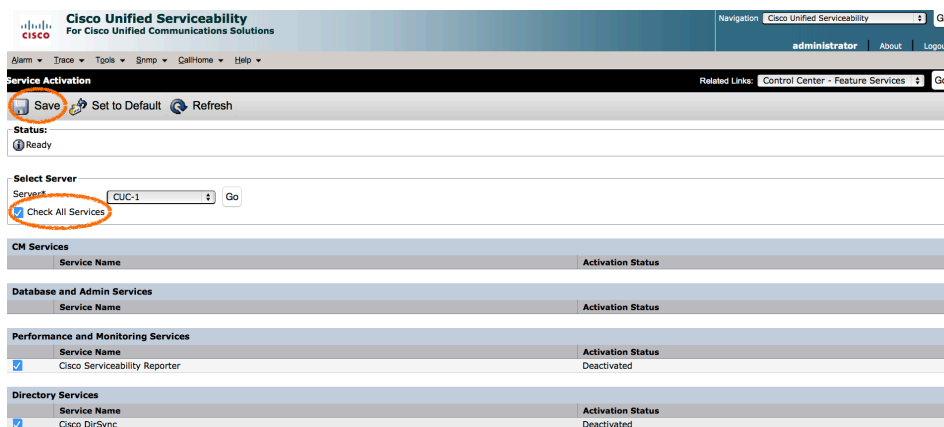
Enable all Services on the HQ CUC Server

Let's now move to the HQ CUC server and enable all services.

Navigation: **Cisco Unified Serviceability**
Go
(login)



Tools > Service Activation > Check All Services (checked) > Save > OK



Service Activation

Save Set to Default Refresh

Status: Ready

Select Server: CUC-1 Go

☒ Check All Services

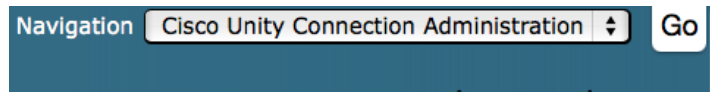
Service Name	Activation Status
CM Services	
Database and Admin Services	
Performance and Monitoring Services	
Directory Services	

Define a Phone System

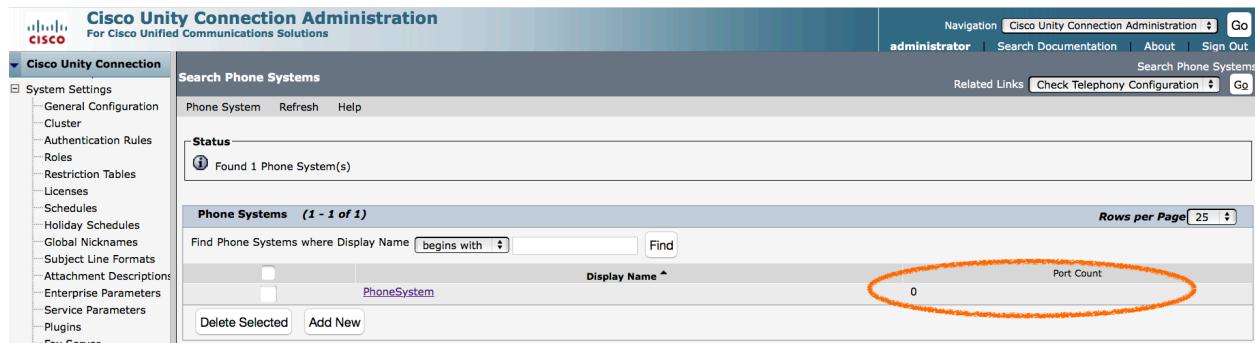
We now navigate back to the Cisco Unity Connection Administrator interface, and configure a Phone System.

Navigation: **Cisco Unity Connection Administration**

Go
(login)

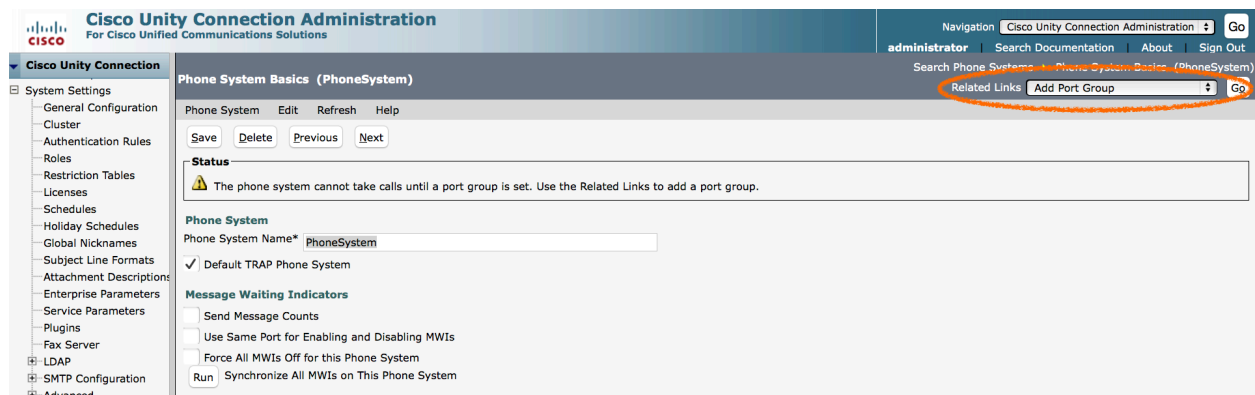


Telephony Integrations > Phone System



Currently, this phone system doesn't have any ports. However, we don't add ports directly to a phone system. Instead, we add a port group to the phone system, and then we can add ports to that port group.

PhoneSystem > Add Port Group (in Related Links menu) > Go



Port Group Type: **SCCP**

Device Name Prefix: **CiscoUM1-VI** (NOTE: This must match the naming of the voice mail ports in CUCM. So, you might want to copy/paste this string.)

MWI On Extension: **2510**

MWI Off Extension: **2511**

IPv4 Address or Host Name: **192.168.1.71** (NOTE: This is the IP address of the Publisher server.)

Save

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Cisco Unity Connection

- System Settings
 - General Configuration
 - Cluster
 - Authentication Rules
 - Roles
 - Restriction Tables
 - Licenses
 - Schedules
 - Holiday Schedules
 - Global Nicknames
 - Subject Line Formats
 - Attachment Descriptions
 - Enterprise Parameters
 - Service Parameters
 - Plugins
 - Fax Server
- LDAP
- SMTP Configuration
- Advanced
- Telephony Integrations
 - Phone System**
 - Port Group**
 - Port
 - Speech Connect Port
 - Trunk
- Security
- Tools

New Port Group

Port Group Reset Help

Save

New Port Group

Phone System PhoneSystem

Create From ☒ Port Group Type SCCP ☐ Port Group

Port Group Description

Display Name* PhoneSystem-1

Device Name Prefix* CiscoUM1-VI

MWI On Extension 2510

MWI Off Extension 2511

Primary Server Settings

IPv4 Address or Host Name 192.168.1.71

IPv6 Address or Host Name

Port 2000

TLS Port 2443

Save

Now that we have a Port Group created, we can add ports.

Add Ports (in *Related Links* menu) > Go

Navigation Cisco Unity Connection Administration Go

administrator | Search Documentation | About | Sign Out

Search Port Groups Port Group Basics (PhoneSystem-1)

Related Links Add Ports Go

Number of Ports: 2

Save

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Cisco Unity Connection

- System Settings
 - General Configuration
 - Cluster
 - Authentication Rules
 - Roles
 - Restriction Tables
 - Licenses
 - Schedules
 - Holiday Schedules
 - Global Nicknames
 - Subject Line Formats
 - Attachment Descriptions
 - Enterprise Parameters
 - Service Parameters
 - Plugins
 - Fax Server
- LDAP
- SMTP Configuration
- Advanced
- Telephony Integrations
 - Phone System**
 - Port Group
 - Port
 - Speech Connect Port
 - Trunk
- Security

New Port

Port Reset Help

Save

New Phone System Port

☒ Enabled

Number of Ports **2**

Phone System PhoneSystem

Port Group PhoneSystem-1

Server cuc-1

Port Behavior

☒ Answer Calls

☒ Perform Message Notification

☒ Send MWI Requests (may also be disabled by the port group)

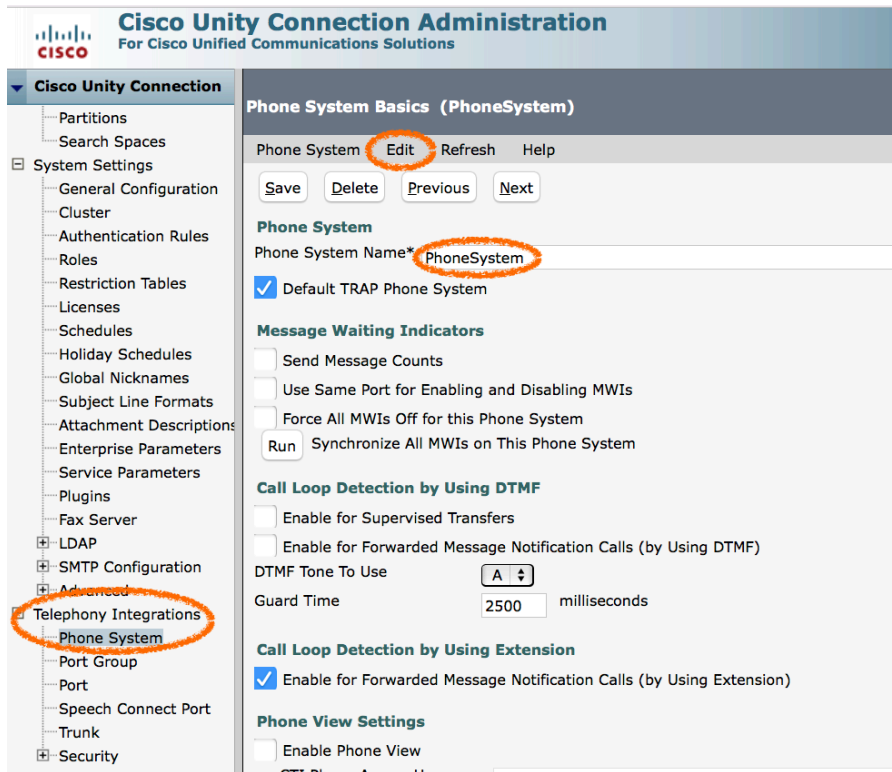
☒ Allow TRAP Connections

Security Mode Non-secure

Save

Next, we want to enable CUC to talk with the CUCM server, perhaps to perform a user import (as one example). While we could create a specific AXL (Administration XML) user that belongs to an AXL administrative group, we'll just use the **administrator** account, since it has access to everything we need. To set this up, let's go back into our Phone System.

Telephony Integrations > Phone System > PhoneSystem

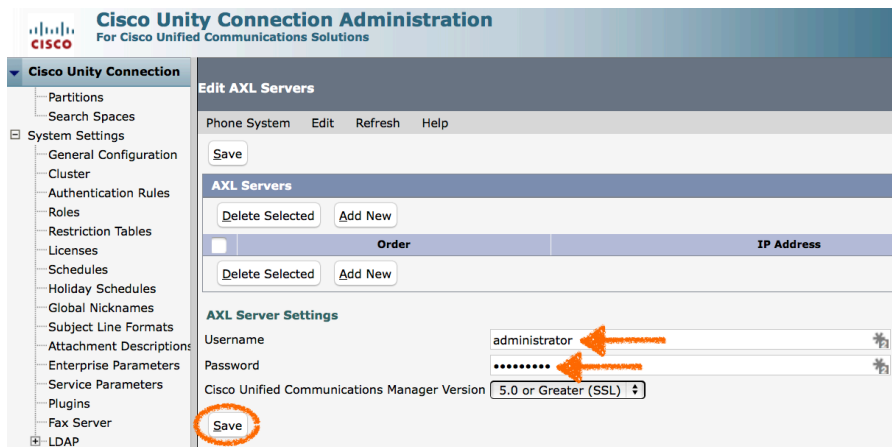


Edit > Cisco Unified Communications Manager AXL Servers

Username: **administrator**

Password: **kwtrain01**

Save



Now, let's define a couple of AXL servers.

Add New

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Cisco Unity Connection

- Partitions
- Search Spaces
- System Settings
 - General Configuration
 - Cluster
 - Authentication Rules
 - Roles
 - Restriction Tables
 - Licenses
 - Schedules
 - Holiday Schedules
 - Global Nicknames
 - Subject Line Formats
 - Attachment Descriptions
 - Enterprise Parameters
 - Service Parameters
 - Plugins
 - Fax Server
- LDAP
- SMTP Configuration
- Advanced

Edit AXL Servers

Phone System Edit Refresh Help

Save

AXL Servers

Delete Selected Add New

	Order	IP Address
<input type="checkbox"/>		

Delete Selected Add New

AXL Server Settings

Username administrator

Password [redacted]

Cisco Unified Communications Manager Version 5.0 or Greater (SSL)

Save

Order: 0

IP Address: 192.168.1.72

Port: 443 (NOTE: This is the HTTPS default port.)

Add New

Order: 1

IP Address: 192.168.1.71

Port: 443

Save

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Cisco Unity Connection

- Partitions
- Search Spaces
- System Settings
 - General Configuration
 - Cluster
 - Authentication Rules
 - Roles
 - Restriction Tables
 - Licenses
 - Schedules
 - Holiday Schedules
 - Global Nicknames
 - Subject Line Formats
 - Attachment Descriptions
 - Enterprise Parameters
 - Service Parameters
 - Plugins
 - Fax Server
- LDAP
- SMTP Configuration
- Advanced
- Telephony Integrations
 - Phone System
 - Port Group
 - Port
 - Speech Connect Port

Edit AXL Servers

Phone System Edit Refresh Help

Save

Status

- Updated Server
- Updated AXL Credentials

AXL Servers

Delete Selected Add New

	Order	IP Address	Port	
<input type="checkbox"/>	0	192.168.1.72	443	Test
<input type="checkbox"/>	1	192.168.1.71	443	Test

Delete Selected Add New

AXL Server Settings

Username administrator

Password [redacted]

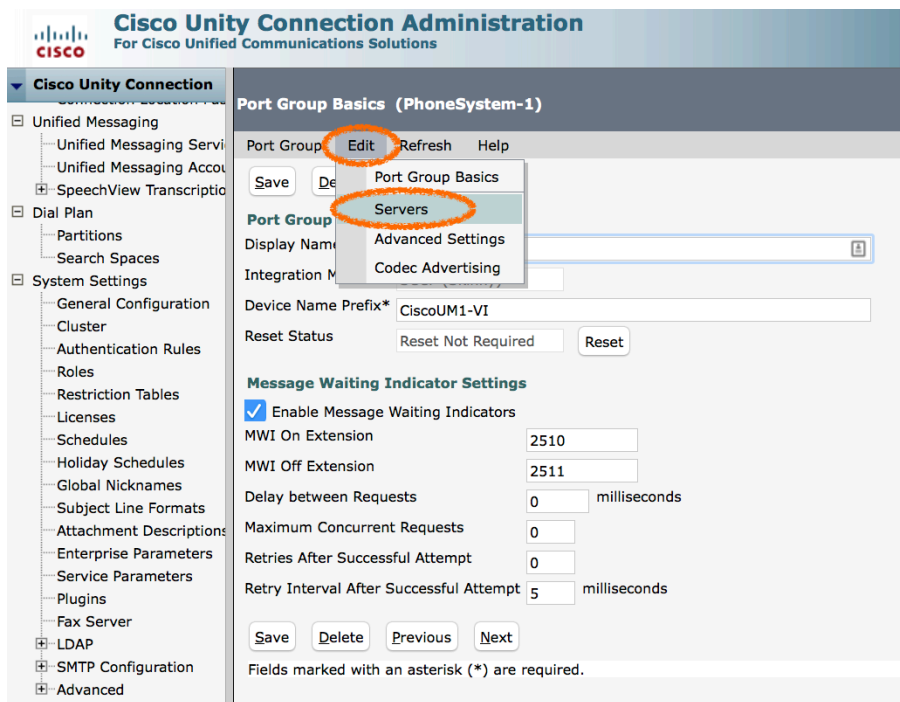
Cisco Unified Communications Manager Version 5.0 or Greater (SSL)

Save

Let's also define server information for our port group.

Telephony Integrations > Port Group > PhoneSystem-1

Edit > Servers



Update the CUCM servers as follows:

Order: **0**, IPv4 Address or Host Name, **192.168.1.72**, Port: **2000** (NOTE: This is the default SCCP port.), TLS Port: **2443** (NOTE: This is the default SCCP port of 2000, plus the default HTTPS port of 443.)

Order: **1**, IPv4 Address or Host Name, **192.168.1.71**, Port: **2000**, TLS Port: **2443**

Update the TFTP servers as follows:

Order: **0**, IPv4 Address or Host Name, **192.168.1.72**

Order: **1**, IPv4 Address or Host Name, **192.168.1.71**

Save

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration Go

administrator Search Documentation About Sign Out
Search Port Groups Port Group Basics (PhoneSystem-1) Edit Servers
Related Links Check Telephony Configuration Go

Cisco Unity Connection

Unified Messaging
Unified Messaging Servi
Unified Messaging Accou
SpeechView Transcription
Dial Plan
Partitions
Search Spaces
System Settings
General Configuration
Cluster
Authentication Rules
Roles
Restriction Tables
Licenses
Schedules
Holiday Schedules
Global Nicknames
Subject Line Formats
Attachment Description
Enterprise Parameters
Service Parameters
Plugins
Fax Server
LDAP
SMTP Configuration
Advanced
Telephony Integrations
Phone System
Port Group
Port
Speech Connect Port
Trunk
Security
Tools
Task Management

Edit Servers
Port Group Edit Refresh Help

Status
One or more port groups need to be reset.
Created Server(s)

Save

Cisco Unified Communications Manager Servers
Delete Selected Add

Order	IPv4 Address or Host Name	IPv6 Address or Host Name	Port	TLS Port	Server Type
0	192.168.1.72		2000	2443	Cisco Unified Communications Manager
1	192.168.1.71		2000	2443	Cisco Unified Communications Manager

Delete Selected Add

☒ Reconnect to a Higher-order Cisco Unified Communications Manager When Available

TFTP Servers
Delete Selected Add

Order	IPv4 Address or Host Name	IPv6 Address or Host Name
0	192.168.1.72	
1	192.168.1.71	

Delete Selected Add

IPv6 Addressing Mode
Preference for Signaling IPv4

Save

To make all of this take effect, we need to reset our port group.

Telephony Integrations > Port Group > PhoneSystem-1 > Reset

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Cisco Unity Connection

Unified Messaging
Unified Messaging Servi
Unified Messaging Accou
SpeechView Transcription
Dial Plan
Partitions
Search Spaces
System Settings
General Configuration
Cluster
Authentication Rules
Roles
Restriction Tables
Licenses
Schedules

Port Group Basics (PhoneSystem-1)
Port Group Edit Refresh Help

Save Delete Previous Next

Status
One or more port groups need to be reset.

Port Group
Display Name* PhoneSystem-1
Integration Method SCCP (Skinny)
Device Name Prefix* CiscoUM1-VI
Reset Status Reset Required

Reset

Verification: Let's take a few seconds to make sure our HQ IP phones get to our CUC server when we press the **Messages** button on either IP phone.

Add Users to the CUC Server

At this point, we have CUC integrated with CUCM, via SCCP. However, we don't have any users or user mailboxes configured. Let's do that next.

Before creating a user, we might want to create a user template that contains a collection of settings that can be applied to new users, without individually configuring all of the new users with those settings.

Also, we might want to create an authentication rule, which can dictate a user's password policy. In or case, Task 7.1 specifies a trivial PIN of 12345. That won't work with the default authentication rule. So, let's update it.

System Settings > Authentication Rules > Recommended Voice Mail Authentication Rule

Minimum Credential Length: 5

Check for Trivial Passwords: (*uncheck*)

Save

The screenshot shows the Cisco Unity Connection Administration interface. The left sidebar lists various configuration areas, with 'Authentication Rules' selected. The main panel displays the 'Edit Authentication Rule' page for the 'Recommended Voice Mail Authentication Rule'. The 'Minimum Credential Length' is set to 5, and the 'Check for Trivial Passwords' checkbox is unchecked. Arrows point to the 'Save' button and the 'Check for Trivial Passwords' checkbox.

Next, let's modify the default voice mail user template.

Templates > User Templates > voicemailusertemplate

The screenshot shows the Cisco Unity Connection Administration interface. The left sidebar lists various configuration areas, with 'User Templates' selected. The main panel displays the 'User Templates' page, showing a list of templates. The 'voicemailusertemplate' is selected, and an arrow points to it. The 'administratortemplate' is also visible in the list.

Since Task 7.1 makes no mention of us having to record our name, we can say that we don't want our users to self-enroll.

Set for Self-enrollment at Next Sign-In: (*unchecked*)

The screenshot shows the 'Edit User Template Basics' page for a voicemail user template. The left sidebar contains a navigation tree with categories like Users, Class of Service, Templates, and Contacts. The 'User Templates' section is expanded, and 'User Templates' is selected. The main content area has tabs for 'User Template', 'Edit', 'Refresh', and 'Help'. The 'Save' button is circled in orange. The 'Name' section includes fields for 'Alias*' (voicemailusertemplate), 'Display Name*' (Voice Mail User Template), and 'Display Name Generation' (First Name, Then Last Name). The 'Phone' section includes fields for 'Outgoing Fax Server', 'Partition', 'Search Scope', 'Phone System', 'Class of Service', and 'Active Schedule'. The 'Set for Self-enrollment at Next Sign-In' checkbox is unchecked. Other checkboxes include 'List in Directory' (checked), 'Send Non-Delivery Receipts on Failed Message Delivery' (checked), 'Skip PIN When Calling From a Known Extension' (unchecked), and 'Use Short Calendar Caching Poll Interval' (unchecked).

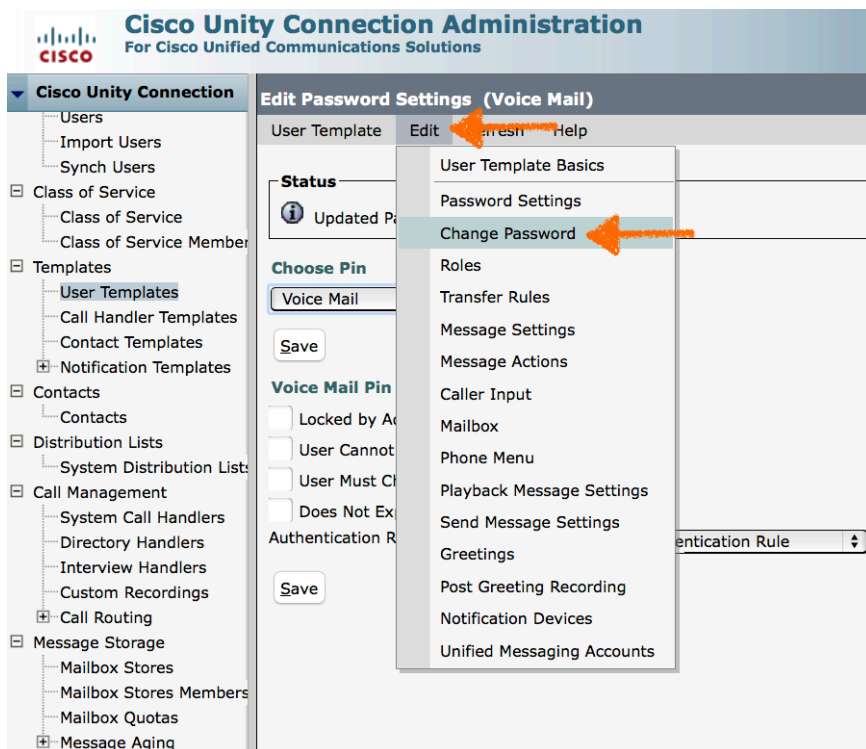
Next, let's update the password settings associated with this default user template.

Edit > Password Settings > User Must Change at Next Sign-In (*unchecked*) > Save

The screenshot shows the 'Edit Password Settings' page for a voice mail user template. The left sidebar is the same as the previous screenshot. The main content area has tabs for 'User Template', 'Edit', 'Refresh', and 'Help'. The 'Choose Pin' dropdown is set to 'Voice Mail'. The 'Save' button is circled in orange. The 'Voice Mail Pin Settings' section includes checkboxes for 'Locked by Administrator' (unchecked), 'User Cannot Change' (unchecked), 'User Must Change at Next Sign-In' (unchecked), and 'Does Not Expire' (unchecked). The 'Authentication Rule' dropdown is set to 'Recommended Voice Mail Authentication Rule'. A 'Save' button is at the bottom.

Note that we're using the default authentication rule that we modified a few moments ago. So, we can create a password that meets the criteria of the authentication rule.

Edit > Change Password



Pin: **12345**
 Confirm Pin: **12345**
Save



Next, let's import users from CUCM into CUC.

Users > Import Users > Find End Users In (Phone System) > Find
Check the checkboxes next to hqphone1 and hqphone2.

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration: 3 Go
administrator | Search Documentation | About | Sign Out

Cisco Unity Connection

- Users
 - Import Users
 - Synch Users
- Class of Service
 - Class of Service Member
- Templates
 - User Templates
 - Call Handler Templates
 - Contact Templates
 - Notification Templates
- Contacts
 - Contacts
- Distribution Lists
 - System Distribution List
- Call Management
 - System Call Handlers
 - Directory Handlers
 - Interview Handlers
 - Custom Recordings
- Call Routing

Import Users
Import Users Refresh Help

Status
Found 2 Unified Communications Manager User(s)

Find
Find End Users In PhoneSystem
Where Alias Begins With Find

Import With
Based on Template voicemailusertemplate

Directory Search Results
Import Selected Import All 25 Rows Per Page

	Alias	First Name	Last Name	Extension
<input checked="" type="checkbox"/>	hqphone1	HQ	Phone 1	2001
<input checked="" type="checkbox"/>	hqphone2	HQ	Phone 2	2002

Import Selected Import All

We should receive a message indicating we had two successful user imports.

Import Users
Import Users Refresh Help

Status
Importing users completed. Number of successes: 2 Number of failures: 0

Find
Find End Users In PhoneSystem
Where Alias Begins With Find

Verification: Let's quickly verify that by pressing the Messages button on HQ Phone 1 and on HQ Phone 2, we can log into (using the PIN of 12345) the mailbox associated with each IP phone's user.

Task 7.2 Cisco Unity Connection (CUC) SIP Integration Configuration

Associate Users with IP Phone and DN

Just as we did on the HQ CUC server, we should associate each BR1 user with an IP phone and a DN on that IP phone.

User Management > End User > Find > br1phone1 > Device Association

Find > (Check box next to BR1 Phone 1) > Save Selected/Changes > Go (Next to Related Links)

Device Name	Directory Number	Description
SEP0024C40D8CFC	3002	BR1 Phone 2
SEP0CD996919A55	3001	BR1 Phone 1

Primary Extension > 3001 > Save

Repeat for br1phone2 user.

Configure SIP Trunk Security Profile

Next, we create modify the non secure SIP Trunk Security Profile.

System > Security > SIP Trunk Security Profile > Find > Non Secure SIP Trunk Profile

Check the following boxes (i.e. all boxes beginning with the word “Accept”):

Accept presence subscription

Accept out-of-dialog refer

Accept unsolicited notification

Accept replaces header

Save
Reset
Reset
Close

System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help

SIP Trunk Security Profile Configuration

Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Status: Ready

SIP Trunk Security Profile Information

Name* Non Secure SIP Trunk Profile

Description Non Secure SIP Trunk Profile authenticated by null S

Device Security Mode Non Secure

Incoming Transport Type* TCP+UDP

Outgoing Transport Type TCP

Enable Digest Authentication

Nonce Validity Time (mins)* 600

X.509 Subject Name

Incoming Port* 5060

Enable Application level authorization

Accept presence subscription

Accept out-of-dialog refer**

Accept unsolicited notification

Accept replaces header

Transmit security status

Allow charging header

SIP V.150 Outbound SDP Offer Filtering* Use Default Filter

Save Delete Copy Reset Apply Config Add New

Configure SIP Trunk

Now, let's create a SIP trunk that's going to use our newly modified SIP Trunk Security Profile.

Device > Trunk > Add New

Trunk Type: **SIP Trunk**

Device Protocol: **SIP**

Next

System > Call Routing > Media Resources > Advanced Features > Device > Application

Trunk Configuration

Next

Status: Ready

Trunk Information

Trunk Type* SIP Trunk

Device Protocol* SIP

Trunk Service Type* None(Default)

Next

*- indicates required item.

Device Name: **CUC**


Device Pool: **BR1**


Redirecting Diversion Header Delivery – Inbound: *(checked)*

Redirecting Diversion Header Delivery – Outbound: (*checked*)
Destination Address: **10.10.150.73** (The IP address of the BR1 CUC server)
SIP Trunk Security Profile: **Non Secure SIP Trunk Profile**
SIP Profile: **Standard SIP Profile**
Save
Reset
Reset
Close

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Trunk Configuration

 Save

Status
 Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name *	<input type="text" value="CUC"/>
Description	<input type="text"/>
Device Pool *	<input type="text" value="BR1"/>
Common Device Configuration	<input type="text" value="< None >"/>
Call Classification *	<input type="text" value="Use System Default"/>
Media Resource Group List	<input type="text" value="< None >"/>
Location *	<input type="text" value="Hub_None"/>
AAR Group	<input type="text" value="< None >"/>
Tunneled Protocol *	<input type="text" value="None"/>
QSIG Variant *	<input type="text" value="No Changes"/>
ASN.1 ROSE OID Encoding *	<input type="text" value="No Changes"/>
Packet Capture Mode *	<input type="text" value="None"/>
Packet Capture Duration	<input type="text" value="0"/>

☐ Media Termination Point Required
☒ Retry Video Call as Audio
☐ Path Replacement Support
☐ Transmit UTF-8 for Calling Party Name
☐ Transmit UTF-8 Names in QSIG APDU
☐ Unattended Port
☐ SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will cause the call to fail.

Consider Traffic on This Trunk Secure*

Route Class Signaling Enabled*


Use Trusted Relay Point*

☒ PSTN Access
☐ Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)
E.164 Transformation Profile

Multilevel Precedence and Preemption (MLPP) Information
MLPP Domain

Call Routing Information
☒ Remote-Party-Id
☒ Assented-Identity
Assented-Type*
SIP Privacy*

Inbound Calls
Significant Digits*
Connected Line ID Presentation*
Connected Name Presentation*
Calling Search Space
AAR Calling Search Space
Prefix DN
 Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings		Default Prefix Settings	
Number Type	Prefix	Strip Digits	Calling Search Space
Incoming Number	Default	0	<input type="text" value="< None >"/>

☒ Use Device Pool CSS

Connected Party Settings
Connected Party Transformation CSS
☒ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS: < None > [1]

Use Device Pool Called Party Transformation CSS: [1]

Calling Party Transformation CSS: < None > [1]

Use Device Pool Calling Party Transformation CSS: [1]

Calling Party Selection*: Originator [1]

Calling Line ID Presentation*: Default [1]

Calling Name Presentation*: Default [1]

Calling and Connected Party Info Format*: Deliver DN only in connected party [1]

Redirecting Diversion Header Delivery - Outbound: [1]

Redirecting Party Transformation CSS: < None > [1]

Use Device Pool Redirecting Party Transformation CSS: [1]

Caller Information

Caller ID DN: [1]

Caller Name: [1]

Maintain Original Caller ID DN and Caller Name in Identity Headers

Normalization Script

Normalization Script: < None > [1]

Enable Trace: [1]

1	Parameter Name	Parameter Value

Geolocation Configuration

Geolocation: < None > [1]

Geolocation Filter: < None > [1]

Send Geolocation Information: [1]

Save Delete Reset Add New

Configure Route Pattern

Since we're using a SIP trunk, there's nothing that has to register with the BR1 CUCM (as the HQ CUC voice mail ports did with the SCCP integration). Instead, these servers just point to one another over this SIP trunk. However, we do need a route pattern that directs calls over this trunk that are destined for the BR1 CUC.

Call Routing > Route/Hunt > Route Pattern > Add New

Route Pattern: **3500**

Route Partition: **INTERNAL**

Gateway/Route List: **CUC** (This is the trunk we previously created.)

Provide Outside Dial Tone: *(unchecked)*

Save

System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help

Route Pattern Configuration

Save

Status

Status: Ready

Pattern Definition

Route Pattern*: 3500 [1]

Route Partition: INTERNAL [1]

Description: [1]

Numbering Plan: -- Not Selected -- [1]

Route Filter: < None > [1]

MLPP Precedence*: Default [1]

Apply Call Blocking Percentage: [1]

Resource Priority Namespace Network Domain: < None > [1]

Route Class*: Default [1]

Gateway/Route List*: CUC [1] (Edit)

Route Option: ☒ Route this pattern ☐ Block this pattern No Error [1]

Call Classification*: OnNet [1]

Allow Device Override: ☐ Provide Outside Dial Tone: ☐ Allow Overlap Sending: ☐ Urgent Priority: ☐

Require Forced Authorization Code: ☐

Authorization Level*: 0 [1]

Require Client Matter Code: ☐

Configure Voice Mail Pilot

We now need to say that this number of 3500 is a voice mail pilot number.

Advanced Features > Voice Mail > Voice Mail Pilot > Find > Default

Voice Mail Pilot Number: **3500**

Calling Search Space: **BR1**

Save

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾

Voice Mail Pilot Configuration

Save Delete Add New

Status
 Status: Ready

Voice Mail Pilot Information

Voice Mail Pilot Number

Calling Search Space

Description

☒ Make this the default Voice Mail Pilot for the system

– Save Delete Add New

Verify the Default Voice Mail Profile

Finally, on the CUCM server we need to reference the voice mail pilot with a voice mail profile. However, since we modified the default voice mail pilot, our voice mail profile should already reference the appropriate voice mail pilot. However, we should verify that.

Advanced Features > Voice Mail > Voice Mail Profile > Find

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Find and List Voice Mail Profiles

Add New Select All Clear All Delete Selected

Status
 2 records found

Voice Mail Profile (1 - 2 of 2) Rows per Page: 50

Find Voice Mail Profile where Voice Mail Profile Name begins with Find Clear Filter

	Name ^	Description	Pilot	Calling Search Space	Copy
	Default	Default voice messaging profile	3500	BR1	
	NoVoiceMail	No Voice Mail			

Add New Select All Clear All Delete Selected

Enable all Services on the BR1 CUC Server

Let's now move to the BR1 CUC server and enable all services.

Navigation: **Cisco Unified Serviceability**
Go

(login)

Navigation Go

Tools > Service Activation > Check All Services (*checked*) > Save > OK

Select Server
Server: Go
☒ Check All Services

CM Services	
Service Name	Activation Status
Database and Admin Services	
Service Name	Activation Status
Performance and Monitoring Services	
Service Name	Activation Status
<input checked="" type="checkbox"/> Cisco Serviceability Reporter	Activated
Directory Services	
Service Name	Activation Status
<input checked="" type="checkbox"/> Cisco DirSync	Activated

Set to Default Refresh

Define a Phone System

We now navigate back to the Cisco Unity Connection Administrator interface, and configure a Phone System.

Navigation: **Cisco Unity Connection Administration**

Go

(login)

Navigation Go

Telephony Integrations > Phone System

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Navigation Go
administrator | Search Documentation | About | Sign Out
Search Phone Systems
Related Links Go

Phone System Refresh Help

Status
Found 1 Phone System(s)

Phone Systems (1 - 1 of 1) Rows per Page 25

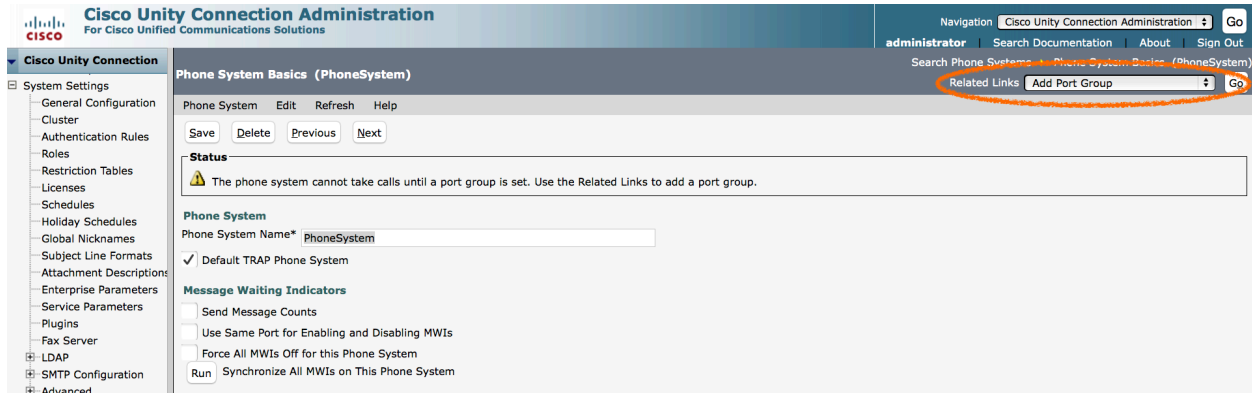
Find Phone Systems where Display Name begins with Find

Display Name	Port Count
PhoneSystem	0

Delete Selected Add New

Currently, this phone system doesn't have any ports. However, we don't add ports directly to a phone system. Instead, we add a port group to the phone system, and then we can add ports to that port group.

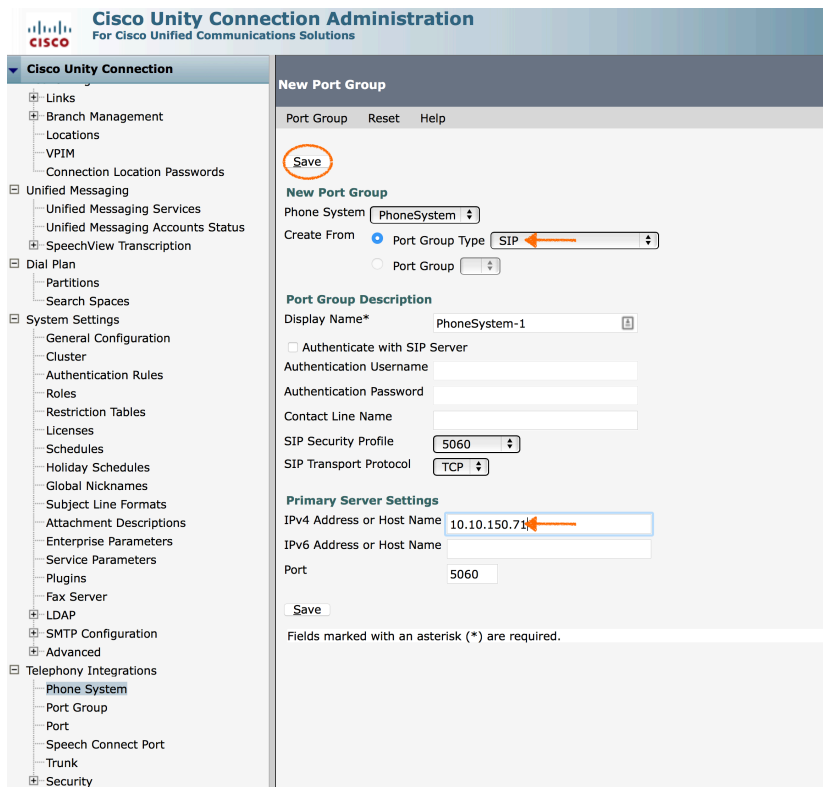
PhoneSystem > Add Port Group (in Related Links menu) > Go



Port Group Type: **SIP**

IPv4 Address or Host Name: **10.10.150.71** (NOTE: This is the IP address of the Publisher server.)

Save



Next, we want to tell the Port Group to register with the SIP server (i.e. the BR1 CUCM Publisher). Then, we'll say that we want to add ports.

Register with SIP Server: *(checked)*

Save

Reset

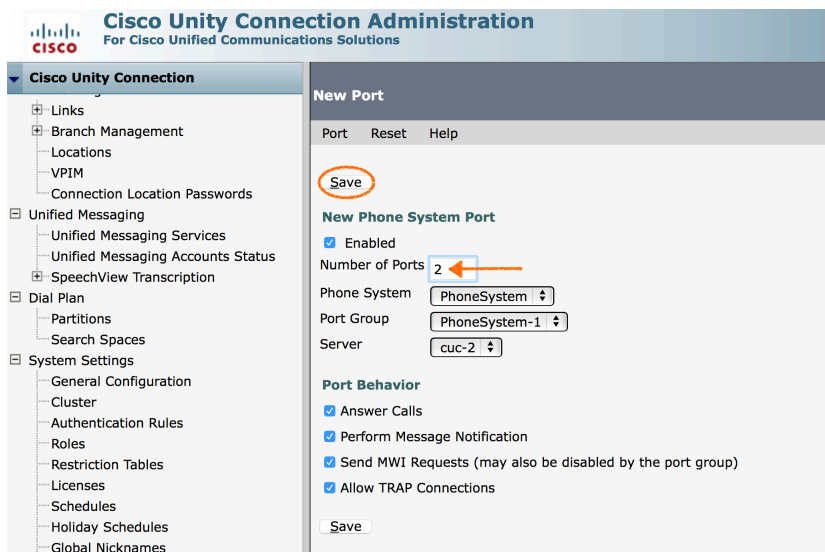
Add Ports *(in Related Links menu)* > **Go**

The screenshot shows the Cisco Unity Connection Administration interface. The left sidebar contains a navigation tree with categories like Links, Branch Management, Locations, VPM, Connection Location Passwords, Unified Messaging, Unified Messaging Services, Unified Messaging Accounts Status, SpeechView Transcription, Dial Plan, Partitions, Search Spaces, System Settings, General Configuration, Cluster, Authentication Rules, Roles, Restriction Tables, Licenses, Schedules, Holiday Schedules, Global Nicknames, Subject Line Formats, Attachment Descriptions, Enterprise Parameters, and Service Parameters. The main content area is titled 'Port Group Basics (PhoneSystem-1)'. It includes a 'Port Group' section with buttons for 'Save', 'Delete', 'Previous', and 'Next'. The 'Save' button is circled in orange. Below this is a 'Status' section with a warning icon and text: 'The phone system cannot take calls if it has no ports. Use the Related Links to add ports.' and 'Created Port Group(s)'. The 'Port Group' section also includes fields for 'Display Name*' (PhoneSystem-1), 'Integration Method' (SIP), and 'Reset Status' (Reset Not Required, with a 'Reset' button circled in orange). The 'Session Initiation Protocol (SIP) Settings' section has a 'Register with SIP Server' checkbox checked, and an 'Authenticate with SIP Server' checkbox unchecked. Below these are fields for 'Authentication Username', 'Authentication Password', and 'Contact Line Name'. The 'SIP Security Profile' is set to '5060' and the 'SIP Transport Protocol' is set to 'TCP'.

The screenshot shows the top navigation bar of the Cisco Unity Connection Administration interface. It includes a 'Navigation' dropdown menu, a 'Go' button, and links for 'administrator', 'Search Documentation', 'About', and 'Sign Out'. Below this is a search bar with the text 'Search Port Groups' and 'Port Group Basics (PhoneSystem-1)'. The 'Related Links' section is circled in orange, and it contains a link labeled 'Add Ports' with a 'Go' button next to it.

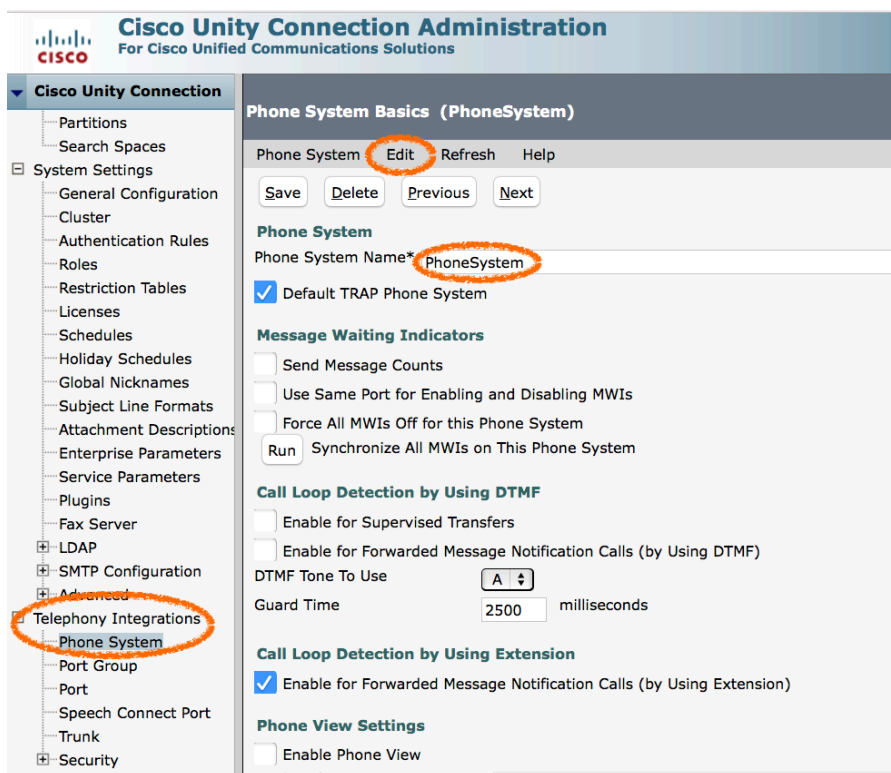
Number of Ports: 2

Save



Next, we want to enable CUC to talk with the CUCM server, perhaps to perform a user import (as one example). While we could create a specific AXL (Administration XML) user that belongs to an AXL administrative group, we'll just use the **administrator** account, since it has access to everything we need. To set this up, let's go back into our Phone System.

Telephony Integrations > Phone System > PhoneSystem



Edit > Cisco Unified Communications Manager AXL Servers

Username: **administrator**

Password: **kwtrain01**

Save

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Cisco Unity Connection

- Partitions
- Search Spaces
- System Settings
 - General Configuration
 - Cluster
 - Authentication Rules
 - Roles
 - Restriction Tables
 - Licenses
 - Schedules
 - Holiday Schedules
 - Global Nicknames
 - Subject Line Formats
 - Attachment Descriptions
 - Enterprise Parameters
 - Service Parameters
 - Plugins
 - Fax Server
 - LDAP

Edit AXL Servers

Phone System Edit Refresh Help

Save

AXL Servers

Delete Selected Add New

Order	IP Address
-------	------------

Delete Selected Add New

AXL Server Settings

Username administrator

Password *****

Cisco Unified Communications Manager Version 5.0 or Greater (SSL)

Save

Now, let's define a couple of AXL servers.

Add New

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Cisco Unity Connection

- Partitions
- Search Spaces
- System Settings
 - General Configuration
 - Cluster
 - Authentication Rules
 - Roles
 - Restriction Tables
 - Licenses
 - Schedules
 - Holiday Schedules
 - Global Nicknames
 - Subject Line Formats
 - Attachment Descriptions
 - Enterprise Parameters
 - Service Parameters
 - Plugins
 - Fax Server
 - LDAP
 - SMTP Configuration
 - Advanced

Edit AXL Servers

Phone System Edit Refresh Help

Save

AXL Servers

Delete Selected Add New

Order	IP Address
-------	------------

Delete Selected Add New

AXL Server Settings

Username administrator

Password *****

Cisco Unified Communications Manager Version 5.0 or Greater (SSL)

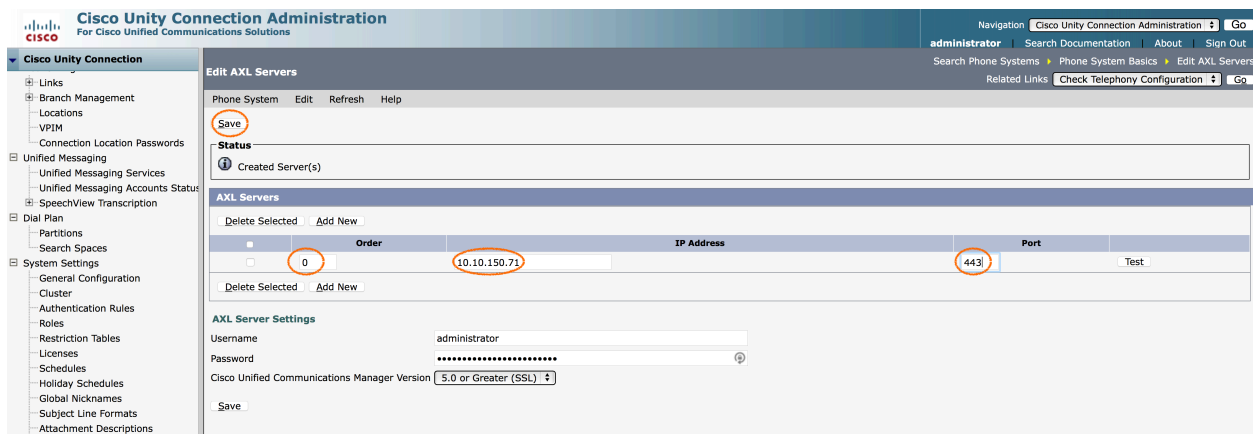
Save

Order: **0**

IP Address: **10.10.150.71**

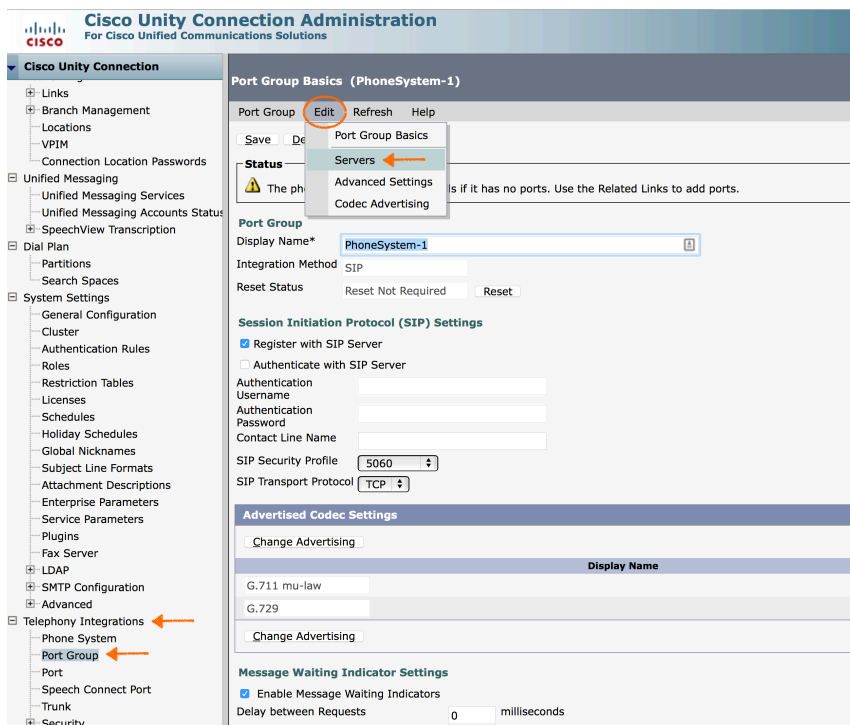
Port: **443** (NOTE: This is the HTTPS default port.)

Save



Let's also define server information for our port group.

Telephony Integrations > Port Group > PhoneSystem-1



Edit > Servers

Update the TFTP server as follows:

Order: 0, IPv4 Address or Host Name, **10.10.150.71**

Save

SIP Servers

Order	IPv4 Address or Host Name	IPv6 Address or Host Name	Port	TLS Port
0	10.10.150.71		5060	5061

TFTP Servers

Order	IPv4 Address or Host Name	IPv6 Address or Host Name
0	10.10.150.71	

IPv6 Addressing Mode

Preference for Signaling: IPv4
 Preference for Media: IPv4

Save

We're now prompted to reset our PhoneSystem-1 Port Group.

Telephony Integrations > Port Group > PhoneSystem-1 > Reset

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Port Group Basics (PhoneSystem-1)

Port Group: PhoneSystem-1

Integration Method: SIP

Reset Status: Reset Required **Reset**

Session Initiation Protocol (SIP) Settings

☒ Register with SIP Server

☐ Authenticate with SIP Server

Authentication Username:
 Authentication Password:
 Contact Line Name:
 SIP Security Profile: 5060
 SIP Transport Protocol: TCP

Verification: Let's take a few seconds to make sure our BR1 IP phones get to their CUC server when we press the **Messages** button on either IP phone. (**NOTE:** If either of the IP phones don't call 3500 when pressing the **Messages** button, reset the IP phone.)

Add Users to the CUC Server

At this point, we have CUC integrated with CUCM, via SIP. However, we don't have any users or user mailboxes configured. Let's do that next.

Before creating a user, we might want to create a user template that contains a collection of settings that can be applied to new users, without individually configuring all of the new users with those settings.

Also, we might want to create an authentication rule, which can dictate a user's password policy. In or case, Task 7.2 specifies a trivial PIN of 12345. That won't work with the default authentication rule. So, let's update it.

System Settings > Authentication Rules > Recommended Voice Mail Authentication Rule

Minimum Credential Length: 5

Check for Trivial Passwords: (*uncheck*)

Save

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

▼ Cisco Unity Connection

Partitions
Search Spaces
System Settings
General Configuration
Cluster
Authentication Rules
Roles
Restriction Tables
Licenses
Schedules
Holiday Schedules
Global Nicknames
Subject Line Formats
Attachment Descriptions
Enterprise Parameters
Service Parameters
Plugins
Fax Server
LDAP
SMTP Configuration
Advanced
Telephony Integrations
Phone System
Port Group
Port
Speech Connect Port
Trunk
Security

Edit Authentication Rule (Recommended Voice Mail Authentication Rule) Search Authentication Rules

Authentication Rule Refresh Help

Save Delete Previous Next

Edit Authentication Rule

Display Name*

Recommended Voice Mail Authentication Rule

Failed Sign-In 3 Attempts No Limit for Failed Sign-Ins

Reset Every Failed Sign-In Attempts 30 Minutes

Lockout Duration 30 Minutes Administrator Must Unlock

Minimum Duration between Credential Changes 1440 Minutes

Credential Expires After 180 Days Never Expires

Expiration Warning Days 15 Days

Minimum Credential Length 5

Stored Number of Previous Credentials 5

Check for Trivial Passwords

Save Delete Previous Next

Fields marked with an asterisk (*) are required.

Next, let's modify the default voice mail user template.

Templates > User Templates > voicemailusertemplate

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

▼ Cisco Unity Connection

Users
Import Users
Synch Users
Class of Service
Class of Service
Class of Service Member
Templates
User Templates
Call Handler Templates
Contact Templates
Notification Templates
Contacts
Contacts
Distribution Lists
System Distribution List

Search User Templates

User Template Refresh Help

Status

Found 2 User Template(s)

User Templates (1 - 2 of 2)

Find User Templates where Alias begins with Find

Alias

administratortemplate

voicemailusertemplate

Delete Selected Add New

Since Task 7.2 makes no mention of us having to record our name, we can say that we don't want our users to self-enroll.

Set for Self-enrollment at Next Sign-In: (*unchecked*)

The screenshot shows the 'Edit User Template Basics' page for 'voicemailusertemplate'. The left sidebar contains a navigation tree with categories like Users, Class of Service, Templates, Contacts, Distribution Lists, Call Management, Message Storage, and Mailbox Stores. The main content area has tabs for 'User Template', 'Edit', 'Refresh', and 'Help'. The 'User Template' tab is active, showing fields for Name, Alias*, Display Name*, Display Name Generation (radio buttons for 'First Name, Then Last Name' and 'Last Name, Then First Name'), Phone, Outgoing Fax Server, Partition, Search Scope, Phone System, Class of Service, and Active Schedule. The 'Save' button is circled in orange. Below the 'Active Schedule' field, there are checkboxes for 'Set for Self-enrollment at Next Sign-In', 'List in Directory', and 'Send Non-Delivery Receipts on Failed Message Delivery'. There are also checkboxes for 'Skip PIN When Calling From a Known Extension' and 'Use Short Calendar Caching Poll Interval'. A 'Location' field is at the bottom.

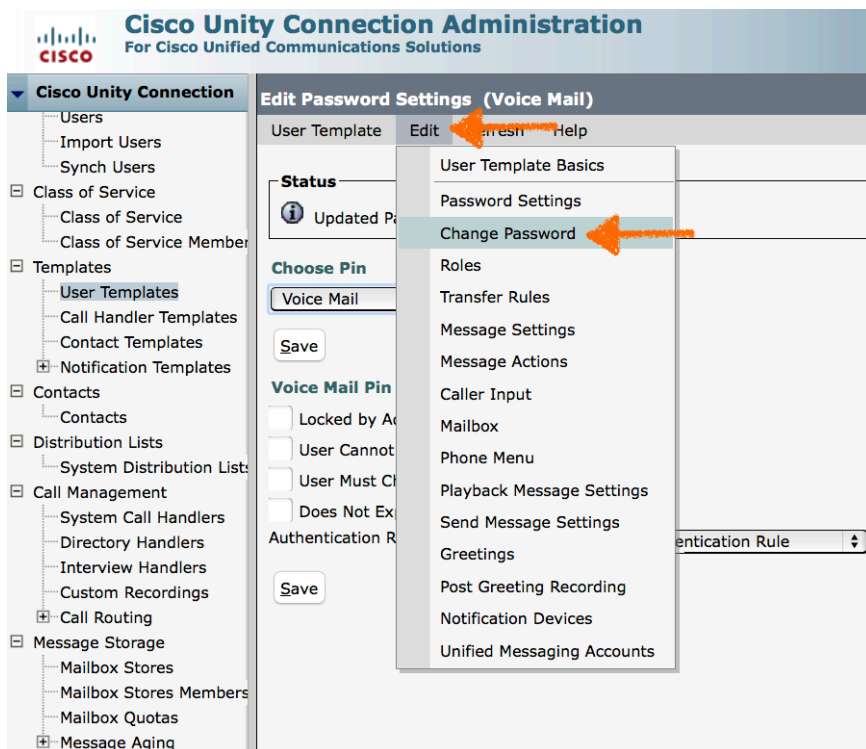
Next, let's update the password settings associated with this default user template.

Edit > Password Settings > User Must Change at Next Sign-In (*unchecked*) > Save

The screenshot shows the 'Edit Password Settings (Voice Mail)' page. The left sidebar is the same as the previous screenshot. The main content area has tabs for 'User Template', 'Edit', 'Refresh', and 'Help'. The 'Edit' tab is active, showing a 'Choose Pin' dropdown menu set to 'Voice Mail'. Below this is a 'Save' button, which is circled in orange. Under the 'Voice Mail Pin Settings' section, there are checkboxes for 'Locked by Administrator', 'User Cannot Change', 'User Must Change at Next Sign-In' (which is unchecked), and 'Does Not Expire'. An arrow points to the 'User Must Change at Next Sign-In' checkbox. At the bottom, there is an 'Authentication Rule' dropdown menu set to 'Recommended Voice Mail Authentication Rule' and another 'Save' button.

Note that we're using the default authentication rule that we modified a few moments ago. So, we can create a password that meets the criteria of the authentication rule.

Edit > Change Password



Pin: **12345**
 Confirm Pin: **12345**
Save



Next, let's import users from CUCM into CUC.

Users > Import Users > Find End Users In (Phone System) > Find
Check the checkboxes next to **hqphone1 and **hqphone2**.**

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration | Go
administrator | Search Documentation | About | Sign Out

Cisco Unity Connection **Import Users**
Import Users Refresh Help

Status
Found 2 Unified Communications Manager User(s)

Find
Find End Users In PhoneSystem
Where Alias Begins With Find

Import With
Based on Template voicemailusertemplate

Directory Search Results
Import Selected Import All 25 Rows Per Page

	Alias	First Name	Last Name	Extension
<input checked="" type="checkbox"/>	br1phone1	BR1	Phone 1	3001
<input checked="" type="checkbox"/>	br1phone2	BR1	Phone 2	3002
<input checked="" type="checkbox"/>	Import Selected Import All			

We should receive a message indicating we had two successful user imports.

Import Users
Import Users Refresh Help

Status
Importing users completed. Number of successes: 2 - Number of failures: 0

Find
Find End Users In PhoneSystem
Where Alias Begins With Find

Verification: Let's quickly verify that by pressing the Messages button on BR1 Phone 1 and on BR1 Phone 2, we can log into (using the PIN of 12345) the mailbox associated with each IP phone's user.

Task 7.3 CUC Customization

To complete this task, we need to configure the **Caller Input** option for the **hqphone1** user.

Users > Users > hqphone1

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration | administrator | Search Documentation | About | Sign Out

Cisco Unity Connection

- Users
 - Users
 - Import Users
 - Synch Users
- Class of Service
 - Class of Service
 - Class of Service Membership
- Templates
 - User Templates
 - Call Handler Templates
 - Contact Templates
 - Notification Templates
- Contacts
 - Contacts
- Distribution Lists
 - System Distribution Lists
- Call Management
 - System Call Handlers
 - Directory Handlers
 - Interview Handlers
 - Custom Recordings
- Call Routing
 - Call Routing
- Message Storage
 - Mailbox Stores
 - Mailbox Stores Membership
 - Mailbox Quotas
 - Message Aging
- Networking
 - Links
 - Branch Management

Search Users

User Refresh Help

Status: Found 7 User(s)

Search Limits: Limit search to All

Users (1 - 7 of 7)

Find Users where	Alias	Extension	First Name	Last Name	Display Name
	administrator				administrator
	hqphone1	2001	HQ	Phone 1	HQ Phone 1
	hqphone2	2002	HQ	Phone 2	HQ Phone 2
	operator	99990			Operator
	Replication		Replication	Agent	Replication Agent (suc-1)
	undeliverablemessagesmailbox	99999			Undeliverable Messages
	UnityConnection		Cisco Unity Connection	Messaging System	Cisco Unity Connection Messaging System

Delete Selected Add New Bulk Edit Show Dependencies

Key: Local User Remote User Cisco Unity User

Edit > Caller Input

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration | administrator | Search Documentation | About | Sign Out

Cisco Unity Connection

- Users
 - Users
 - Import Users
 - Synch Users
- Class of Service
 - Class of Service
 - Class of Service Membership
- Templates
 - User Templates
 - Call Handler Templates
 - Contact Templates
 - Notification Templates
- Contacts
 - Contacts
- Distribution Lists
 - System Distribution Lists
- Call Management
 - System Call Handlers
 - Directory Handlers
 - Interview Handlers
 - Custom Recordings
- Call Routing
 - Call Routing
- Message Storage
 - Mailbox Stores
 - Mailbox Stores Membership
 - Mailbox Quotas
 - Message Aging
- Networking
 - Links
 - Branch Management

Edit User Basics (hqphone1)

User Edit Refresh Help

Save User Basics

Status: Unified Communications Manager end user. Some fields may be disabled.

Name: Message Waiting Indicators

Alias*: Transfer Rules

First N: Message Settings

Last N: Caller Input

Display: Mailbox

SMTP: Phone Menu

Initials: Playback Message Settings

Title: Send Message Settings

Employ: Message Actions

LDAP: Greetings

Inte: Post Greeting Recording

Do: Notification Devices

Phone: Alternate Extensions

Extens: Alternate Names

Cross: Private Distribution Lists

Outgo: Unified Messaging Accounts

SMTP Proxy Addresses

8

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration | administrator | Search Documentation | About | Sign Out

Cisco Unity Connection

- Users
 - Users
 - Import Users
 - Synch Users
- Class of Service
 - Class of Service
 - Class of Service Membership
- Templates
 - User Templates
 - Call Handler Templates
 - Contact Templates
 - Notification Templates
- Contacts
 - Contacts
- Distribution Lists
 - System Distribution Lists
- Call Management
 - System Call Handlers
 - Directory Handlers
 - Interview Handlers
 - Custom Recordings
- Call Routing
 - Call Routing
- Message Storage
 - Mailbox Stores
 - Mailbox Stores Membership
 - Mailbox Quotas
 - Message Aging

Caller Input

User Edit Refresh Help

Save

Key	Action	Target	Status
*	Send caller to	Sign-In	Locked
#	Skip greeting		Locked
0	Send caller to	Operator	Unlocked
1	Ignore key		Unlocked
2	Ignore key		Unlocked
3	Ignore key		Unlocked
4	Ignore key		Unlocked
5	Ignore key		Unlocked
6	Ignore key		Unlocked
7	Ignore key		Unlocked
8	Ignore key		Unlocked
9	Ignore key		Unlocked

Wait for Additional Digits 1500 milliseconds

Prepend Digits to Dialed Extensions

Enable

Digits to Prepend

Save

User with Mailbox: hqphone2

Go Directly to Greetings: (*selected*)
Save

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Cisco Unity Connection

- Users
 - Users
 - Import Users
 - Synch Users
- Class of Service
 - Class of Service
 - Class of Service Membership
- Templates
 - User Templates
 - Call Handler Templates
 - Contact Templates
 - Notification Templates
- Contacts
 - Contacts
- Distribution Lists
 - System Distribution Lists
- Call Management
 - System Call Handlers
 - Directory Handlers
 - Interview Handlers
 - Custom Recordings
- Call Routing
- Message Storage
 - Mailbox Stores
 - Mailbox Stores Membership
 - Mailbox Quotas
 - Message Aging
- Networking

Edit Caller Input (8)

User Edit Refresh Help

Save

Edit Caller Input

Key 8

☐ Ignore Additional Input (Locked)

Action

☐ Call Action Ignore

Extension Description

Transfer Type Release to Switch

Rings to Wait For 4

☐ Call Handler Goodbye

☒ Attempt Transfer

☐ Go Directly to Greetings

☐ Interview Handler

☐ Directory Handler System Directory Handler

☐ Conversation Broadcast Message Administrator

☒ User with Mailbox hqphone2

☐ Attempt Transfer

☒ Go Directly to Greetings

Save

Verification: We can now quickly verify that a call to HQ Phone 1 (where HQ Phone 1 does not answer within 10 seconds) can be diverted to the voicemail box of HQ Phone 2 by pressing **8** as HQ Phone 1's greeting is being played.

Module 14: Doing EVERYTHING on Cisco Unified Contact Center Express (UCCX)

Task 8.1 UCCX Integration

Configure IPCC Extensions on IP phones

The task tells us that we're going to have two IP Phone Agents, hqphone1 and hqphone2. So, the first thing we do is go into the user configuration screen for those users, and assign them to an appropriate IPCC extension (i.e. a DN of 2001 for hqphone1 and 2002 for hqphone2).

User Management > End User > Find > hqphone1

IPCC Extension: 2001 in INTERNAL

Save

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go
administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

End User Configuration Related Links: Back to Find List Users Go

Save Delete Add New

Directory Number Associations

Primary Extension: < None >
IPCC Extension: 2001 in INTERNAL

Go
hqphone2
 IPCC Extension: **2002 in INTERNAL**
Save

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go
administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

End User Configuration Related Links: Back to Find List Users Go

Save Delete Add New

Directory Number Associations

Primary Extension: 2002 in INTERNAL
IPCC Extension: 2002 in INTERNAL

Mobility Information

Create an IP Phone Service

Let's go to Cisco's documentation (which we will have available in the lab) for some assistance in getting the URL for the IP phone service needed for an IP phone agent.

NOTE: When practicing finding information in documentation (simulating what you might need to do on the lab), you can start from the following location:

<http://www.cisco.com/cisco/web/psa/default.html>

NOTE: Cisco periodically changes the menu structure and labeling of their documentation site. Therefore, what you see on Cisco's current site might differ somewhat from the following screen shots.

Install and Upgrade > Products > Customer Collaboration > Contact Center Solutions > Unified Contact Center Express > Unified Contact Center Express 9.0(2) > Cisco Unified Contact Center Express Installation and Upgrade Guide, Release 9.0(2) > Cisco Desktop Product Suite Installation Guide

Install and Upgrade

Select Your Product or Technology

HOME
SUPPORT & DOWNLOADS
PRODUCT/TECHNOLOGY SUPPORT

- Download Software
- Release and General Information
- Reference Guides
- Design
- Install and Upgrade**
- Configure
- Maintain and Operate
- Troubleshoot

Select

Select a Product

Or Browse Products and Technologies

Support Home > Products > Customer Collaboration

Recently Used Products
My Added Devices
Add Device

Products >

Technology

End-of-Sale / End-of-Life

Application Networking Services
Cisco Interfaces and Modules
Cloud and Systems Management
Collaboration Endpoints
Conferencing
Connected Safety and Security
IOS and NX-OS Software
Optical Networking
Routers
Security
Servers - Unified Computing
Service Exchange
Software
Storage Networking
Switches
Unified Communications
Universal Gateways and Access Servers
Video
Wireless

Contact Center Solutions
Options for Contact Center Solutions

Install and Upgrade

Select Your Product or Technology

HOME
SUPPORT & DOWNLOADS
PRODUCT/TECHNOLOGY SUPPORT

- Download Software
- Release and General Information
- Reference Guides
- Design
- Install and Upgrade**
- Configure
- Maintain and Operate
- Troubleshoot

Select

Select a Product

Or Browse Products and Technologies

Support Home > Products > Customer Collaboration > Contact Center Solutions > Unified Contact Center Express

Recently Used Products
My Added Devices
Add Device

Products >

Technology

End-of-Sale / End-of-Life

Hosted Collaboration Solution for Contact Center
Packaged Contact Center Enterprise
Unified Contact Center Enterprise
Unified Contact Center Express >
Unified Contact Center Hosted

Unified Contact Center Express
Customer Response Solution Downloads
Customer Response Solutions 6.0
Unified Contact Center Express 10.6(1)
Unified Contact Center Express 10.5(1)
Unified Contact Center Express 10.0(1)
Unified Contact Center Express 9.0(2)
Unified Contact Center Express 9.0(1)
Unified Contact Center Express 8.5(1)
Unified Contact Center Express 8.0(2)
Unified Contact Center Express 8.0(1)
Unified Contact Center Express 7.0(2)
Unified Contact Center Express 7.0(1)

Install and Upgrade Guides

- [Cisco Desktop Product Suite Installation Guide \(ICD 4.5\(5\) - PDF\) \(PDF - 929 KB\)](#)
- [Cisco Unified Contact Center Express Installation and Upgrade Guide, Release 9.0\(2\)](#)
- [Cisco Unified Contact Center Express Windows To Linux Upgrade Guide, Release 9.0\(2\)](#)

Search the PDF for service url.

Creating an IP Phone Service

From the Cisco CallManager Administration web-based application, follow these steps to create a new IP phone service.

► To create a new IP phone service:

- From the menu at the top of the page, click **Feature > IP Phone Service**.
- On the Cisco IP Phone Services Configuration page, enter the following information:

Service Name. Enter the service name that will be shown in the IP phone Services window.

Service Description. Optional. Enter a description of the service.

Service URL. Enter the URL for the service. For example:
<http://192.168.252.44:6293/ipphone/jsp/sciphonexml/IPAgentInitial.jsp>
 where:

The task is requiring One Button Logon. Therefore, this URL needs to be slightly modified, replacing **Initial** with **Login**. The IP Address also needs to be updated with the IP address of the UCCX server. Therefore, the service URL is:

http://192.168.1.74:6293/ipphone/jsp/sciphonexml/IPAgentLogin.jsp

We now take that URL and create the IP phone service.

Device > Device Settings > Phone Services > Add New

Service Name: **IP Phone Agent**

Service URL: **http://192.168.1.74:6293/ipphone/jsp/sciphonexml/IPAgentLogin.jsp**

Enable: *(checked)*

Save

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾

IP Phone Services Configuration

Save

Status
 Status: Ready

Service Information

Service Name* IP Phone Agent

ASCII Service Name* IP Phone Agent

Service Description

Service URL* http://192.168.1.74:6293/ipphone/jsp/sciphonexml/

Secure-Service URL

Service Category* XML Service

Service Type* Standard IP Phone Service

Service Vendor

Service Version

☒ Enable

☐ Enterprise Subscription

Save

The One Button Login feature provides login credentials as part of the service. So, we need to add parameters to the service.

New Parameter

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾

IP Phone Services Configuration

Save
 Delete
 Update Subscriptions
 Add New

Status

Add successful

Service Information

Service Name*

ASCII Service Name*

Service Description

Service URL*

Secure-Service URL

Service Category*

Service Type*

Service Vendor

Service Version

☒ Enable

Service Parameter Information

Parameters

- Save
 Delete
 Update Subscriptions
 Add New

Parameter Name: **Ext**
 Parameter Display Name: **Ext**
 Parameter Description: **Extension**
Save > Add New

Configure Cisco IP Phone Service Parameter

Save
 Help

Status

Status: Ready

Service Parameter Information

Parameter Name*

Parameter Display Name*

Default Value

Parameter Description*

☒ Parameter is Required
☐ Parameter is a Password (mask contents)

Save
 Save And Close

Parameter Name: **ID**
 Parameter Display Name: **ID**

Parameter Description: **Agent ID**
Save > Add New

Configure Cisco IP Phone Service Parameter

Save Delete Copy Add New Help

Status
Add successful

Service Parameter Information

Parameter Name*
ID

Parameter Display Name*
ID

Default Value

Parameter Description*
Agent ID

☒ Parameter is Required
☐ Parameter is a Password (mask contents)

Save Save And Close Delete Copy Add New

Parameter Name: **Pwd**
Parameter Display Name: **Pwd**
Parameter Description: **Agent Password**
Parameter is a Password: *(checked)*
Save and Close > Save

Configure Cisco IP Phone Service Parameter

Save Help

Status
Status: Ready

Service Parameter Information

Parameter Name*
Pwd

Parameter Display Name*
Pwd

Default Value

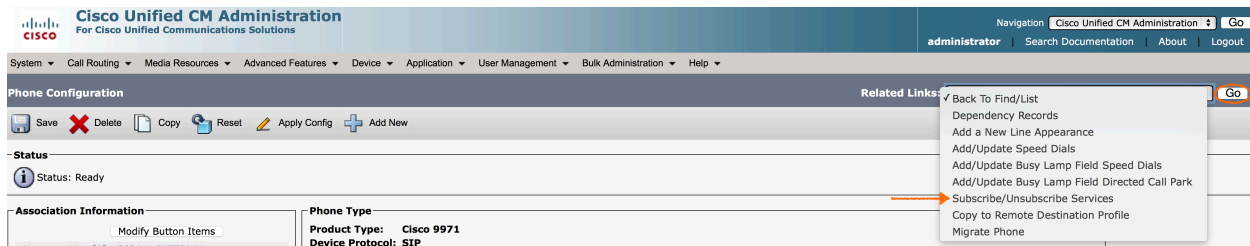
Parameter Description*
Agent Password

☒ Parameter is Required
☒ Parameter is a Password (mask contents)

Save Save And Close

[Subscribe IP Phones to the IP Phone Service](#)

Device > Phone > Find > 9971 (i.e. HQ Phone 1)
Related Links: **Subscribe/Unsubscribe Services**
Go



Select a Service: **IP Phone Agent**

Next

Ext: **2001**

ID: **hqphone1**

Pwd: **cisco**

Subscribe > Save > (close window)

Subscribed Cisco IP Phone Services for SEP0CD996912474

Next Help

Status
Status: Ready

Service Information
Service Subscription: New
Select a Service* **IP Phone Agent**
Service Description

Subscribed Services

Next Close

*- Indicates required item.

Subscribed Cisco IP Phone Services for SEP0CD996912474

Save Help

Status
Status: Ready

Service Information
Service Subscription: IP Phone Agent
Service Name* **IP Phone Agent**
ASCII Service Name* **IP Phone Agent**
Ext **2001** (Description)
ID **hqphone1** (Description)
Pwd ********* (Description)

Subscribed Services

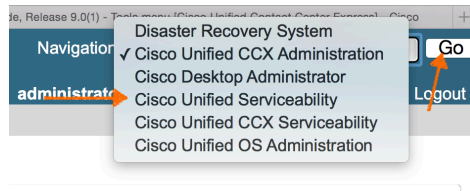
Subscribe Back

Repeat this process for hqphone2.

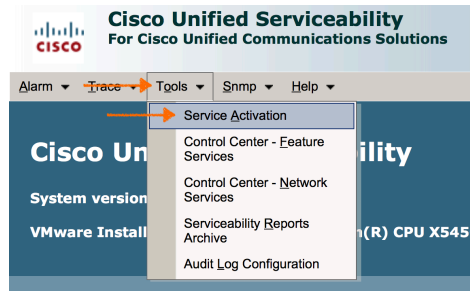
Activate UCCX Services

Just as we did with CUCM and CUC, let's activate all services on our UCCX server, to accommodate any task we might face.

Navigation > Cisco Unified Serviceability > Go



Tools > Service Activation



Check All Services: (*checked*)

Save > OK

Now, we can navigate back to the main administration screen.



Related Links > Cisco Unified CCX Administration > Go

Examine Current UCCX Configuration

Now that we have our IP phones subscribed to the IP Phone Agent service and configured for One Button Login, let's examine the current configuration of UCCX.

Check System Parameters

System > System Parameters

Verify the following settings:

Codec: **G711**

Agent State after Ring No Answer: **Not Ready**

Cisco Unified CCX Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CCX Administration | Go

System Applications Subsystems Wizards Tools Help

administrator Search Documentation About Logout

System Parameters Configuration

Update Clear

Generic System Parameters		
Parameter Name	Parameter Value	Suggested Value
System Time Zone*	Eastern Standard Time	

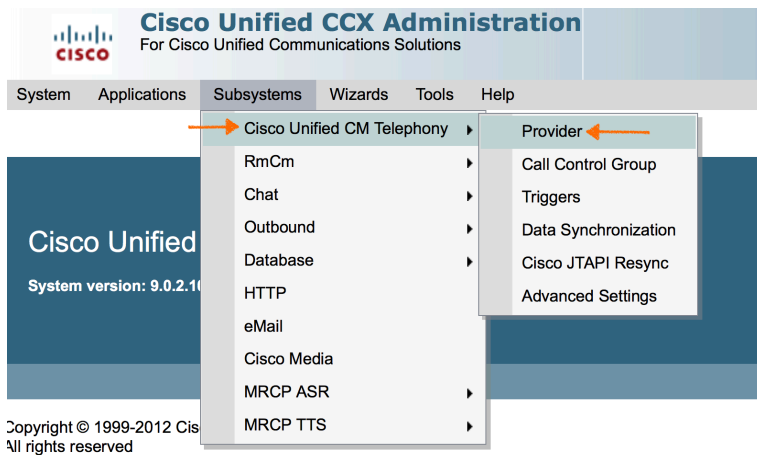
Internationalization Parameters		
Parameter Name	Parameter Value	Suggested Value
Customizable Locales		
Default Currency*	American Dollar (USD) Edit	American Dollar

Media Parameters		
Parameter Name	Parameter Value	Suggested Value
Codec	G711	G711
Recording Count*	0	0
Default TTS Provider	< NONE >	
User Prompts override System Prompts	<input type="radio"/> Disable <input checked="" type="radio"/> Enable	Disable

Application Parameters		
Parameter Name	Parameter Value	Suggested Value
Supervisor Access	No Access to Teams	
Max Number of Executed Steps*	1000	1000
Additional Tasks*	0	0
Default Session Timeout*	30 minutes	30 minutes
Enterprise Call Info Parameter Separator*		
Agent State after Ring No Answer*	<input type="radio"/> Ready <input checked="" type="radio"/> Not Ready	Not Ready
Number of HR sessions*	0 (Number of Seats : 6)	0
Number of Outbound seats*	6 (Maximum limit : 6)	

Check Cisco Unified CM Telephony Settings

Subsystems > Cisco Unified CM Telephony > Provider



Modify Cisco Unified CM Telephony Provider Information

Cisco Unified CCX Administration
For Cisco Unified Communications Solutions

System Applications Subsystems Wizards Tools Help

Cisco Unified CM Telephony Provider

Modify Cisco Unified CM Telephony Provider Information

Primary Cisco Unified CM Telephony Provider	192.168.1.72
Secondary Cisco Unified CM Telephony Provider	192.168.1.71
User Prefix	jtapi

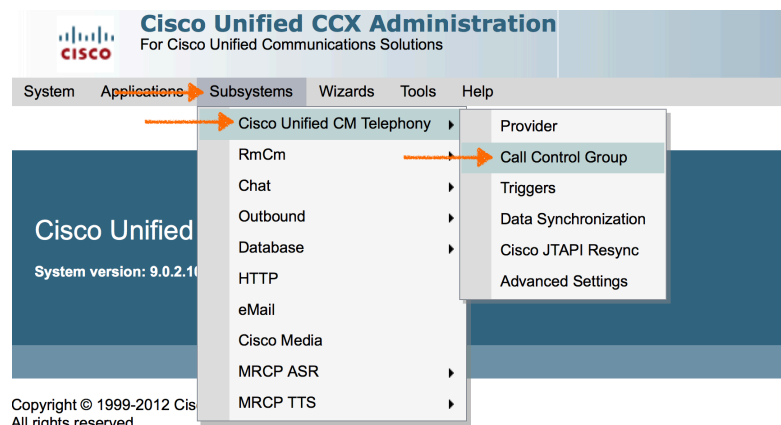
We want both of our HQ CUCM servers listed as **Selected AXL Service Providers** and **Selected CTI Managers**, with the Subscriber listed first. In this case, it looks correct.

However, if it were different, we would want to move the appropriate **Available AXL Service Providers** and **Available CTI Managers** server(s) to the appropriate location in the **Selected AXL Service Providers** and **Selected CTI Managers** fields, and then click **Update**.

Check for an Existing Cisco Unified CCM Telephony Call Control Group

Next, we check to see if we have any ports defined in a Call Control Group on UCCX. If we do, we want to see if they exist and are reachable on our CUCM cluster.

Subsystems > Cisco Unified CM Telephony > Call Control Group > IPCC



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Cisco Unified CCX Administration
For Cisco Unified Communications Solutions

System Applications Subsystems Wizards Tools Help

Cisco Unified CM Telephony Call Control Group Configuration

Add New Refresh All

status

1 records found

Ports List	
Group ID	Description
2	IPCC

Add New Refresh All

Number Of Licensed IVR Ports:12
 Number Of Licensed Outbound IVR Ports:6

We see that we do have a Call Control Group with two CTI ports, with DNs of 2701 and 2702, which are the DNs Task 8.1 says we should use. However, we need to see if this information is synched with our CUCM cluster.

Subsystems > Cisco Unified CM Telephony > Data Synchronization

Cisco Unified CCX Administration
For Cisco Unified Communications Solutions

System Applications Subsystems Wizards Tools Help

Cisco Unified CM Telephony

- Provider
- Call Control Group
- Triggers
- Data Synchronization**
- Cisco JTAPI Resync
- Advanced Settings

Group Information

Group ID* 2

Description* IPCC

Number Of CTI Ports* 2

Media Termination Support ☐ Yes ☒ No

Group Type* ☒ Inbound

Call Control Groups: *(checked)*
Data Check > OK

The output suggests that the CTI ports do not exist in CUCM.

Cisco Unified CCX Administration
For Cisco Unified Communications Solutions

System Applications Subsystems Wizards Tools Help

Cisco Unified CM Telephony Data Synchronization

Data Check Data Resync

Status

Data Check operation is completed Successfully.

Please select at least one component.

Call Control Group(s) ☒ Trigger(s) ☐ CM Telephony User(s) ☐

Data Check Results	Status	Node Data
Components Selected		
▼ Call Control Group(s)		
▼ v2 - IPCC	X	
▼ CTI Ports	X	Out of Sync
Missing	X	[IPCC_2701, IPCC_2702]
▼ CTI Lines	✓	In Sync
Device Association	X	Out of Sync
▼ Attributes	✓	In Sync

Let's verify that those ports are missing in CUCM.

(Back on the CUCM Administrative Interface)
Device > Phone > Find

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go administrator Search Documentation About Logout

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

Find and List Phones Related Links: Actively Logged In Device Report Go

Add New Select All Clear All Delete Selected Reset Selected Apply Config to Selected

Status

2 records found

Phone (1 - 2 of 2) Rows per Page 50

Find Phone where Device Name begins with Find Clear Filter

Device Name(Line)	Description	Device Pool	Device Protocol	Status	IP Address	Copy	Super Copy
SEP0CD996912424	HQ Phone 1	HQ	SIP	Registered with 192.168.1.72	10.10.120.19		
SEP0CC8821098E9	HQ Phone 2	HQ	SCCP	Registered with 192.168.1.72	10.10.120.17		

Add New Select All Clear All Delete Selected Reset Selected Apply Config to Selected

We need a couple of CTI ports (with DNs of 2701 and 2702) to be used by UCCX, and they don't currently exist. Before we tell UCCX to synchronize with CUCM, let's make sure that DNs of 2701 and 2702 don't exist in the CUCM database. If they do exist, we should delete them. Otherwise, when we tell UCCX to synchronize with CUCM, DNs of 2703 and 2704 would be used.

Call Routing > Directory Number > Find

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go administrator Search Documentation About Logout

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

Find and List Directory Numbers

Add New Select All Clear All Delete Selected

Status

4 records found

Directory Number (1 - 4 of 4) Rows per Page 50

Find Directory Number where Directory Number begins with Find Clear Filter

Pattern/Directory Number	Partition	Description	Copy
2001	INTERNAL	HQ Phone 1	
2002	INTERNAL	HQ Phone 2	
2701		CRS Line description	
2702		CRS Line description	

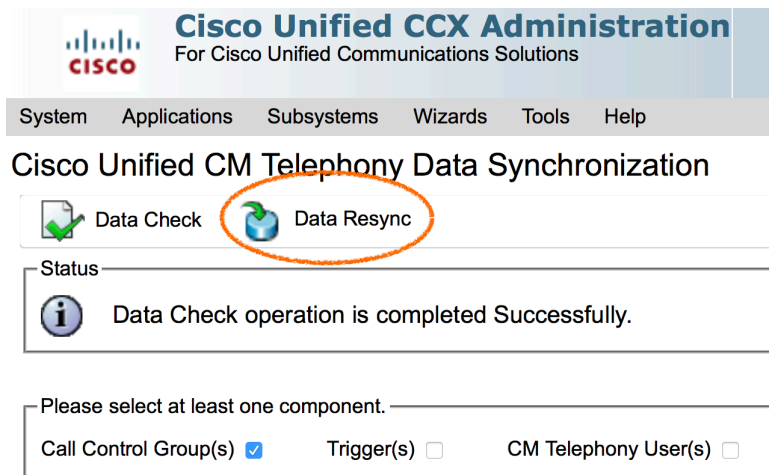
Add New Select All Clear All Delete Selected

In this case, DNS of 2701 and 2702 do exist. So, we should select the checkboxes next to them, and click the **Delete Selected** button, and then click **OK**.

Now, we can go back and perform the synchronization from UCCX.

(Back on the UCCX Administrative Interface)

Data Resync > OK

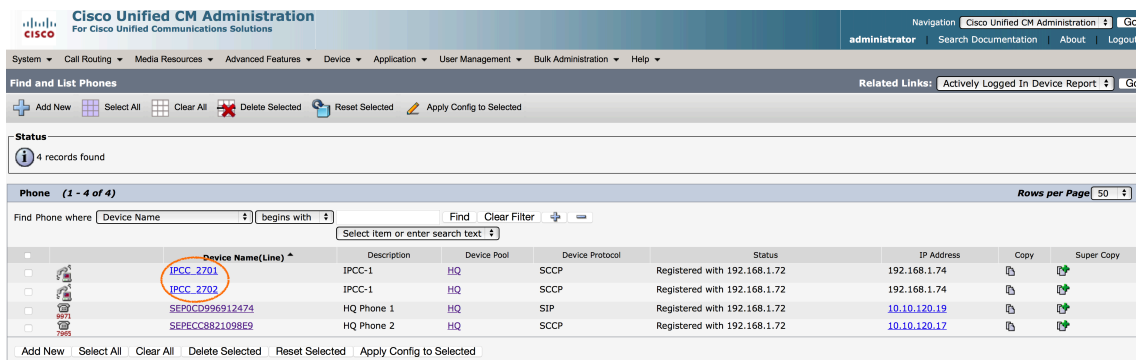


The screenshot shows the Cisco Unified CCX Administration interface. The top navigation bar includes System, Applications, Subsystems, Wizards, Tools, and Help. The main heading is "Cisco Unified CM Telephony Data Synchronization". Below this, there are two buttons: "Data Check" (with a green checkmark icon) and "Data Resync" (with a globe icon). The "Data Resync" button is circled in orange. Below the buttons, a status message states: "Data Check operation is completed Successfully." At the bottom, there is a section titled "Please select at least one component." with three checkboxes: "Call Control Group(s)" (checked), "Trigger(s)" (unchecked), and "CM Telephony User(s)" (unchecked).

After the synchronization completes, we should have two CTI ports created on CUCM, which we can quickly verify.

(Back on the CUCM Administrative Interface)

Device > Phone > Find



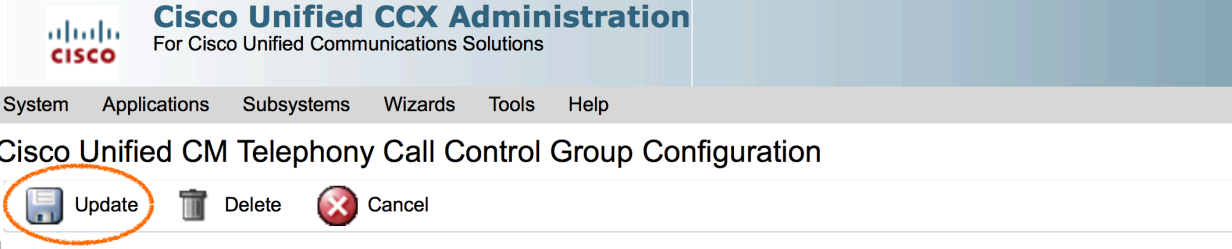
The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main heading is "Find and List Phones". Below this, there is a table with 4 records found. The table has columns: Phone, Device Name(Line), Description, Device Pool, Device Protocol, Status, IP Address, Copy, and Super Copy. The first two rows are circled in orange.

Phone	Device Name(Line)	Description	Device Pool	Device Protocol	Status	IP Address	Copy	Super Copy
IPCC-1	IPCC-2701	IPCC-1	HQ	SCCP	Registered with 192.168.1.72	192.168.1.74	Copy	Super Copy
IPCC-1	IPCC-2702	IPCC-1	HQ	SCCP	Registered with 192.168.1.72	192.168.1.74	Copy	Super Copy
SEP0CD996912474	HQ Phone 1	HQ Phone 1	HQ	SIP	Registered with 192.168.1.72	10.10.120.19	Copy	Super Copy
SEP0CC8821098E9	HQ Phone 2	HQ Phone 2	HQ	SCCP	Registered with 192.168.1.72	10.10.120.17	Copy	Super Copy

These newly created Ports don't have a CSS set, and we shouldn't set the CSS from within CUCM. Instead, we should set it from UCCX.

(Back on the UCCX Administrative Interface)




Subsystems > Cisco Unified CM Telephony > Call Control Group > IPCC > Show More... > DN Calling Search Space > HQ > Update



Cisco Unified CCX Administration
For Cisco Unified Communications Solutions

System Applications Subsystems Wizards Tools Help

Cisco Unified CM Telephony Call Control Group Configuration

 Update  Delete  Cancel

Directory Number Information

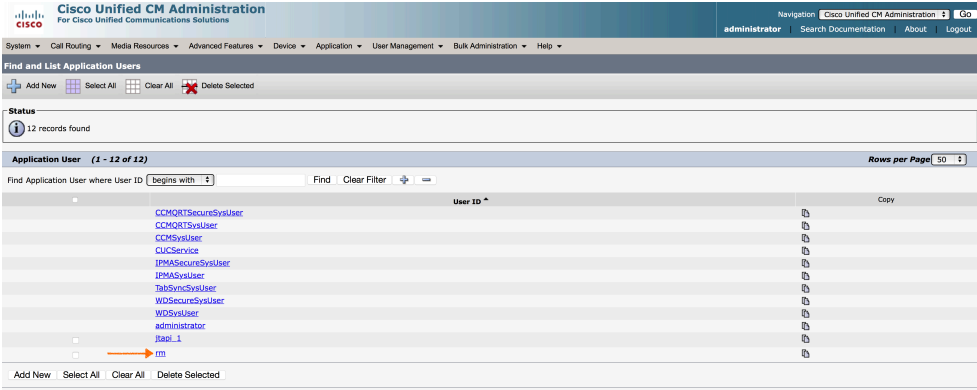
Device Name Prefix*	IPCC
Starting Directory Number*	2701
List of CTI Ports	IPCC_2701,IPCC_2702
Device Pool	HQ
DN Calling Search Space	HQ
Location	Hub_None
Partition	None

Show Less...

Associate IP Phone Agent Phones with the “rm” User

Since UCCX and CUCM were already integrated, there is an Application User named **rm** that already exists. We can confirm this on CUCM.

User Management > Application User > Find



Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration > User Management > Bulk Administration > Help

Find and List Application Users

Add New Select All Clear All Delete Selected

Status: 12 records found

Application User (1 - 12 of 12)	User ID *	Copy
CCMQRTRSecureSysUser		
CCMQRTRSysUser		
CCMSysUser		
CUCService		
IPMASecureSysUser		
IPMASysUser		
TabSvcsSysUser		
WDSecureSysUser		
WDSysUser		
administrator		
jsapi_1		
rm		

Add New Select All Clear All Delete Selected

We need to add **HQ Phone 1** and **HQ Phone 2** as Controlled Devices for this **rm** user.

rm

Find more Phones

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Application User Configuration

Save Delete Copy Add New

Status
Status: Ready

Application User Information

User ID * Edit Credential
 Password
 Confirm Password
 Digest Credentials
 Confirm Digest Credentials
 BLF Presence Group *
☐ Accept Presence Subscription
☐ Accept Out-of-dialog REFER
☐ Accept Unsolicited Notification
☐ Accept Replaces Header

Device Information

Available Devices [Find more Phones](#)
[Find more Route Points](#)

Controlled Devices

Available Profiles

CTI Controlled Device Profiles

Select the checkboxes next to HQ Phone 1 and HQ Phone 3.

Add Selected

Find and List Phones Related Links: [Actively Logged In Device Report](#) Go

Select All Clear All **Add Selected** Close

Status
4 records found

Phone (1 - 4 of 4) Rows per Page 50

Find Phone where begins with Find Clear Filter

		Device Name(Line) ^	Description	Device Pool	Device Protocol	Status	IP Address	Copy	Super Copy
<input type="checkbox"/>		IPCC_2701	IPCC-1	HQ	SCCP	Registered with 192.168.1.72	192.168.1.74		
<input type="checkbox"/>		IPCC_2702	IPCC-1	HQ	SCCP	Registered with 192.168.1.72	192.168.1.74		
<input checked="" type="checkbox"/>		SEP0CD996912474	HQ Phone 1	HQ	SIP	Registered with 192.168.1.72	10.10.120.19		
<input checked="" type="checkbox"/>		SEPECC8821098E9	HQ Phone 2	HQ	SCCP	Registered with 192.168.1.72	10.10.120.17		

Select All Clear All Add Selected Close

Save

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾

Application User Configuration

Save Delete Copy Add New

Status
Status: Ready

Application User Information

User ID* rm Edit Credential

Password

Confirm Password

Digest Credentials

Confirm Digest Credentials

BLF Presence Group* Standard Presence group

☐ Accept Presence Subscription

☐ Accept Out-of-dialog REFER

☐ Accept Unsolicited Notification

☐ Accept Replaces Header

Device Information

Available Devices IPCC_2701 IPCC_2702 SEP0CD996912474 SEPECC8821098E9 Find more Phones Find more Route Points

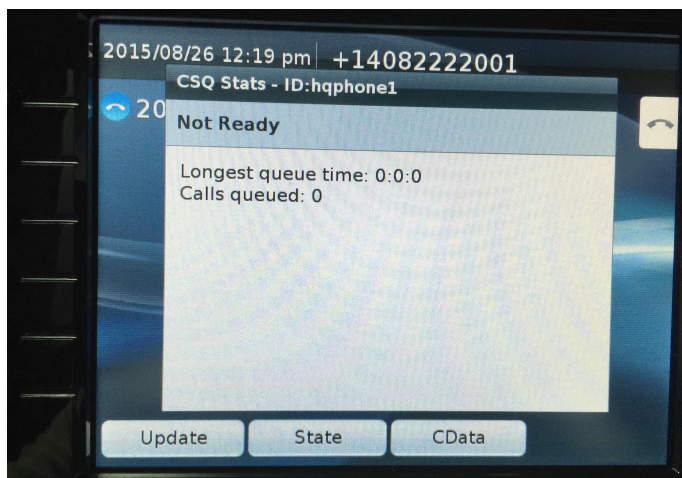
Controlled Devices SEP0CD996912474 SEPECC8821098E9

Available Profiles

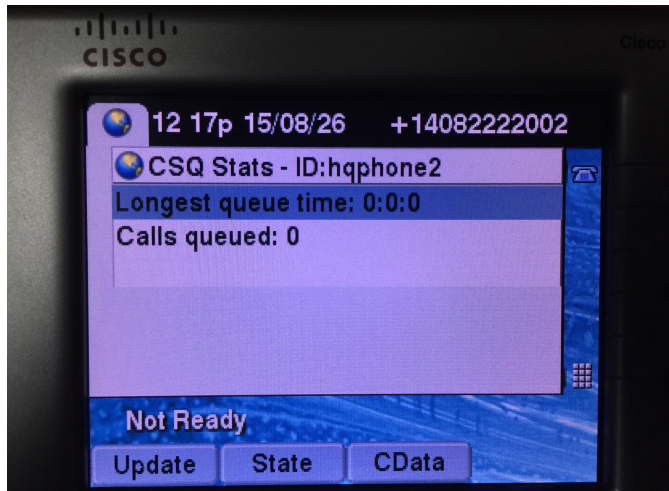
At this point, HQ Phone 1 and HQ Phone 2 should be able to log into the IP Phone Agent service using One Button Login.

You can access this service on HQ Phone 2 by pressing the **Services** button. However, HQ Phone 1 (i.e. the 9971 IP Phone) does not have such a button. Instead, on HQ Phone 1, press the **Applications** button (i.e. the button that looks like a gear). You are then presented with a screen containing the **IP Phone Agent** service. Press that service (on the IP phone's screen) to activate the service.

After logging in, HQ Phone 1 and HQ Phone 2 should appear as seen below:



HQ Phone 1 – Logged into the IP Phone Agent Service

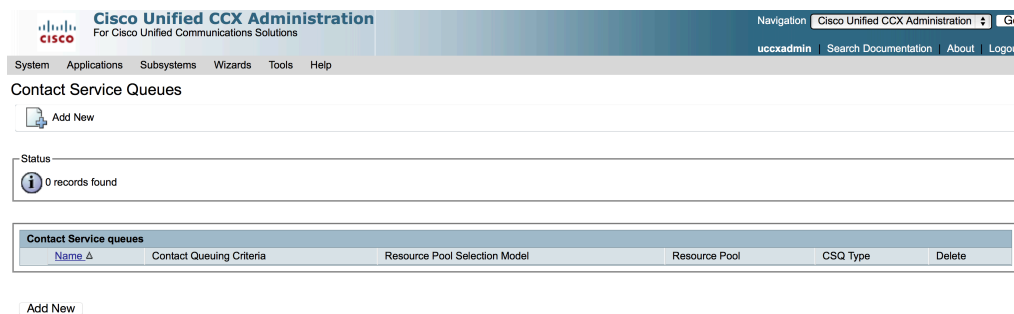


HQ Phone 2 – Logged into the IP Phone Agent Service

Check for Any Existing Contact Service Queues (CSQs)

When our IP Phone Agent users login, they can log into a CSQ. Let's check to see if we have any existing CSQs.

Subsystems > RmCm > Contact Service Queues



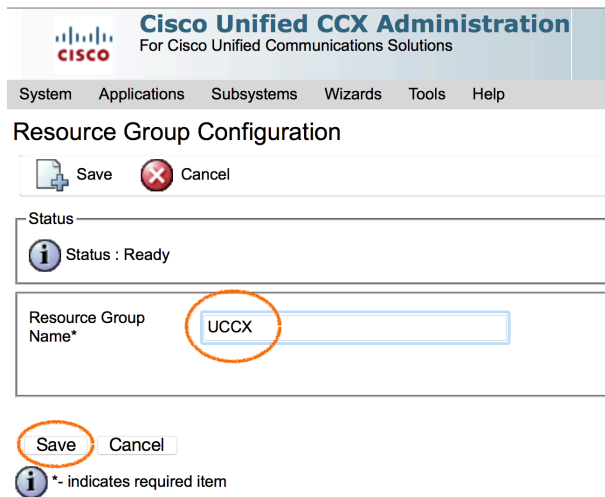
We don't have any CSQs created. So, let's create a Resource Group; populate it with our IP Phone Agents; and assign it to a newly-created CSQ.

Create a Resource Group

Subsystems > RmCm > Resource Groups > Add New

Resource Group Name: **UCCX**

Save



Cisco Unified CCX Administration
For Cisco Unified Communications Solutions

System Applications Subsystems Wizards Tools Help

Resource Group Configuration

Save Cancel

Status

Status : Ready

Resource Group Name* **UCCX**

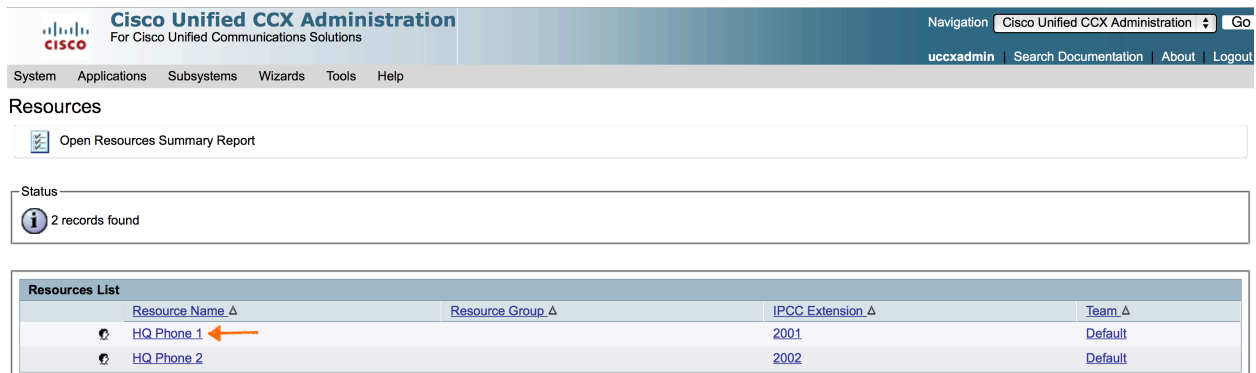
Save Cancel

i *- indicates required item

Assign IP Phone Agents to Resource Group

We can now populate our Resource Group with our IP Phone Agents.

Subsystems > RmCm > Resources > HQ Phone 1



Cisco Unified CCX Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CCX Administration Go
uccxadmin Search Documentation About Logout

System Applications Subsystems Wizards Tools Help

Resources


Open Resources Summary Report

Status

i 2 records found




Resource Name ^Δ	Resource Group ^Δ	IPCC Extension ^Δ	Team ^Δ
HQ Phone 1		2001	Default
HQ Phone 2		2002	Default

Resource Group: **UCCX**
Update


Cisco Unified CCX Administration
 For Cisco Unified Communications Solutions

System Applications Subsystems Wizards Tools Help

Resource Configuration


 Update
  Cancel
  Open Printable Report of this Resource Configuration

Resource Name	HQ Phone 1
Resource ID	hqphone1
IPCC Extension	2001
Resource Group	UCCX
Automatic Available*	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Assigned Skills	Unassigned Skills
Competence Level	5 (1-Beginner, 10-Expert)
Team	Default

HQ Phone 2




Resource Group: UCCX

Update


Cisco Unified CCX Administration
 For Cisco Unified Communications Solutions

System Applications Subsystems Wizards Tools Help

Resource Configuration

 Update
  Cancel
  Open Printable Report of this Resource Configuration

Resource Name	HQ Phone 2
Resource ID	hqphone2
IPCC Extension	2002
Resource Group	UCCX
Automatic Available*	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Assigned Skills	Unassigned Skills
Competence Level	5 (1-Beginner, 10-Expert)
Team	Default

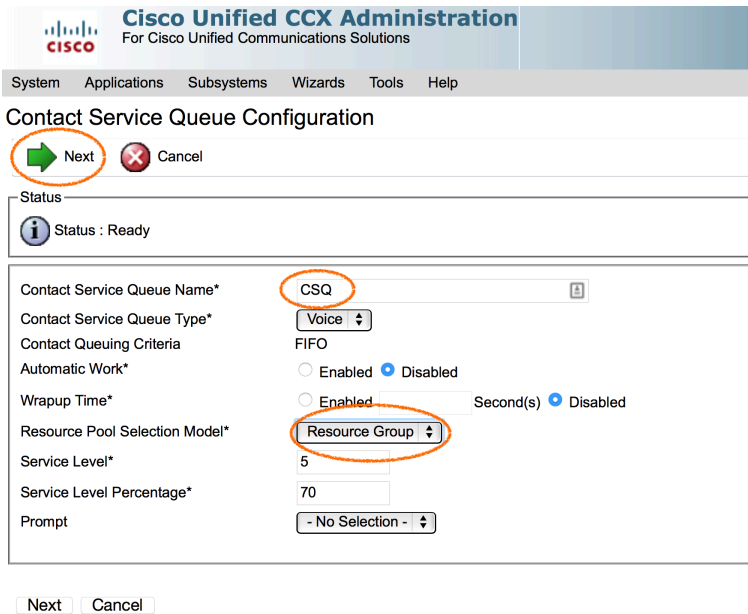
Create a CSQ

Subsystems > RmCm > Contact Service Queues > Add New

Contact Service Queue Name: **CSQ**

Resource Pool Selection Model: **Resource Group**

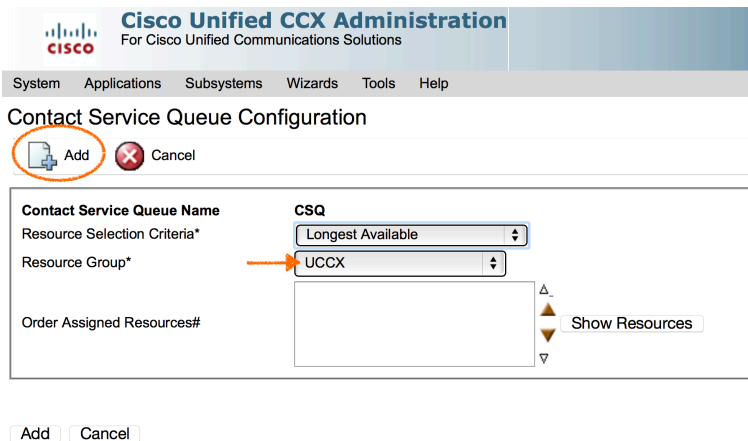
Next



The screenshot shows the 'Contact Service Queue Configuration' page in the Cisco Unified CCX Administration interface. The 'Next' button is circled in orange. The 'Contact Service Queue Name*' field is set to 'CSQ'. The 'Contact Service Queue Type*' is 'Voice'. The 'Contact Queuing Criteria' is 'FIFO'. The 'Automatic Work*' is 'Disabled'. The 'Wrapup Time*' is 'Enabled'. The 'Resource Pool Selection Model*' is 'Resource Group', which is circled in orange. The 'Service Level*' is '5' and the 'Service Level Percentage*' is '70'. The 'Prompt' is '- No Selection -'. The 'Status' is 'Ready'.

Resource Group: **UCCX**

Add

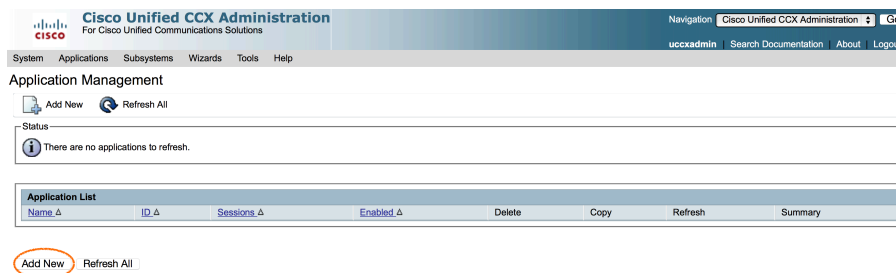


The screenshot shows the 'Contact Service Queue Configuration' page in the Cisco Unified CCX Administration interface. The 'Add' button is circled in orange. The 'Contact Service Queue Name' is 'CSQ'. The 'Resource Selection Criteria*' is 'Longest Available'. The 'Resource Group*' is 'UCCX', which is highlighted with an orange arrow. The 'Order Assigned Resources#' field is empty. The 'Show Resources' button is visible. The 'Add' and 'Cancel' buttons are at the bottom.

Add an Application

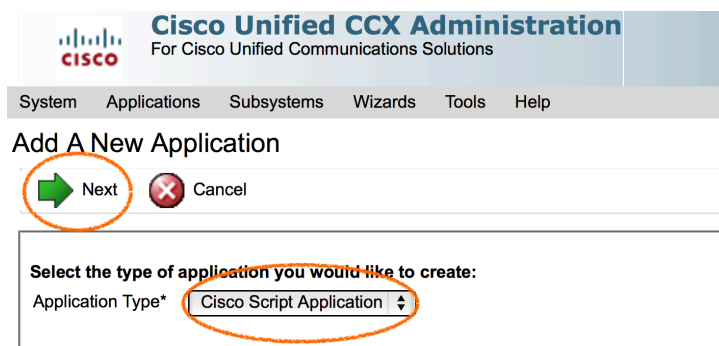
To make sure everything is working up to this point, let's create an application.

Applications > Application Management > Add New



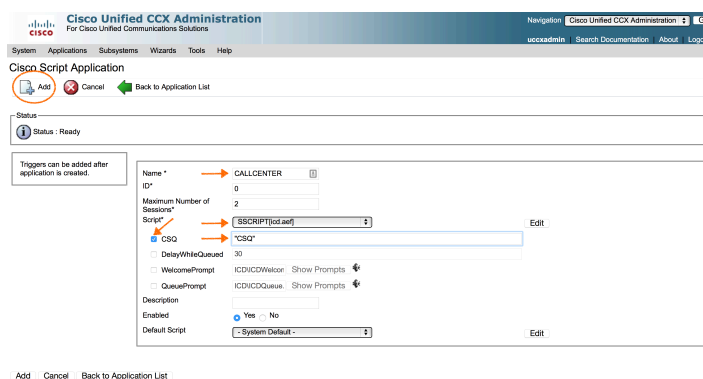
We're required to create an application named **CALLCENTER** in Task 8.2. So, we can create it now to test our configuration thus far, and then modify it in Task 8.2.

Application Type: Cisco Script Application Next



Next Cancel
 ⓘ *- indicates required item

Name: **CALLCENTER**
 Maximum Number of Sessions: **2**
 Script: **SSCRIPT[icd.aef]**
 CSQ: **"CSQ"** (checked)
Add



Add a new trigger

Cisco Unified CCX Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CCX Administration Go
uccxadmin Search Documentation About Logout

System Applications Subsystems Wizards Tools Help

Cisco Script Application

Update Delete Cancel Back to Application List

Status
The operation has been executed successfully.

Add new trigger

Name	CALLCENTER
ID*	0
Maximum Number of Sessions*	2
Script*	SSCRIPT[icd.aef] Edit
<input checked="" type="checkbox"/> CSQ	"CSQ"
<input type="checkbox"/> DelayWhileQueued	30
<input type="checkbox"/> WelcomePrompt	ICD\ICDWelcome Show Prompts
<input type="checkbox"/> QueuePrompt	ICD\ICDQueue Show Prompts
Description	CALLCENTER
Enabled	<input checked="" type="radio"/> Yes <input type="radio"/> No
Default Script	- System Default - Edit

Update Delete Cancel Back to Application List

Trigger Type: Unified CM Telephony Trigger Next

192.168.1.74

Add a New Trigger

Trigger Type* Unified CM Telephony Trigger

Next Cancel

*- indicates required item

Show More...

Directory Number: **2700**

Device Name: **UCCX-TRIGGER**

Description: **UCCX-TRIGGER**

Cal Control Group: **IPCC(2)**

Device Pool: **HQ**

Calling Search Space: **HQ**

Add

Cisco Unified CM Telephony Trigger Configuration

Save Delete Clear Cancel

Status: Ready

Directory Information

Directory Number* 2700

Trigger Information

Language* English (United States) [en_US] Edit

Application Name* CALLCENTER

Device Name* UCCX-TRIGGER

Description* UCCX-TRIGGER

Call Control Group* IPCC(2)

Advanced Trigger Information

Enabled Yes No

Maximum Number Of Sessions Default Unchecked Default value is same as Number of Sessions set on the Application

Idle Timeout (in ms) 5000

Override Media Termination Yes No

CTI Route Point Information

Alerting Name ASCII

Device Pool HQ

Location Hub_None

Directory Number Settings

Partition None

Voice Mail Profile None

Calling Search Space HQ

Calling Search Space for Redirect Default Calling Search Space

Presence Group Standard Presence group

Call Forward and Pickup Settings

Forward Busy Voice Mail Destination Calling Search Space None

Line Settings

Display

External Phone Number Mask

We should now be able to place a test call to **222-2700** from the **HQ PSTN phone**, and the **icd.aef** application should answer by saying, “Thank you for calling. All of our representatives are assigning other callers at this time. Your call is very important to us. Please stay on the line, and your call will be handled in the order it was received.”

NOTE: The requirement of Task 8.2 is that the CALLAGENT application be accessible from the HQ PSTN phone. There is no requirement that the application be accessible from a BR1 or a BR2 phone. Therefore, we don’t need to configure any transcoding resources.

We’re now ready to move onto Task 8.2, where we’ll make a copy of the icd.aef script; modify it; and apply it to the CALLAGENT application.

Task 8.2 UCCX Scripting

Record Custom Prompts

The goal of this task is for a caller to hear the message “Your estimated wait time is **x** minutes,” where **x** is the caller’s wait time in minutes. Therefore, we need to record a couple of custom audio prompts.

- **Your estimated wait time is**
- **minutes**

Cisco Unity Connection can be used to record these prompts.

Call Management > System Call Handlers > Add New

Display Name: **Prompt Recorder**

Save

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

New Call Handler

Call Handler Reset Help

Save

Call Handler

Display Name* Prompt Recorder

Extension

Call Handler Template System Call Handler Template

Save

Fields marked with an asterisk (*) are required.

Play/Record

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Edit Call Handler Basics (Prompt Recorder)

Call Handler Edit Refresh Help

Save Delete Previous Next

Status

Do remember to take backup of report data before renaming the Call Handler display name.

Created Call Handler(s)

Call Handler

Display Name* Prompt Recorder

Creation Time 2015-08-27 11:15:43.485

Phone System PhoneSystem

Active Schedule All Hours View

☒ Use System Default Time Zone

Time Zone (GMT-05:00) America/Kentucky/Louisville

Language ☐ Use System Default Language ☒ Inherit Language from Caller

English(United States)

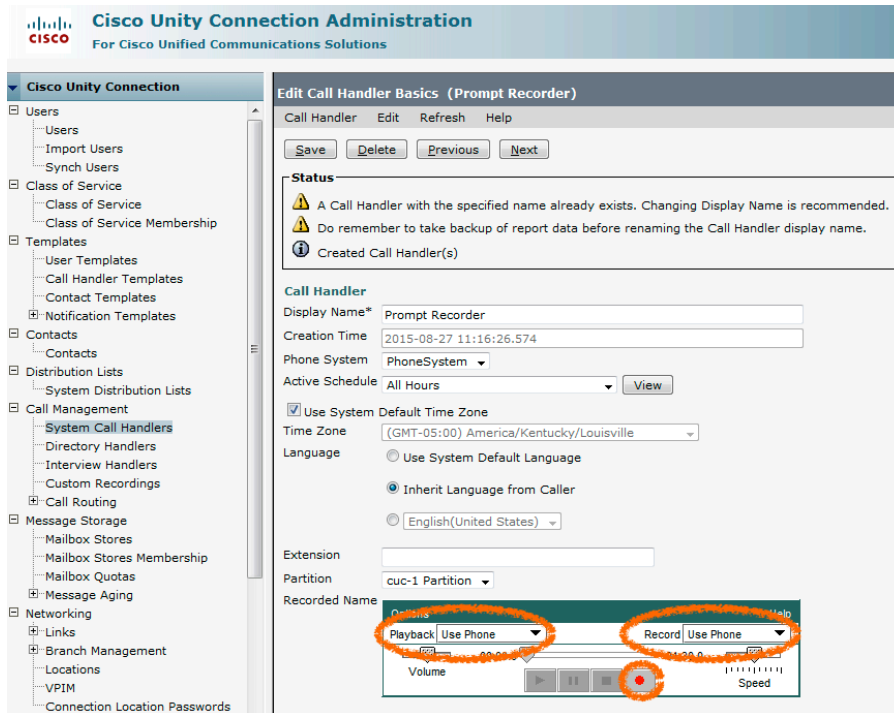
Extension

Partition

Recorded Name Play/Record

Set the **Playback** and **Record** options to **Use Phone**.

Click the red Record button. (If prompted, enter a DN of 2001.)



HQ Phone 1 rings.

Answer it, and after the beep, say, **“Your estimated wait time is.”**

Hang up.

Options > Save Recording As...

Save the audio prompt to your computer’s local hard drive.



Repeat this process for the “minutes” prompt.

Upload Custom Prompts

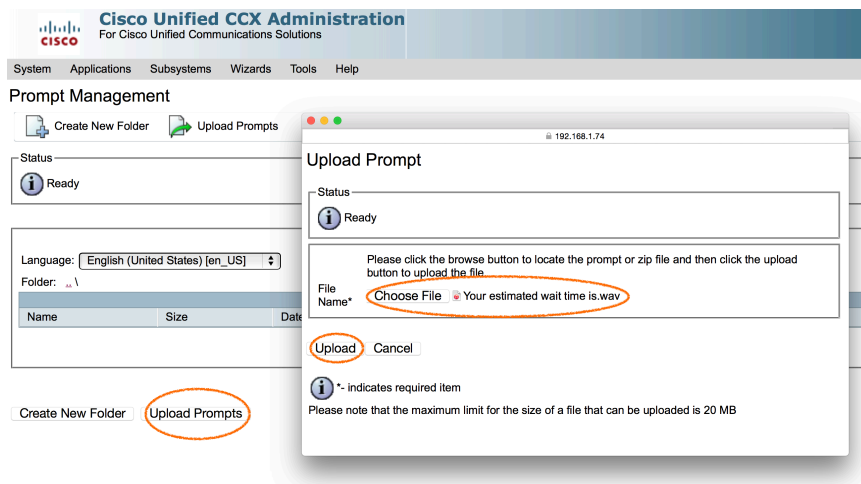
To make the newly reordered prompts available to a script, we need to upload them to UCCX.

(In the UCCX administration screen)

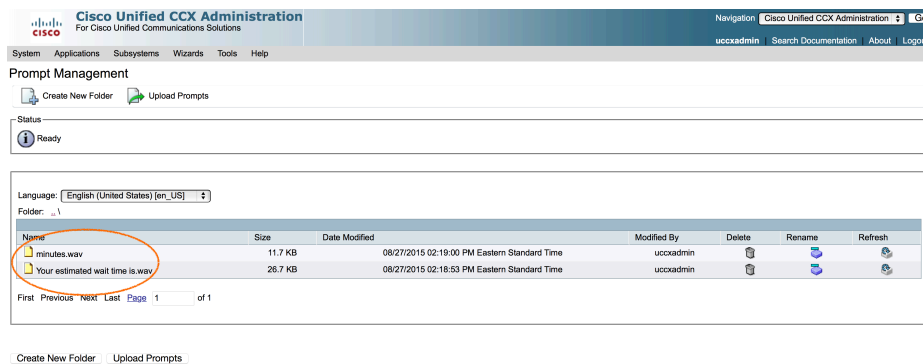
Applications > Prompt Management > en_US > Upload Prompts

Choose File to browse the “Your estimated wait time is” prompt.

Upload



Repeat this process for the “minutes” prompt.

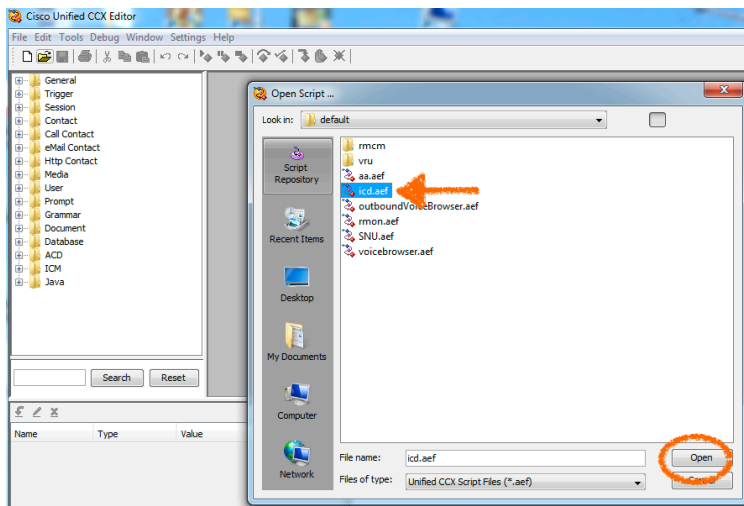


Make a Copy of the icd.aef Script

Open up the UCCX Editor from your desktop (Microsoft Windows) computer, and log into the editor using the **uccxadmin** account.



Open up the **icd.aef** script located in the **C:\Program Files (x86)\wfavvid_902\Scripts\system\default** directory.



Then, save a copy of the script (**File > Save As...**) to the local PC, giving it a name of **TASK_8.2**. We'll be modifying this copy to meet the requirements of this task.

Create a Variable for Each of the Recorded Prompts

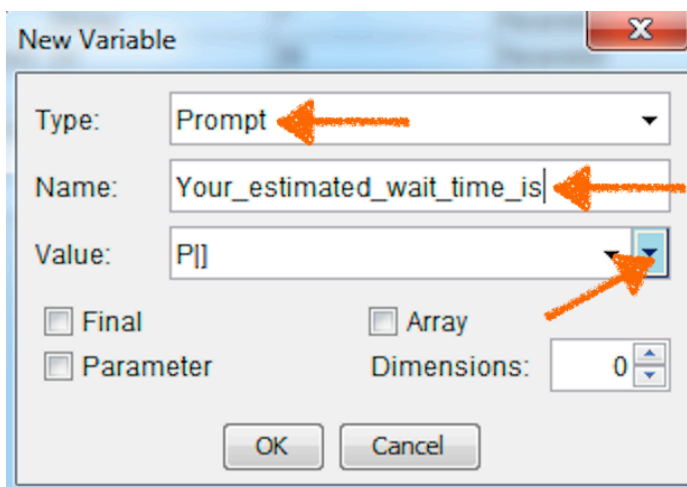
Click the **New Variable** button.

Name	Type	Value	Attributes
CSQ	String	""	Parameter
DelayWhileQueued	int	30	Parameter
QueuePrompt	Prompt	SP[ICD\ICDQueue....	Parameter
SRS_TempResou...	User	null	
WelcomePrompt	Prompt	SP[ICD\ICDWelco...	Parameter
resourceID	String	""	

Type: **Prompt**

Name: **Your_estimated_wait_time_is**

Browse for a specific prompt.



The 'New Variable' dialog box is shown with the following fields and annotations:

- Type:** A dropdown menu set to 'Prompt'. An orange arrow points to the dropdown arrow.
- Name:** A text box containing 'Your_estimated_wait_time_is'. An orange arrow points to the text box.
- Value:** A text box containing 'P[]'. An orange arrow points to the dropdown arrow on the right side of the text box.
- Final:** An unchecked checkbox.
- Array:** An unchecked checkbox.
- Parameter:** An unchecked checkbox.
- Dimensions:** A numeric field set to '0'.
- Buttons:** 'OK' and 'Cancel' buttons at the bottom.

Click the **Prompt** tab.

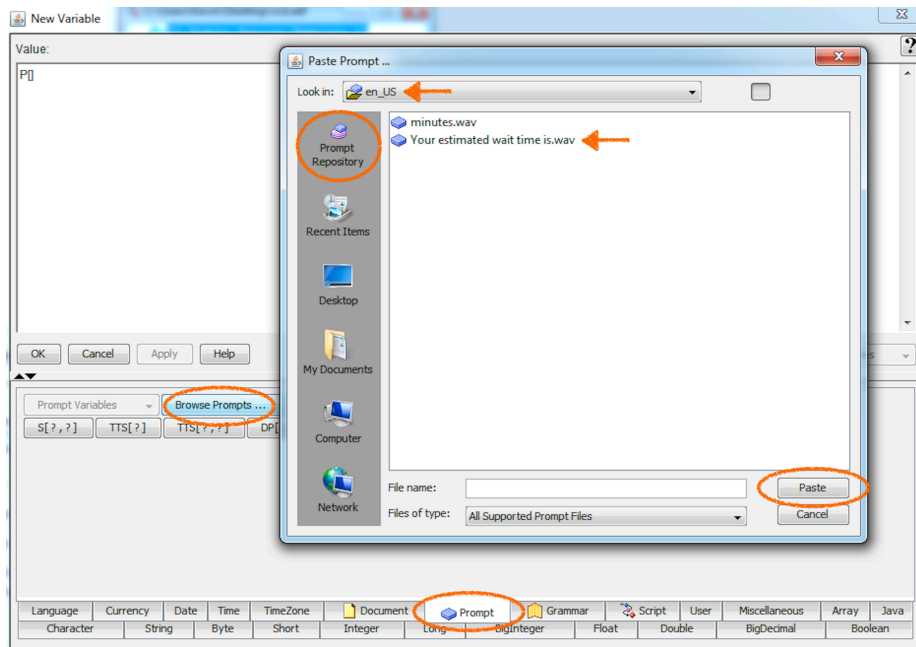
Browse Prompts...

Prompt Repository

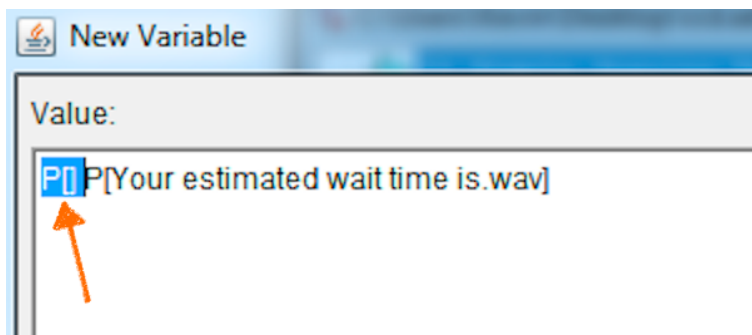
en_US

Your estimated wait time is.wav

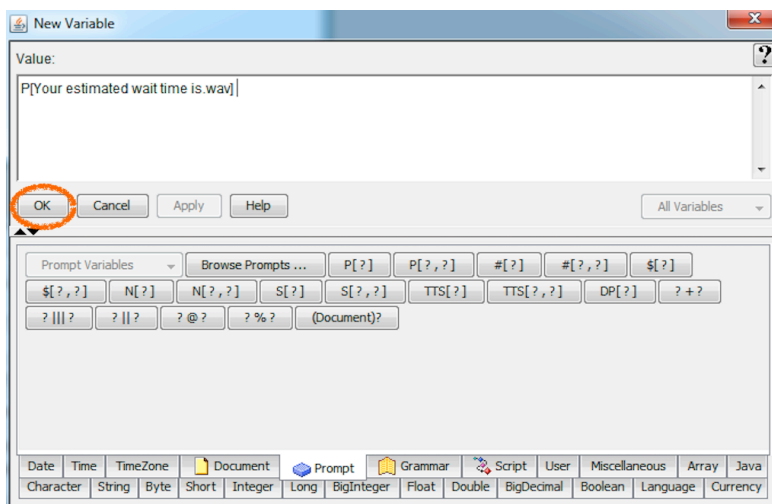
Paste



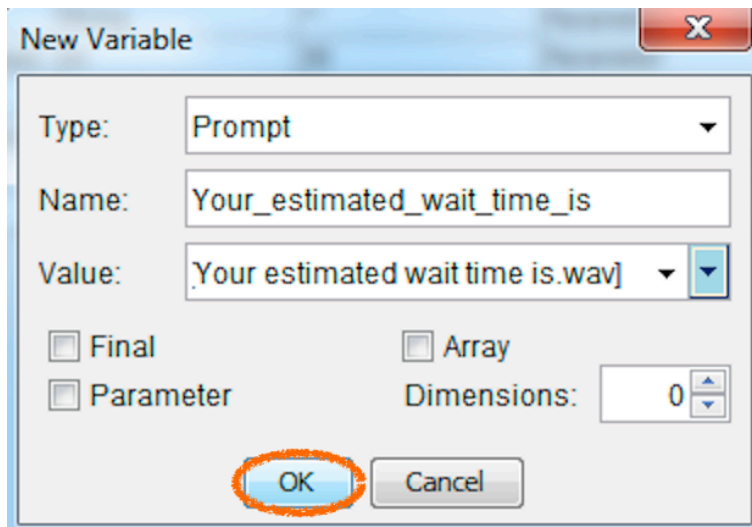
Delete the leading **P[]** from the variable value.



OK



OK



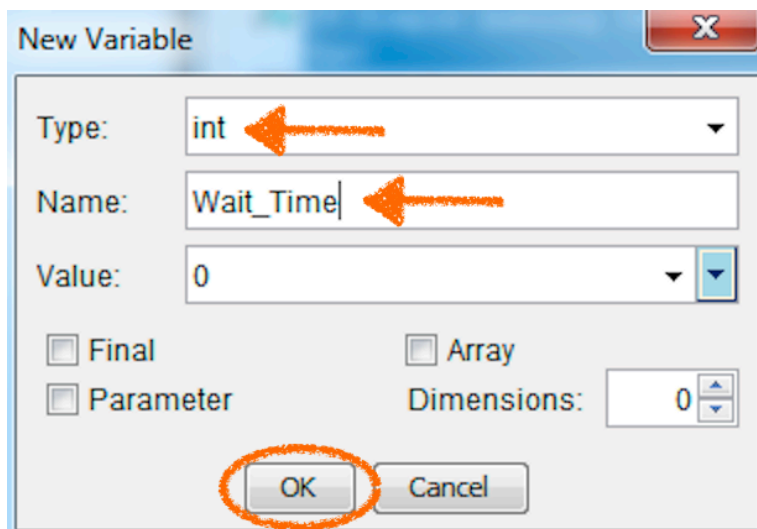
Repeat this process for the “minutes” prompt.

Create a Variable for the Estimated Wait Time, Which is an Integer

Type: **Int**

Name: **Wait_Time**

OK



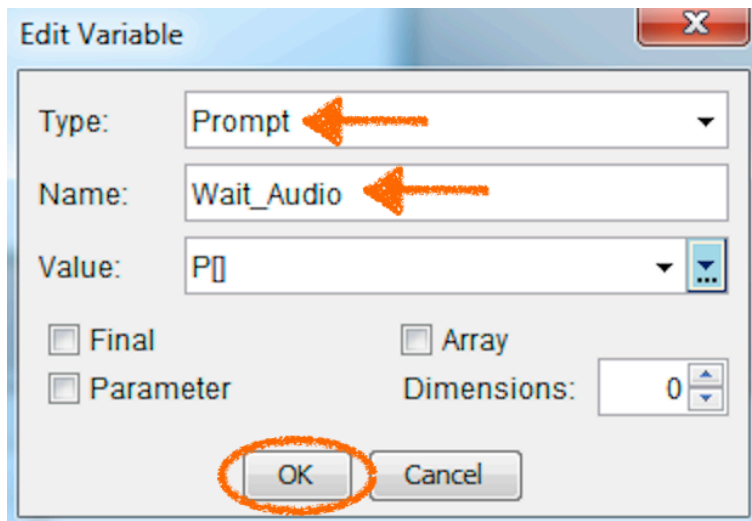
Create a Variable for the Estimated Hold Time Audio

The type of the **Wait_Time** variable is an **integer**. However, its value needs to be played out as an audible prompt. So, we need a new variable to contain the playable prompt. Therefore, we next create a variable to store the playable prompt of estimated wait time in minutes.

Type: **Prompt**

Name: **Wait_Audio**

OK

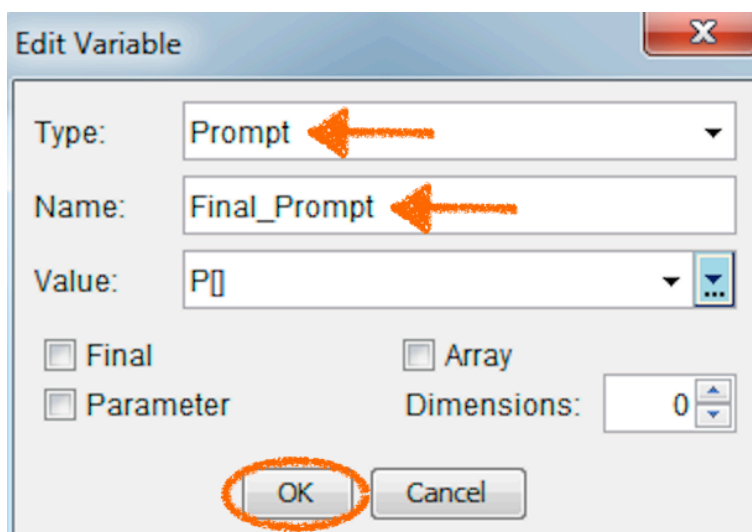


Create a Variable to Store the Final Prompt Played to the User

Type: **Prompt**

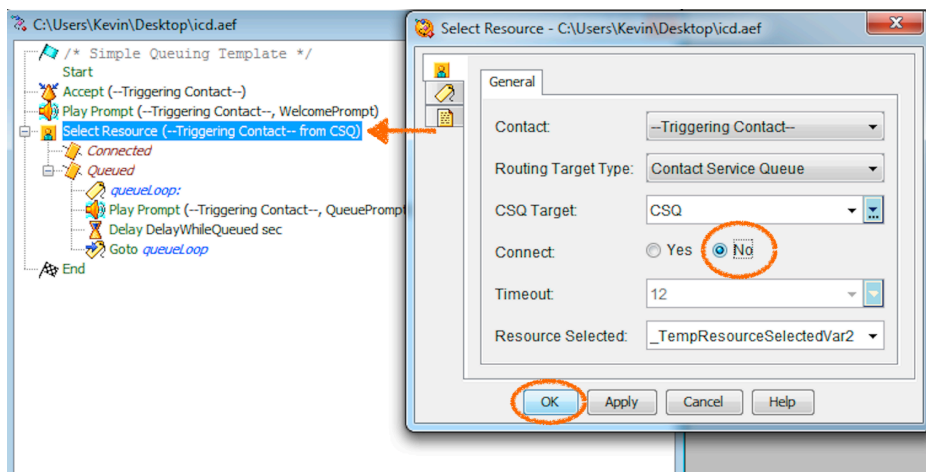
Name: **Final_Prompt**

OK



Edit the Script

Starting with our copy of the default ICD script, we need to tell the **Select Resource** object not to immediately connect the call.



After both the Welcome and Queue prompts play, we need to determine the estimated wait time of the caller in queue. This is done with the **ACD > Get Reporting Statistic** object. Drag and drop the **Get Reporting Statistic** object from the object library (in the left pane) to the second **Play Prompt** object in the script (in the right pane). This will cause the **Get Reporting Statistic** object to appear just below the second **Play Prompt** object. Then, right-click on the **Get Reporting Statistic** object, and select Properties.

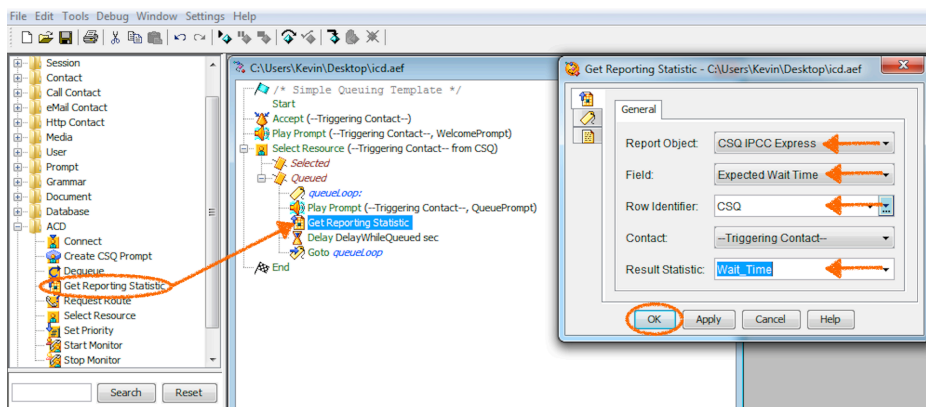
Report Object: **CSQ IPCC Express**

Field: **Expected Wait Time**

Row Identifier: **CSQ**

Result Statistic: **Wait_Time**

OK



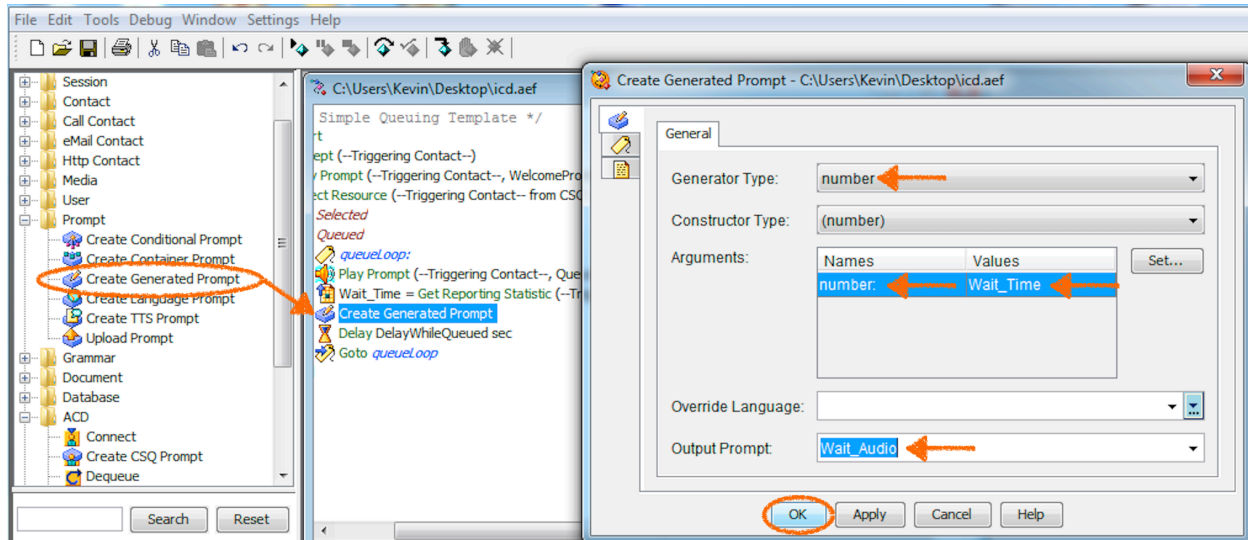
The value of the **Wait_Time** variable is not in the form of a prompt, which can be played out. So, we need to convert the numeric value into an audible prompt. This conversion is done with the **Prompt > Create Generated Prompt** object.

Generator Type: **number**

Arguments: **number: Wait_Time**

Output Prompt: **Wait_Audio**

OK



Now that we have the wait time (in minutes) in audible form, we can combine it with our custom prompts to create the final prompt. This is done with the **Prompt > Create Container Prompt** object.

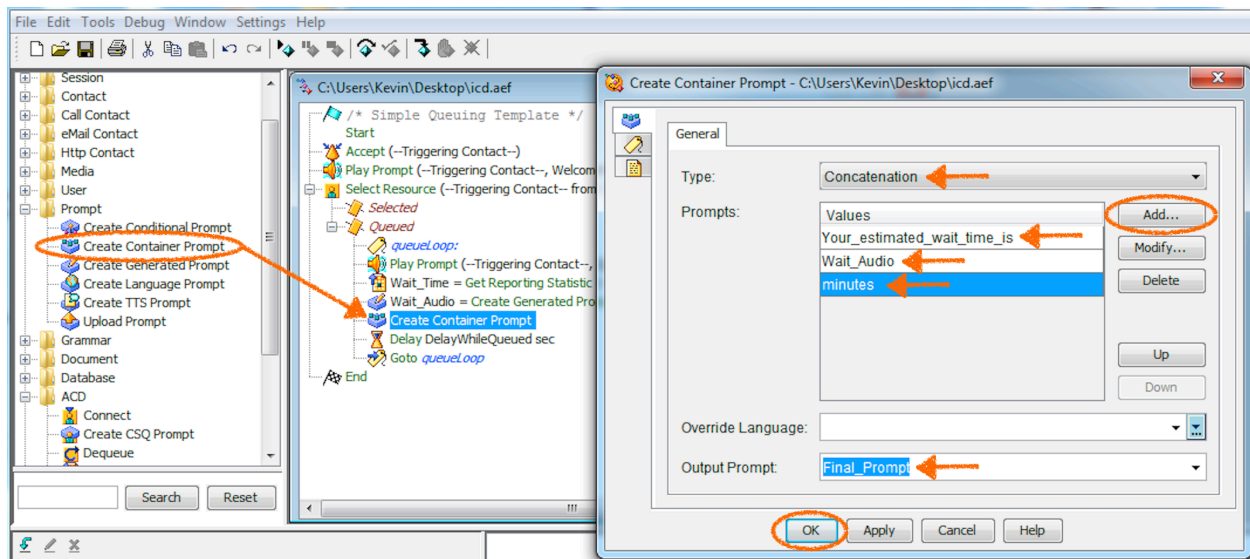
Type: **Concatenation**

Prompts:

- **Your_estimated_wait_time_is**
- **Wait_Audio**
- **Minutes**

Output Prompt: **Final_Prompt**

OK

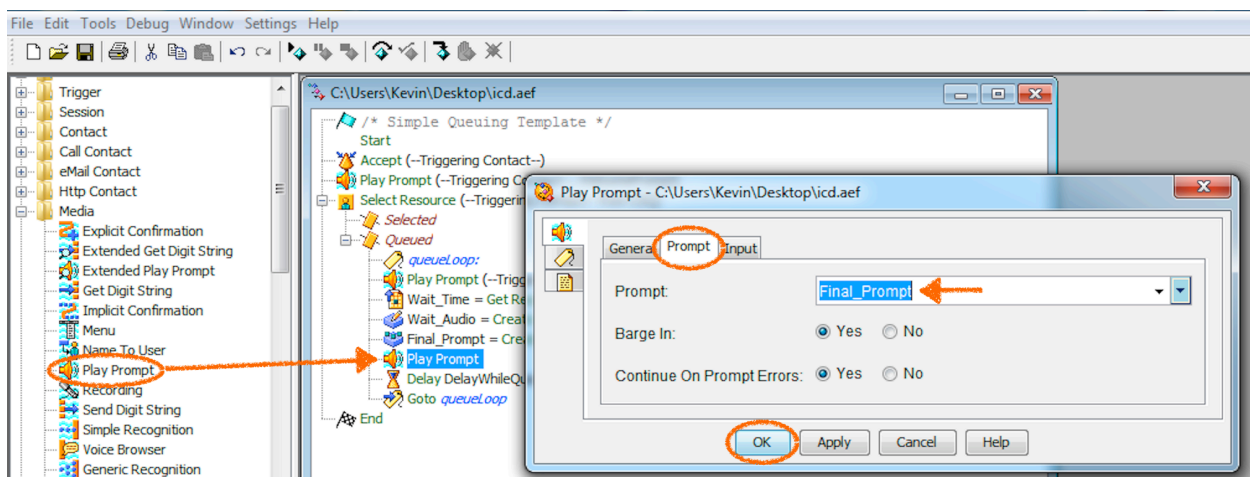


Finally, we need to play out the constructed prompt, with the **Media > Play Prompt** object.

Prompt Tab

Prompt: **Final_Prompt**

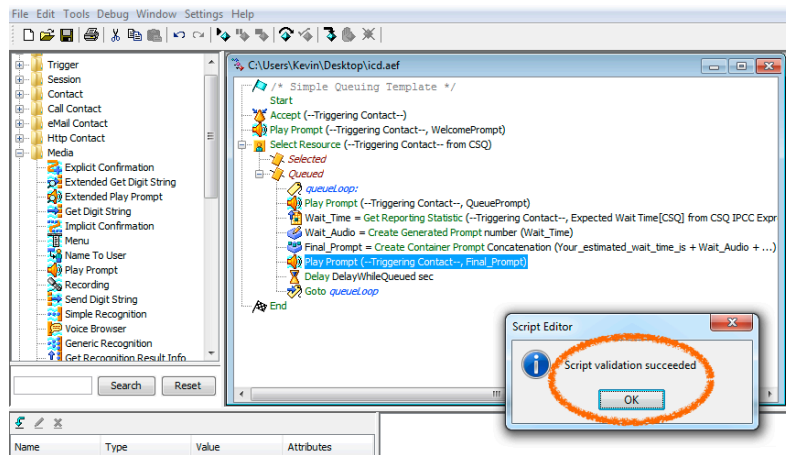
OK



Let's now save our script and do a validation.

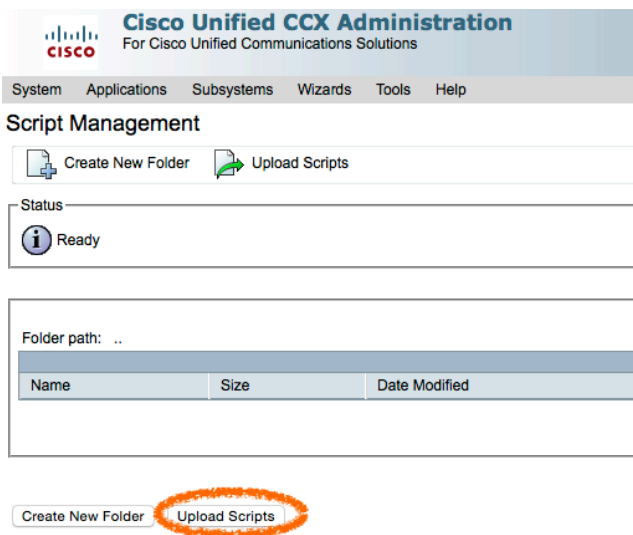
File > Save

Tools > Validate

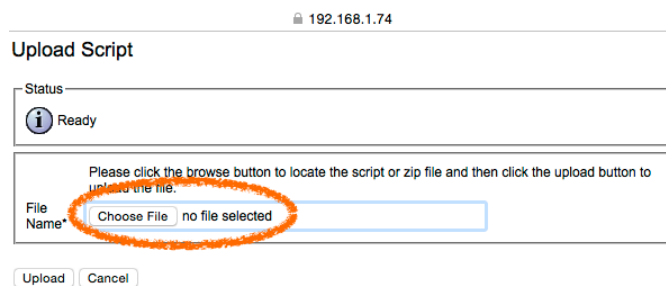


Upload the Script

Applications > Script Management > Upload Scripts



Choose File to browse to the saved script named **TASK_8.2**.



Upload > Return to Script Management

192.168.1.74

Upload Script

Status

i Ready

Please click the browse button to locate the script or zip file and then click the upload button to upload the file.

File Name* TASK_8.2.aef

Update Application to Point to New Script

Applications > Application Management > CALLCENTER

Script: **SCRIPT[TASK_8.2.aef]**

CSQ: **"CSQ"** (checked)

Update

Cisco Unified CCX Administration
For Cisco Unified Communications Solutions

System Applications Subsystems Wizards Tools Help

Cisco Script Application

Status

i Status : Ready

Unified CM Telephony Trigger: 2700
[Add new trigger](#)

Name	CALLCENTER
ID*	0
Maximum Number of Sessions*	2
Script	SCRIPT[TASK_8.2.aef] <input type="button" value="Edit"/>
<input checked="" type="checkbox"/> CSQ	"CSQ" <input type="button" value="Edit"/>
<input type="checkbox"/> DelayWhileQueued	
<input type="checkbox"/> WelcomePrompt	ICD/ICDWelcome.wav <input type="button" value="Show Prompts"/>
<input type="checkbox"/> QueuePrompt	ICD/ICDQueue.wav <input type="button" value="Show Prompts"/>
Description	CALLCENTER
Enabled	<input checked="" type="radio"/> Yes <input type="radio"/> No
Default Script	- System Default - <input type="button" value="Edit"/>

i *- indicates required item

Verification

From a **PSTN phone**, call **222-2700**. Wait for the call to be queued, and listen to the message. Note that the wait time will initially be a **-1**, because we haven't actually answered any calls from the IP Phone Agents.

Module 15: Doing EVERYTHING on the Cisco IM and Presence Servers

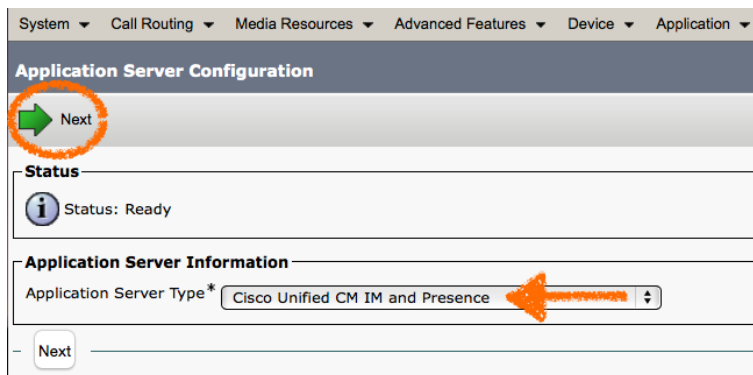
Task 9.1 IM and Presence Integration with CUCM

Define the IM & Presence Server in CUCM

System > Application Server > Add New

Application Server Type: Cisco Unified CM IM and Presence

Next



System > Call Routing > Media Resources > Advanced Features > Device > Application

Application Server Configuration

Next

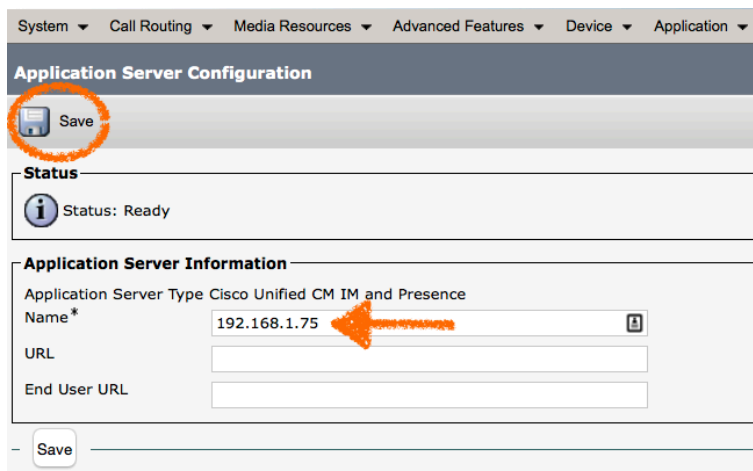
Status
Status: Ready

Application Server Information
Application Server Type* Cisco Unified CM IM and Presence

Next

Name: 192.168.1.75

Save



System > Call Routing > Media Resources > Advanced Features > Device > Application

Application Server Configuration

Save

Status
Status: Ready

Application Server Information
Application Server Type Cisco Unified CM IM and Presence
Name* 192.168.1.75
URL
End User URL

Save

Create a SIP Trunk to the IM & Presence Server

System > Security > SIP Trunk Security Profile > Find > Non Secure SIP Trunk Profile

Accept presence subscription: (Checked)

Accept out-of-dialog refer**: (Checked)

Accept unsolicited notification: (Checked)

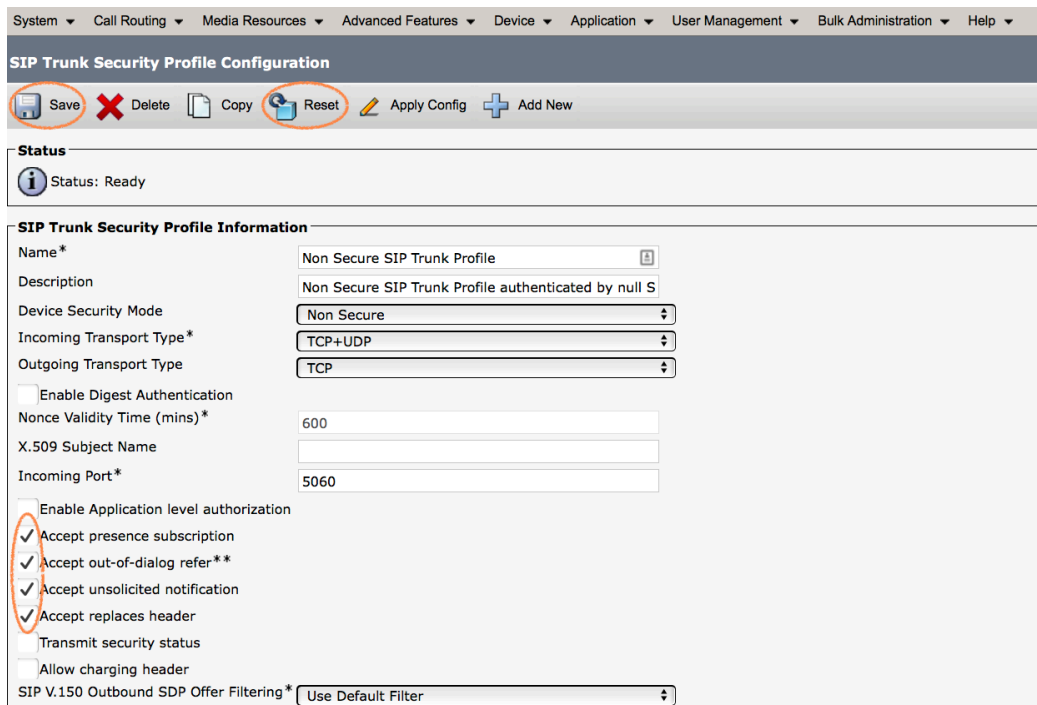
Accept replaces header: (*Checked*)

Save

Reset







Reset

Close




System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾


SIP Trunk Security Profile Configuration

 Save  Delete  Copy  Reset  Apply Config  Add New

Status

 Status: Ready

SIP Trunk Security Profile Information

Name* Non Secure SIP Trunk Profile 

Description Non Secure SIP Trunk Profile authenticated by null S

Device Security Mode Non Secure ▾

Incoming Transport Type* TCP+UDP ▾

Outgoing Transport Type TCP ▾

☐ Enable Digest Authentication

Nonce Validity Time (mins)* 600

X.509 Subject Name

Incoming Port* 5060

☐ Enable Application level authorization

☒ Accept presence subscription

☒ Accept out-of-dialog refer**

☒ Accept unsolicited notification

☒ Accept replaces header

☐ Transmit security status

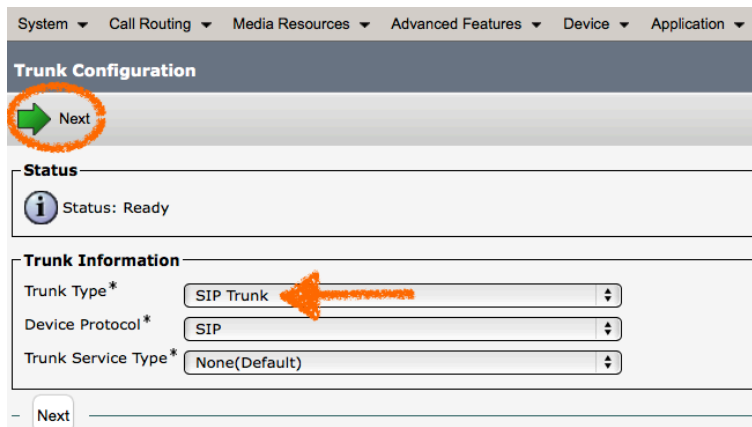
☐ Allow charging header

SIP V.150 Outbound SDP Offer Filtering* Use Default Filter ▾

Device > Trunk > Add New


Trunk Type: SIP Trunk

Next




System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾


Trunk Configuration

 Next

Status


 Status: Ready

Trunk Information

Trunk Type* SIP Trunk 

Device Protocol* SIP ▾

Trunk Service Type* None(Default) ▾

 Next

Device Name: **PRESENCE**

Device Pool: **HQ**

Destination Address: **192.168.1.75**

MTP Preferred Originating Codec* 711ulaw

BLF Presence Group* Standard Presence group

SIP Trunk Security Profile* Non Secure SIP Trunk Profile

Rerouting Calling Search Space < None >

Out-Of-Dialog Refer Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile* Standard SIP Profile

DTMF Signaling Method* No Preference

Normalization Script

Normalization Script < None >

Enable Trace

Parameter Name	Parameter Value
1	

Geolocation Configuration

Geolocation < None >

Geolocation Filter < None >

Send Geolocation Information

Save Delete Reset Add New

Define a Phone for the Jabber Client

Device > Phone > Add New

Phone Type: **Cisco Unified Client Services Framework**

Next

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾

Add a New Phone

Next

Status

Status: Ready

Select the type of phone you would like to create

Phone Type* Cisco Unified Client Services Framework

Next

Device name: **CSFhqphone2**

Device Pool: **HQ**

Phone Button Template: **Standard Client Service Framework**

Calling Search Space: **HQ**

Owner User ID: **hqphone 2**

Device Security Profile: **Cisco Unified Client Services Framework – Standard SIP Non-Secure**

SIP Profile: **Standard SIP Profile**


Save

OK

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Phone Configuration

 Save

Status
Status: Ready

Phone Type
Product Type: Cisco Unified Client Services Framework
Device Protocol: SIP

Device Information
☒ Device is trusted
Device Name * CSF794phone2
Description
Device Pool * HQ
Common Device Configuration < None >
Phone Button Template * Standard Client Services Framework
Common Phone Profile * Standard Common Phone Profile
Calling Search Space HQ
AAR Calling Search Space < None >
Media Resource Group List < None >
User Hold MOH Audio Source < None >
Network Hold MOH Audio Source < None >
Location * Hub_None
AAR Group < None >
User Locale < None >
Network Locale < None >
Built In Bridge * Default
Device Mobility Mode * Default
Owner ☒ User / Anonymous (Public/Shared Space)
Owner User ID * hqphone2
Mobility User ID < None >
Primary Phone < None >
Use Trusted Relay Point * Default
Always Use Prime Line * Default
Always Use Prime Line for Voice Message * Default
Geolocation < None >
☐ Ignore Presentation Indicators (Internal calls only)
☒ Allow Control of Device from CTI
☒ Logged Into Hunt Group
☐ Remote Device
☐ Require off-premise location

Number Presentation Transformation
Caller ID For Calls From This Phone
Calling Party Transformation CSS < None >
☒ Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)
Remote Number
Calling Party Transformation CSS < None >
☒ Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)





Protocol Specific Information
Packet Capture Mode * None
Packet Capture Duration 0
BLF Presence Group * Standard Presence group
SIP Dial Rules < None >
MTP Preferred Originating Codec * 711ulaw
Device Security Profile * Cisco Unified Client Services Framework - Standard
Rerouting Calling Search Space < None >
SUBSCRIBE Calling Search Space < None >
SIP Profile * Standard SIP Profile

Line [1] – Add a new DN

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Phone Configuration

 Save  Delete  Copy  Reset  Apply Config  Add New

Status
Add successful

Association Information
Modify Button Item
Line [1] - Add a new DN

Phone Type
Product Type: Cisco Unified Client Services Framework
Device Protocol: SIP

Device Information

Since the DN of 2002 in the INTERNAL partition has already been defined, we can just specify that DN and partition to have to other DN values inherited.

Directory Number: **2002**
Route Partition: **INTERNAL**
Save

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Directory Number Configuration

Related Links: Configure Device (CSFhqphone2) | Go

Save | Delete | Reset | Apply Config | Add New

Status: Ready

Directory Number Information

Directory Number*: 2002

Route Partition: INTERNAL

Description: HQ Phone 2

Alerting Name: HQ Phone 2

ASCII Alerting Name: HQ Phone 2

☒ Allow Control of Device from CTI

Associated Devices: SEPECC8821098E9 CSFhqphone2

Edit Device
Edit Line Appearance

Dissociate Devices

Associate End Users

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Directory Number Configuration

Related Links: Configure Device (CSFhqphone2) | Go

Save | Delete | Reset | Apply Config | Add New

Recording Option*: Call Recording Disabled

Recording Profile: < None >

Monitoring Calling Search Space: < None >

Propagate Selected

Multiple Call/Call Waiting Settings on Device CSFhqphone2

Note: The range to select the Max Number of calls is: 1-6

Maximum Number of Calls*: 6

Busy Trigger*: 2 (Less than or equal to Max. Calls)

Forwarded Call Information Display on Device CSFhqphone2

☒ Caller Name

☐ Caller Number

☐ Redirected Number

☒ Dialed Number

Users Associated with Line

Associate End Users

Save | Delete | Reset | Apply Config | Add New

Find
hqphone2 (checked)
Add Selected

192.168.1.71

Find and List Users

☐ Select All
 ☐ Clear All
 ☒ Add Selected
 ☐ Close

Status

3 records found

User (1 - 3 of 3) Rows per Page 50

Find User where

<input type="checkbox"/>	User ID ^	First Name	Last Name	Department
<input type="checkbox"/>	hqphone1	HQ	Phone 1	
<input checked="" type="checkbox"/>	hqphone2	HQ	Phone 2	
<input type="checkbox"/>	uccxadmin	UCCX	Admin	

Save
Go

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

administrator | [Search Documentation](#) | [About](#) | [Logout](#)

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Directory Number Configuration

Related Links: [Configure Device \(CSFhqphone2\)](#)

Recording Profile

Monitoring Calling Search Space

Multiple Call/Call Waiting Settings on Device CSFhqphone2

Note: The range to select the Max Number of calls is: 1-6

Maximum Number of Calls*

Busy Trigger* (Less than or equal to Max. Calls)

Forwarded Call Information Display on Device CSFhqphone2

☒ Caller Name
☐ Caller Number
☐ Redirected Number
☒ Dialed Number

Users Associated with Line

<input type="checkbox"/>	Full Name	User ID	Permission
<input type="checkbox"/>	Phone 2, HQ	hqphone2	

Reset
Reset
Close

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Phone Configuration | Related Links: Back To Find/List

Save | Delete | Copy | **Reset** | Apply Config | Add New

Status
Status: Ready

Association Information Modify Button Items 1 Line [1] - 2002 in INTERNAL ----- Unassigned Associated Items ----- 2 Line [2] - Add a new DN	Phone Type Product Type: Cisco Unified Client Services Framework Device Protocol: SIP Device Information Registration: Unknown IP Address: Unknown <input checked="" type="checkbox"/> Device is Active <input checked="" type="checkbox"/> Device is trusted
--	--

Add UC Services

User Management > User Settings > UC Service > Add New

User Service Type: CTI

Next

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System | Call Routing | Media Resources | Advanced Features | Device | Application

UC Service Configuration

Next

Status
Status: Ready

Add a UC Service

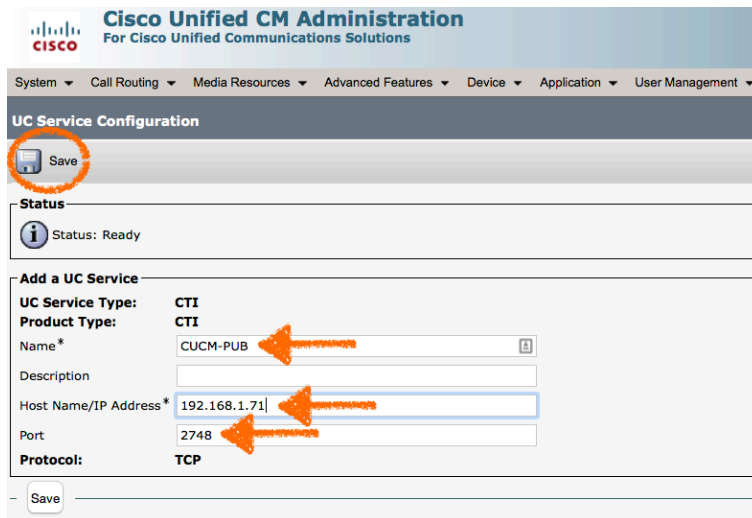
UC Service Type: **CTI**

Next

Name: **CUCM-PUB**

Host Name/IP Address: **192.168.1.71**


Save



Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

UC Service Configuration

 Save

Status
Status: Ready

Add a UC Service

UC Service Type: CTI
Product Type: CTI

Name* CUCM-PUB

Description

Host Name/IP Address* 192.168.1.71

Port 2748

Protocol: TCP

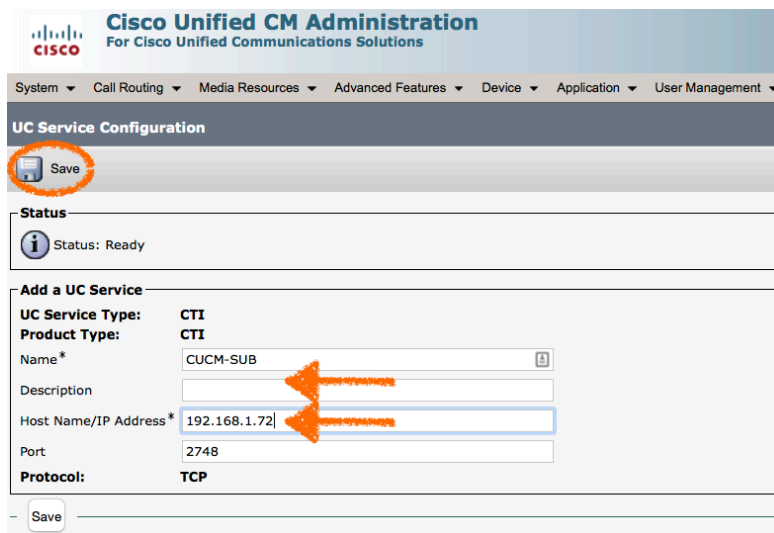
Save

Copy

Name: **CUCM-SUB**

Host Name/IP Address: **192.168.1.72**


Save



Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

UC Service Configuration

 Save

Status
Status: Ready

Add a UC Service

UC Service Type: CTI
Product Type: CTI

Name* CUCM-SUB

Description

Host Name/IP Address* 192.168.1.72

Port 2748

Protocol: TCP

Save

Add New

UC Service Type: **IM and Presence**

Next

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾

UC Service Configuration

➔ Next

Status
Status: Ready

Add a UC Service
UC Service Type: IM and Presence

Next

Name: **CUPS**

Host Name/IP Address: **192.168.1.75**

Save

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾

UC Service Configuration

Save

Status
Status: Ready

Add a UC Service
UC Service Type: IM and Presence
Product Type*: Unified CM (IM and Presence)
Name*: CUPS
Description:
Host Name/IP Address*: 192.168.1.75

Save

Add a Service Profile

Now, we want to add a service profile that references the previously created UC services.

User Management > User Settings > Service Profile > Add New


Name: **PRESENCE**

IM and Presence Profile – Primary: **CUPS**

CTI Profile – Primary: **CUCM-SUB**


CTI Profile – Secondary: **CUCM-PUB**


Save


Cisco Unified CM Administration
 For Cisco Unified Communications Solutions

System ▾ | Call Routing ▾ | Media Resources ▾ | Advanced Features ▾ | Device ▾ | Application ▾ | User Management ▾ | Bulk Administration ▾ | Help ▾

Service Profile Configuration

 Save

Status
 Status: Ready

Name* PRESENCE
 Description
☐ Make this the default service profile for the system

Voicemail Profile
 Primary <None>
 Secondary <None>
 Tertiary <None>
 Credentials source for voicemail service* Not set


MailStore Profile
 Primary <None>
 Secondary <None>
 Tertiary <None>
 Inbox Folder* INBOX
 Trash Folder* Deleted Items
 Polling Interval (in seconds)* 60
☒ Allow dual folder mode

Conferencing Profile
 Primary <None>
 Secondary <None>
 Tertiary <None>
 Server Certificate Verification Any
 Credentials source for web conference service* Not set

Directory Profile
 Primary <None>
 Secondary <None>
 Tertiary <None>
☒ Use UDS for Contact Resolution
☒ Use Logged On User Credential
 Username
 Password
 Search Base 1
 Search Base 2
 Search Base 3
☒ Recursive Search on All Search Bases
 Search Timeout (seconds)* 5
 Base Filter (Only used for Advance Directory)
 Predictive Search Filter (Only used for Advance Directory)

IM and Presence Profile
 Primary CUPS
 Secondary <None>
 Tertiary <None>

CTI Profile
 Primary CUCM-SUB
 Secondary CUCM-PUB
 Tertiary <None>

 Save

Modify End User

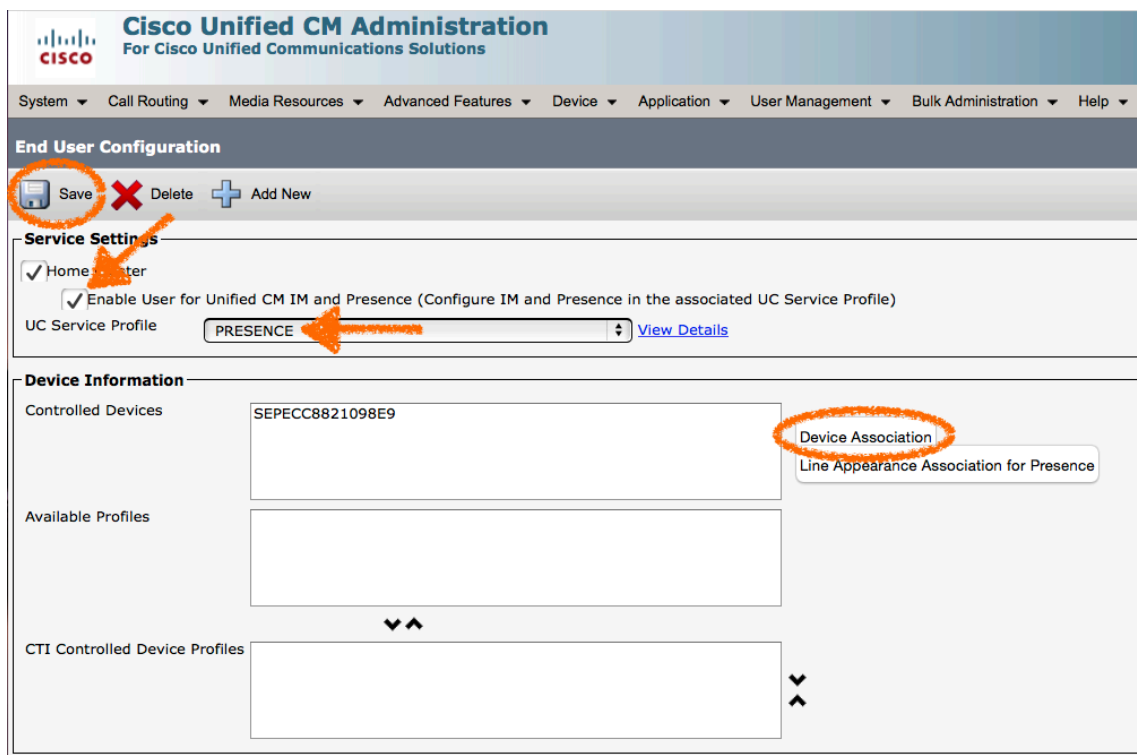
User Management > End User > Find > hqphone2

Enable User for Unified CM IM and Presence: *(checked)*

UC Service Profile: **PRESENCE**

Save

Device Association



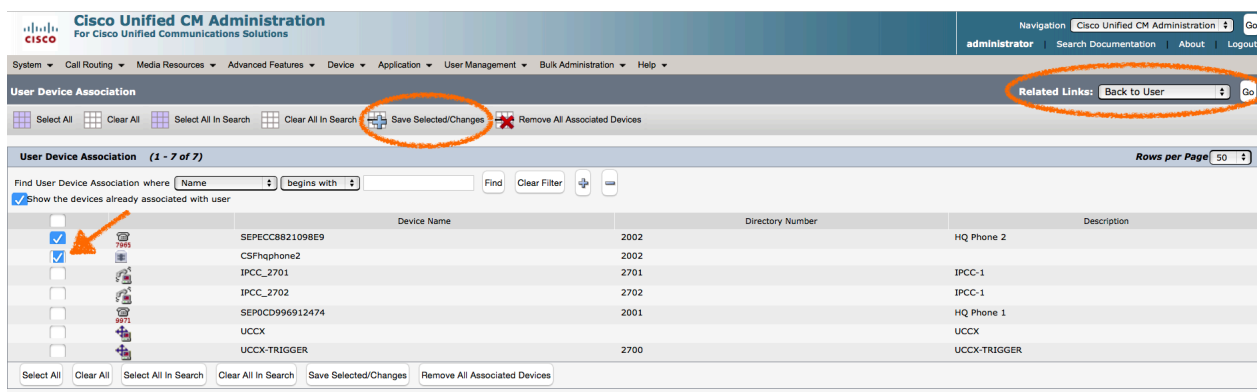
The screenshot shows the 'End User Configuration' page in Cisco Unified CM Administration. The 'Service Settings' section has 'Home' checked and 'Enable User for Unified CM IM and Presence' checked. The 'UC Service Profile' is set to 'PRESENCE'. The 'Device Information' section shows 'Controlled Devices' as 'SEPECC8821098E9'. A callout box points to the 'Device Association' link, which is labeled 'Line Appearance Association for Presence'. The 'Save' button is circled in orange.

Find

CSFhqphone2 (checked)

Save Selected/Changes

Go



The screenshot shows the 'User Device Association' page in Cisco Unified CM Administration. The 'Save Selected/Changes' button is circled in orange. A callout box points to the 'Related Links: Back to User' link, which is also circled in orange. The table below lists the devices associated with the user.



	Device Name	Directory Number	Description
<input checked="" type="checkbox"/>	SEPECC8821098E9	2002	HQ Phone 2
<input checked="" type="checkbox"/>	CSFhqphone2	2002	
<input type="checkbox"/>	IPCC_2701	2701	IPCC-1
<input type="checkbox"/>	IPCC_2702	2702	IPCC-1
<input type="checkbox"/>	SEP0CD996912474	2001	HQ Phone 1
<input type="checkbox"/>	UCCX		UCCX
<input type="checkbox"/>	UCCX-TRIGGER	2700	UCCX-TRIGGER

Add to Access Control Group

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

End User Configuration

Save  Delete  Add New

Maximum Wait Time for Desk Pickup* 10000

Remote Destination Limit* 4

Remote Destination Profiles

[View Details](#)

Multilevel Precedence and Preemption Authorization

MLPP User Identification Number

MLPP Password

Confirm MLPP Password

MLPP Precedence Authorization Level Routine

CAPF Information

Associated CAPF Profiles

[View Details](#)

Permissions Information

Groups

Standard CCM End Users

Standard CTI Enabled

[View Details](#)

Roles

Standard CCM End Users

Standard CCMUSER Administration

Standard CTI Enabled

[View Details](#)

[Add to Access Control Group](#)

[Remove from Access Control Group](#)

Save Delete Add New

Find



Standard CTI Allow Control of All Devices (*checked*)

Add Selected

Close

192.168.1.71

Find and List Access Control Groups

Select All Clear All  Add Selected  Close

Status

22 records found

Access Control Group (1 - 22 of 22) Rows per Page 50

Find Access Control Group where Name begins with Find Clear Filter

	Name ^
<input type="checkbox"/>	Standard Audit Users
<input type="checkbox"/>	Standard CAR Admin Users
<input type="checkbox"/>	Standard CCM Admin Users
<input type="checkbox"/>	Standard CCM Gateway Administration
<input type="checkbox"/>	Standard CCM Phone Administration
<input type="checkbox"/>	Standard CCM Read Only
<input type="checkbox"/>	Standard CCM Server Maintenance
<input type="checkbox"/>	Standard CCM Server Monitoring
<input type="checkbox"/>	Standard CCM Super Users
<input type="checkbox"/>	Standard CTI Allow Call Monitoring
<input type="checkbox"/>	Standard CTI Allow Call Park Monitoring
<input type="checkbox"/>	Standard CTI Allow Call Recording
<input type="checkbox"/>	Standard CTI Allow Calling Number Modification
<input checked="" type="checkbox"/>	Standard CTI Allow Control of All Devices
<input type="checkbox"/>	Standard CTI Allow Control of Phones supporting Connected Xfer and conf
<input type="checkbox"/>	Standard CTI Allow Control of Phones supporting Rollover Mode
<input type="checkbox"/>	Standard CTI Allow Reception of SRTP Key Material
<input type="checkbox"/>	Standard CTI Secure Connection
<input type="checkbox"/>	Standard EM Authentication Proxy Rights
<input type="checkbox"/>	Standard Packet Sniffer Users
<input type="checkbox"/>	Standard RealtimeAndTraceCollection
<input type="checkbox"/>	Standard TabSync User

Select All Clear All Add Selected Close

Save

Permissions Information

Groups	Standard CCM End Users Standard CTI Enabled Standard CTI Allow Control of All Devices	View Details
Roles	Standard CCM End Users Standard CCMUSER Administration Standard CTI Enabled	View Details

[Add to Access Control Group](#)
[Remove from Access Control Group](#)

[Save](#) [Delete](#) [Add New](#)

Permissions Information

Groups	Standard CTI Allow Control of All Devices Standard CCM End Users Standard CTI Enabled	View Details
Roles	Standard CCM End Users Standard CCMUSER Administration Standard CTI Allow Control of All Devices Standard CTI Enabled	View Details

[Add to Access Control Group](#)
[Remove from Access Control Group](#)

[Save](#) [Delete](#) [Add New](#)

Complete the Initial Setup of the IM & Presence Server

(Log into the IM and Presence Server)

Hostname: **CUCM9-PUB1**

IP Address: **192.168.1.71**

Next

Cisco Unified CM IM and Presence Administration

administrator | Help | Logout | About

Post Install Setup

The final install steps for this Cisco Unified Call Manager IM and Presence Service server need to be completed. The following screens will walk you through this process.

The Cisco Unified Communications Manager Publisher is the node that the IM and Presence Service server will communicate with to receive end user updates.

Cisco Unified Communications Manager Publisher configuration:

Hostname*	CUCM9-PUB1
IP Address	192.168.1.71

[Back](#) [Next](#)


*- indicates required item.

AXL User: **administrator**

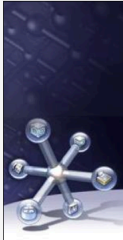
AXL Password: **kwtrain01**

Confirm Password: **kwtrain01**

Next

**Cisco Unified CM IM and Presence Administration**
For Cisco Unified Communications Solutions

administrator | Help | Logout | About




Post Install Setup

AXL is the API that IM and Presence Service uses to communicate with the CUCM Publisher. AXL login information for the CUCM Publisher is required.


AXL Configuration Information:

CUCM Publisher IP Address	192.168.1.71
AXL User*	administrator
AXL Password*	*****
Confirm Password*	*****

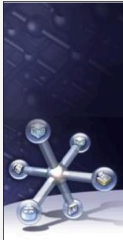
[Back](#) [Next](#)

 *- indicates required item.

Security Password: **kwtrain01**
Confirm Password: **kwtrain01**
Next

**Cisco Unified CM IM and Presence Administration**
For Cisco Unified Communications Solutions

administrator | Help | Logout | About




Post Install Setup

The IPsec Security password is used to secure communication among CUCM and IM and Presence Service nodes. This password must match the security password as configured on the CUCM Publisher node.


Security Password configuration:

Security Password*	*****
Confirm Password*	*****

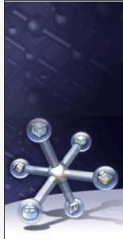
[Back](#) [Next](#)

 *- indicates required item.

Confirm

**Cisco Unified CM IM and Presence Administration**
For Cisco Unified Communications Solutions

administrator | Help | Logout | About




Post Install Setup

Please verify the information below and click Confirm. If any information is incorrect, please go back and correct it before proceeding.

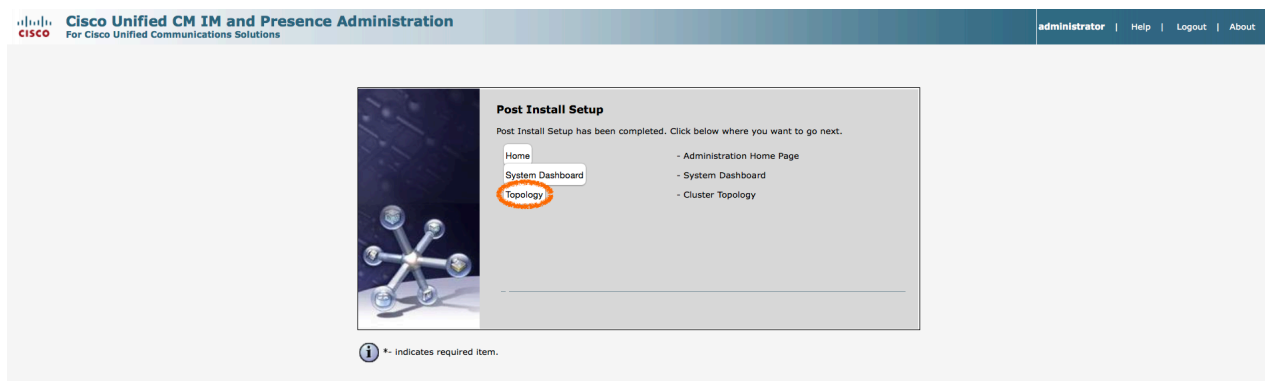
Hostname	CUCM9-PUB1 *
IP Address	192.168.1.71
AXL User	administrator

(* CUCM Hostname does not resolve to a valid IP Address. This is expected behavior if the network does not have DNS.)

[Back](#) [Confirm](#)

 *- indicates required item.

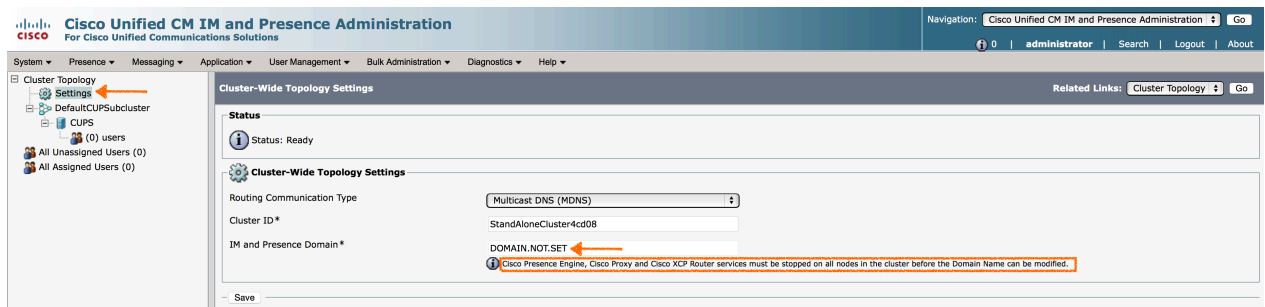
Topology



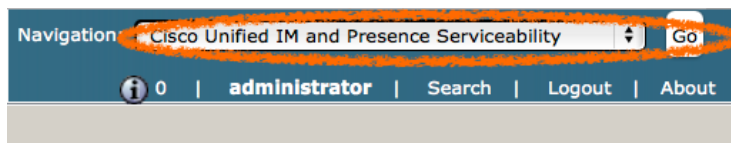
Configure Domain

Settings

Note that the IM and Presence Domain is **DOMAIN.NOT.SET**. Also, notice that this is not currently an editable field. We're told three services need to be stopped before making this change. After a fresh install, the **Cisco Presence Engine** and **Cisco SIP Proxy** services are probably already stopped (and not even enabled). However, the **Cisco XCP Router** service will probably be running. So, we can just stop the **Cisco XCP Router** service; change our domain name (to **cisco.local**); restart the **Cisco XCP Router** service; and then enable the remaining services.



Navigation: **Cisco Unified IM and Presence Serviceability**
Go



Tools > Control Center – Network Services

Cisco XCP Router (selected)

Stop

OK

IM and Presence Services				
	Service Name	Status:	Start Time	Up Time
<input type="radio"/>	Cisco Login Datastore	Running	Wed Sep 2 03:33:17 2015	0 days 08:52:32
<input type="radio"/>	Cisco Route Datastore	Running	Wed Sep 2 03:33:18 2015	0 days 08:52:31
<input type="radio"/>	Cisco Config Agent	Running	Wed Sep 2 11:37:24 2015	0 days 00:48:25
<input type="radio"/>	Cisco OAM Agent	Running	Wed Sep 2 11:37:26 2015	0 days 00:48:23
<input type="radio"/>	Cisco Client Profile Agent	Running	Wed Sep 2 11:37:30 2015	0 days 00:48:19
<input type="radio"/>	Cisco Intercluster Sync Agent	Running	Wed Sep 2 11:37:13 2015	0 days 00:48:36
<input type="radio"/>	Cisco XCP Config Manager	Running	Wed Sep 2 11:37:11 2015	0 days 00:48:38
<input checked="" type="radio"/>	Cisco XCP Router	Running	Wed Sep 2 12:19:07 2015	0 days 00:06:42
<input type="radio"/>	Cisco Server Recovery Manager	Running	Wed Sep 2 11:37:25 2015	0 days 00:48:24
<input type="radio"/>	Cisco Replication Watcher	Running	Wed Sep 2 11:37:10 2015	0 days 00:48:39
<input type="radio"/>	Cisco Presence Datastore	Running	Wed Sep 2 11:37:19 2015	0 days 00:48:30
<input type="radio"/>	Cisco SIP Registration Datastore	Running	Wed Sep 2 11:37:20 2015	0 days 00:48:29

DB Services				
	Service Name	Status:	Start Time	Up Time
<input type="radio"/>	Cisco Database Layer Monitor	Running	Wed Sep 2 03:33:16 2015	0 days 08:52:33

SOAP Services				
	Service Name	Status:	Start Time	Up Time
<input type="radio"/>	SOAP -Real-Time Service APIs	Running	Wed Sep 2 11:37:07 2015	0 days 00:48:42
<input type="radio"/>	SOAP -Performance Monitoring APIs	Running	Wed Sep 2 11:37:08 2015	0 days 00:48:41
<input type="radio"/>	SOAP -Log Collection APIs	Running	Wed Sep 2 11:37:09 2015	0 days 00:48:40

Admin Services				
	Service Name	Status:	Start Time	Up Time
<input type="radio"/>	Cisco IM and Presence Admin	Running	Wed Sep 2 11:37:28 2015	0 days 00:48:21
<input type="radio"/>	Cisco IM and Presence User	Running	Wed Sep 2 11:37:29 2015	0 days 00:48:20

Start Stop Restart Refresh

Navigation: Cisco Unified CM IM and Presence Administration Go

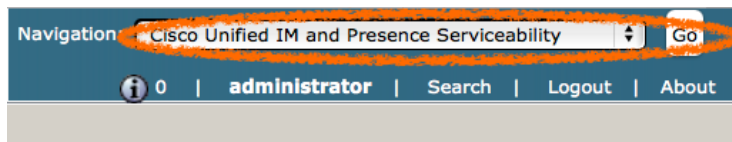


We can now edit the **IM and Presence Domain**.

System > Cluster Topology > Settings IM and Presence Domain: cisco.local Save

Next, we can go back, and start the **Cisco XCP Router** service. Also, we want to enable the remaining services.

Navigation: Cisco Unified IM and Presence Serviceability Go



Tools > Control Center – Network Services

Cisco XCP Router (*selected*)

Start

OK

IM and Presence Services				
	Service Name	Status:	Start Time	Up Time
<input type="radio"/>	Cisco Login Datastore	Running	Wed Sep 2 03:33:17 2015	0 days 08:59:56
<input type="radio"/>	Cisco Route Datastore	Running	Wed Sep 2 03:33:18 2015	0 days 08:59:55
<input type="radio"/>	Cisco Config Agent	Running	Wed Sep 2 11:37:24 2015	0 days 00:55:49
<input type="radio"/>	Cisco OAM Agent	Running	Wed Sep 2 11:37:26 2015	0 days 00:55:47
<input type="radio"/>	Cisco Client Profile Agent	Running	Wed Sep 2 11:37:30 2015	0 days 00:55:43
<input type="radio"/>	Cisco Intercluster Sync Agent	Running	Wed Sep 2 11:37:13 2015	0 days 00:56:00
<input type="radio"/>	Cisco XCP Config Manager	Running	Wed Sep 2 11:37:11 2015	0 days 00:56:02
<input checked="" type="radio"/>	Cisco XCP Router	Not Running		
<input type="radio"/>	Cisco Server Recovery Manager	Running	Wed Sep 2 11:37:25 2015	0 days 00:55:48
<input type="radio"/>	Cisco Replication Watcher	Running	Wed Sep 2 11:37:10 2015	0 days 00:56:03
<input type="radio"/>	Cisco Presence Datastore	Running	Wed Sep 2 11:37:19 2015	0 days 00:55:54
<input type="radio"/>	Cisco SIP Registration Datastore	Running	Wed Sep 2 11:37:20 2015	0 days 00:55:53

DB Services				
	Service Name	Status:	Start Time	Up Time
<input type="radio"/>	Cisco Database Layer Monitor	Running	Wed Sep 2 03:33:16 2015	0 days 08:59:57

SOAP Services				
	Service Name	Status:	Start Time	Up Time
<input type="radio"/>	SOAP -Real-Time Service APIs	Running	Wed Sep 2 11:37:07 2015	0 days 00:56:06
<input type="radio"/>	SOAP -Performance Monitoring APIs	Running	Wed Sep 2 11:37:08 2015	0 days 00:56:05
<input type="radio"/>	SOAP -Log Collection APIs	Running	Wed Sep 2 11:37:09 2015	0 days 00:56:04

Admin Services				
	Service Name	Status:	Start Time	Up Time
<input type="radio"/>	Cisco IM and Presence Admin	Running	Wed Sep 2 11:37:28 2015	0 days 00:55:45
<input type="radio"/>	Cisco IM and Presence User	Running	Wed Sep 2 11:37:29 2015	0 days 00:55:44

☒ Start Stop Restart Refresh

Tools > Service Activation

Check All Services: (*checked*)

Save

OK

Alarm Trace Tools Snmp Help

Service Activation Related Links: Control Center - Feature Services [Go](#)

[Save](#) [Set to Default](#) [Refresh](#)

Status: [Ready](#)

Select Server: [CUPS](#) [Go](#)

☒ Check All Services

IM and Presence Services		
	Service Name	Activation Status
<input checked="" type="checkbox"/>	Cisco SIP Proxy	Deactivated
<input checked="" type="checkbox"/>	Cisco Presence Engine	Deactivated
<input checked="" type="checkbox"/>	Cisco Sync Agent	Deactivated
<input checked="" type="checkbox"/>	Cisco XCP Text Conference Manager	Deactivated
<input checked="" type="checkbox"/>	Cisco XCP Web Connection Manager	Deactivated
<input checked="" type="checkbox"/>	Cisco XCP Connection Manager	Deactivated
<input checked="" type="checkbox"/>	Cisco XCP SIP Federation Connection Manager	Deactivated
<input checked="" type="checkbox"/>	Cisco XCP XMPP Federation Connection Manager	Deactivated
<input checked="" type="checkbox"/>	Cisco XCP Message Archiver	Deactivated
<input checked="" type="checkbox"/>	Cisco XCP Directory Service	Deactivated
<input checked="" type="checkbox"/>	Cisco XCP Authentication Service	Deactivated

Database and Admin Services		
	Service Name	Activation Status
<input checked="" type="checkbox"/>	Cisco AXL Web Service	Deactivated
<input checked="" type="checkbox"/>	Platform SOAP Services	Deactivated
<input checked="" type="checkbox"/>	Cisco Bulk Provisioning Service	Deactivated

Performance and Monitoring Services		
	Service Name	Activation Status
<input checked="" type="checkbox"/>	Cisco Serviceability Reporter	Deactivated

[Save](#) [Set to Default](#) [Refresh](#)

Define the CUCM Publisher as a Presence Gateway

Navigation: **Cisco Unified CM IM and Presence Administration**
Go

Navigation [Cisco Unified CM IM and Presence Administration](#) [Go](#)

[administrator](#) | [About](#) | [Logout](#)

Presence > Gateways > Add New
 Presence Gateway Type: **CUCM**
 Description: **CUCM**
 Presence Gateway: **192.168.1.71**
Save

Cisco Unified CM IM and Presence Administration
For Cisco Unified Communications Solutions

System ▾ Presence ▾ Messaging ▾ Application ▾ User Management ▾ Bulk Administration ▾ Diagnostics ▾ Help ▾

Presence Gateway Configuration

Save

Status
Status: Ready

Presence Gateway Settings (Cisco Unified Communications Manager)
You can configure a Cisco Unified Communications Manager server as a presence gateway. The IM and Presence Service will then trigger phone on/off hook status).

Presence Gateway Type* CUCM
Description* CUCM
Presence Gateway* 192.168.1.71

Save

*- indicates required item.

Specify a Trunk to Carry Presence Information from the CUCM Cluster

Presence > Settings

CUCM IM and Presence Publish Trunk: **PRESENCE**

Save

Cisco Unified CM IM and Presence Administration
For Cisco Unified Communications Solutions

System ▾ Presence ▾ Messaging ▾ Application ▾ User Management ▾ Bulk Administration ▾ Diagnostics ▾ Help ▾

Presence Settings

Save

Status
Status: Ready

Presence Settings
☒ Enable availability sharing
☒ Allow users to view the availability of other users without being prompted for approval
NOTE: this option must be turned on for SIP clients to function properly
☐ Enable use of Email Address when Federating

Maximum Contact List Size (per user)* 200 ☐ No Limit
 Maximum Watchers (per user)* 200 ☐ No Limit

CUCM IM and Presence Publish Trunk PRESENCE

Specify the Cisco SIP Proxy Listener

Presence > Routing > Settings

Preferred Proxy Listener: **Default Cisco SIP Proxy TCP Listener**

Save

OK

Restart All Proxy Services

OK

Cisco Unified CM IM and Presence Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM IM and Presence Administration Go

System Presence Messaging Application User Management Bulk Administration Diagnostics Help

Proxy Configuration Settings

Save

Status
Status: Ready

Restart
Restart All Proxy Services

General Configuration

☐ CVP Enable ACL Configuration

Method/Event Routing Status * On

Preferred Proxy Listener Default Cisco SIP Proxy TCP Listener

Save

Configure Legacy Client Settings

Application > Legacy Clients > Settings

Proxy Listener: **Default Cisco SIP Proxy TCP Listener**

Primary TFTP Server: **192.168.1.72**

Backup TFTP Server: **192.168.1.71**

Save

Even though additional integration options could be implemented, we have now done everything necessary on our CUCM cluster and IM and Presence server to meet the task requirements. Also, we have done everything necessary to support a Cisco Jabber for Windows client, as required by the next task.

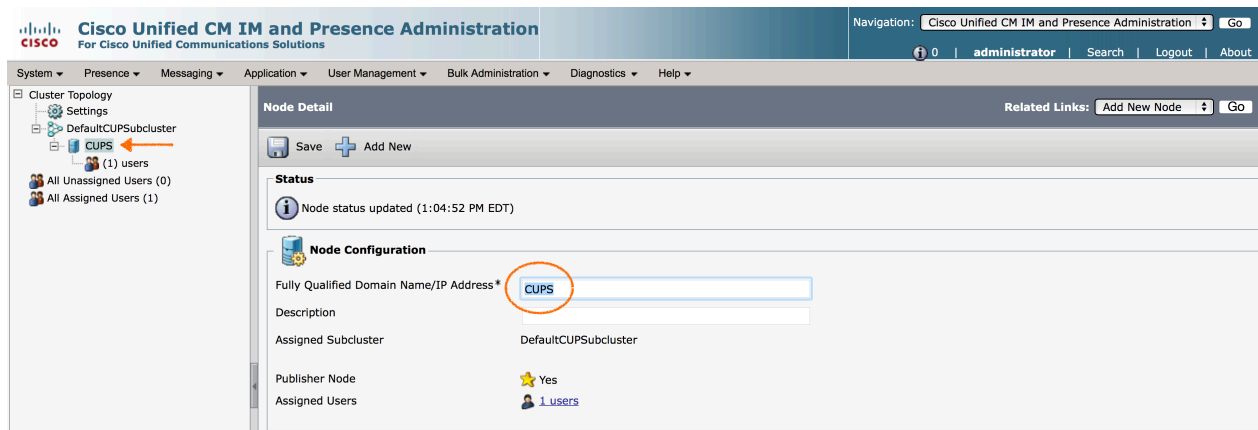
Task 9.2 Cisco Jabber for Windows

The CUCM and IM and Presence server configuration necessary to support the Cisco Jabber for Windows client was performed as part the previous task. So, the focus of this task is getting the Cisco Jabber for Windows client configured.

First, we need to make sure we can resolve the IM and Presence server's hostname and its fully qualified domain name.

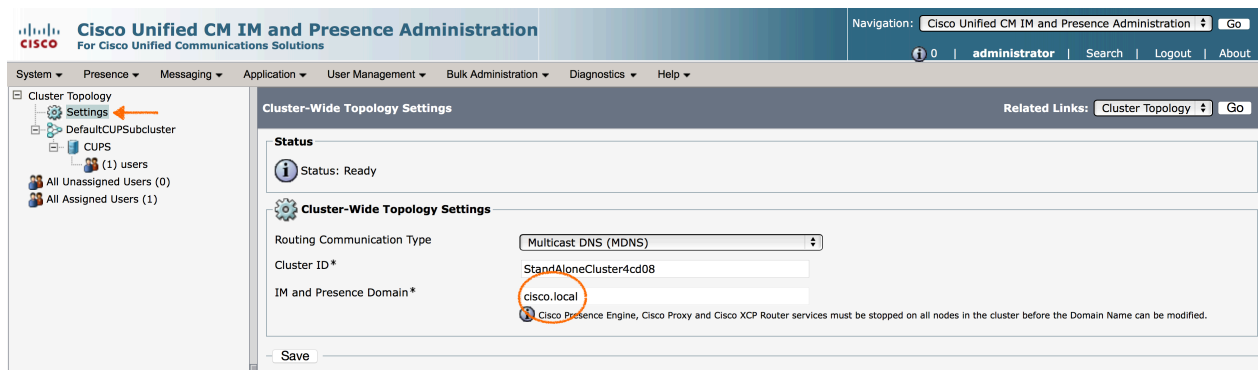
System > Cluster Topology > CUPS (or whatever the server's hostname is)

Note the value in the **Fully Qualified Domain Name/IP Address** field. By default, this value is the IM and Presence server's hostname. In this case, the value is **CUPS**.



Settings

Notice that we configured the IM and Presence Domain value to **cisco.local**. As a result, the fully qualified domain name of the IM and Presence server is **cups.cisco.local**.



The client PC should be able to reach the IM and Presence server based on the hostname (i.e. **CUPS**) and the fully qualified domain name (**cups.cisco.local**).

To check connectivity, open up a command prompt window, and ping both the hostname and the fully qualified domain name. In the real world, if one (or both) of these pings failed, you would probably update the client's DNS server. However, in this lab, we don't have administrative control over the DNS server. Therefore, we need to configure the local **hosts** file to support the name resolution.

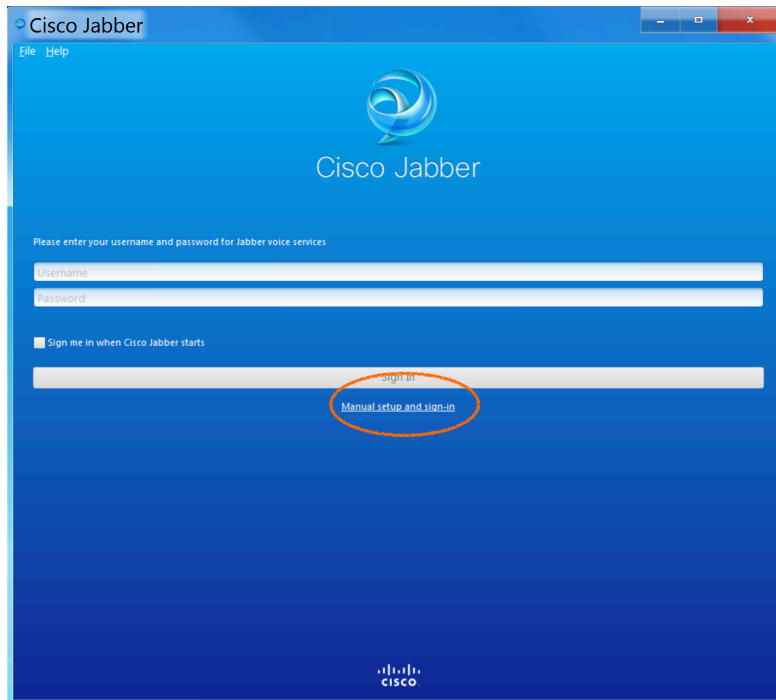
In most recent versions of Microsoft Windows (including Windows 7, Windows 8, and Windows 10), this **hosts** file can be found in the **C:\Windows\System32\drivers\etc** folder.

In our case, the hosts file could be updated to support name resolution for **cups** and **cups.cisco.local** as follows:

```
192.168.1.75 cups.cisco.local
192.168.1.75 cups
```

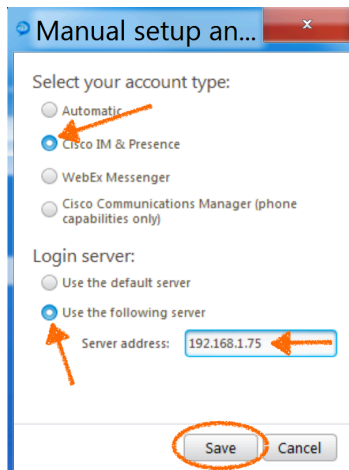
Once the client PC can ping both the IM and Presence server's hostname and fully qualified domain name, launch the Cisco Jabber for Windows client. (**NOTE:** Depending on whether or not the Jabber client has been run before on the Client PC, you might be initially presented with a login box asking for your username in the form of **hqphone2@cups.cisco.local**. The Jabber client will attempt (and fail) to automatically locate the IM and Presence server. Then, you'll be able to do a manual setup.)

Manual setup and sign-in



Cisco IM & Presence (*selected*)
Use the following server (*selected*)
Server address: **192.168.1.75**
Save

NOTE: Different versions of the Cisco Jabber for Windows will look somewhat different and have slightly different configuration options. The version being used in this example is Cisco Jabber for Windows version 9.6.0.



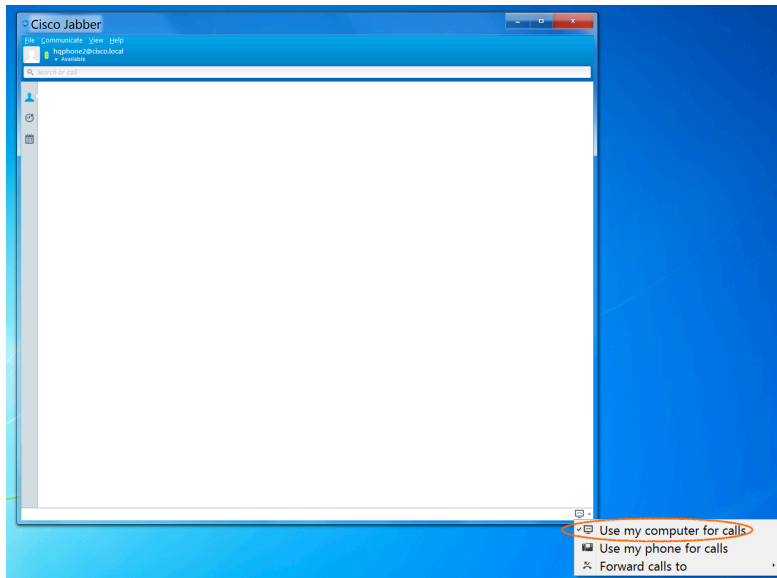
Username: **hqphone2**

Password: **cisco**

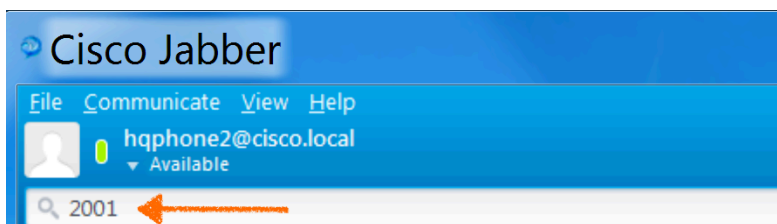
Sign In



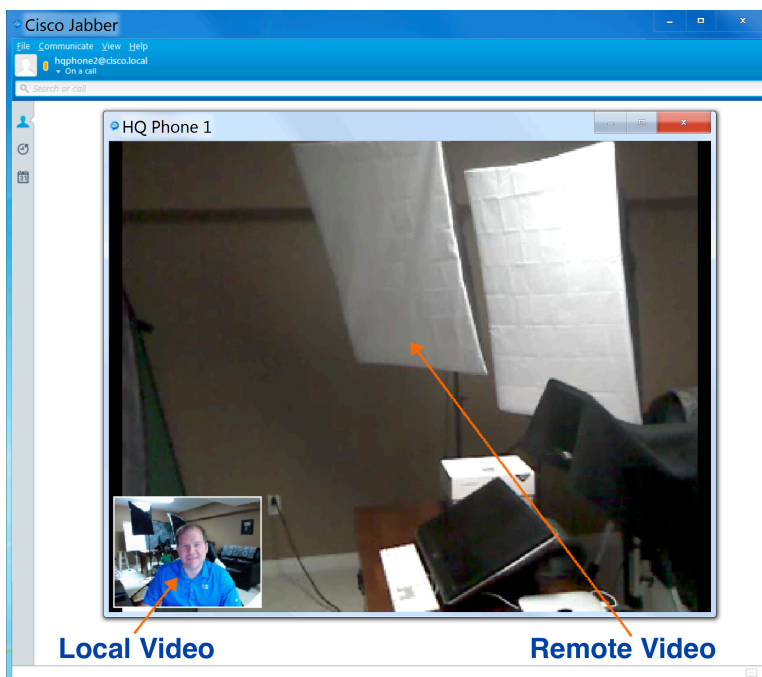
Make sure the Jabber client is configured to **Use my computer for calls**. If we instead had this option set for Use my phone for calls, we would just be using our Jabber client to control the desktop phone (i.e. HQ Phone 2), which does not have video capabilities. Therefore, we would be unable to setup a video call.



Enter a phone number of **2001**, and press **Enter**. Then, answer HQ Phone 1 when it rings.



We should now be able to see video from HQ Phone 1 on our Cisco Jabber for Windows client.



Module 16: Verification

Task 1.1 VLANs

We can check to make sure that the VLANs created on SW1, BR1, and BR2 match our documentation. There's no need to check HQ, because our phones would not be working if the VLANs assigned to the Gigabit Ethernet subinterfaces in HQ didn't match up with SW1.

SW1:

```
SW1#show vlan brief
```

VLAN	Name	Status	Ports
1	default	active	Fa1/0/3, Fa1/0/4, Fa1/0/5 Fa1/0/6, Fa1/0/9, Fa1/0/10 Fa1/0/11, Fa1/0/12, Fa1/0/13 Fa1/0/14, Fa1/0/15, Fa1/0/16 Fa1/0/17, Fa1/0/18, Fa1/0/19 Fa1/0/20, Fa1/0/21, Fa1/0/22 Fa1/0/23, Gi1/0/1, Gi1/0/2
100	SERVER	active	Fa1/0/1, Fa1/0/2
200	VLAN0200	active	Fa1/0/7, Fa1/0/8
300	VLAN0300	active	Fa1/0/7, Fa1/0/8
1002	fddi-default	act/unsup	
1003	token-ring-default	act/unsup	
1004	fddinet-default	act/unsup	
1005	trnet-default	act/unsup	

BR1:

```
BR1#show vlan-switch
```

VLAN	Name	Status	Ports
1	default	active	Fa0/2/2
400	VOICE-VIDEO	active	Fa0/2/0, Fa0/2/1
500	DATA	active	Fa0/2/0, Fa0/2/1, Fa0/2/3
1002	fddi-default	act/unsup	
1003	token-ring-default	act/unsup	
1004	fddinet-default	act/unsup	
1005	trnet-default	act/unsup	

VLAN	Type	SAID	MTU	Parent	RingNo	BridgeNo	Stp	BrdgMode	Trans1	Trans2
1	enet	100001	1500	-	-	-	-	-	1002	1003
400	enet	100400	1500	-	-	-	-	-	0	0
500	enet	100500	1500	-	-	-	-	-	0	0
1002	fddi	101002	1500	-	-	-	-	-	1	1003

1003	tr	101003	1500	1005	0	-	-	srb	1	1002
1004	fdnet	101004	1500	-	-	1	ibm	-	0	0
1005	trnet	101005	1500	-	-	1	ibm	-	0	0

BR2:

BR2#show vlan-switch

VLAN	Name	Status	Ports
1	default	active	Fa0/2/2, Fa0/2/3
600	VLAN0600	active	Fa0/2/0, Fa0/2/1
700	VLAN0700	active	Fa0/2/0, Fa0/2/1
1002	fddi-default	act/unsup	
1003	token-ring-default	act/unsup	
1004	fddinet-default	act/unsup	
1005	trnet-default	act/unsup	

VLAN	Type	SAID	MTU	Parent	RingNo	BridgeNo	Stp	BrdgMode	Trans1	Trans2
1	enet	100001	1500	-	-	-	-	-	1002	1003
600	enet	100600	1500	-	-	-	-	-	0	0
700	enet	100700	1500	-	-	-	-	-	0	0
1002	fddi	101002	1500	-	-	-	-	-	1	1003
1003	tr	101003	1500	1005	0	-	-	srb	1	1002
1004	fdnet	101004	1500	-	-	1	ibm	-	0	0
1005	trnet	101005	1500	-	-	1	ibm	-	0	0

Task 1.2 Address Assignment

HQ:

The HQ phones get their IP addresses from the HQ CUCM publisher, we can navigate to **Device > Phone > Find**, and check to see if the IP addresses assigned to the phones fall within the required range.

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The 'Find and List Phones' page is active, showing a table of phone records. The table has columns for Device Name, Description, Device Pool, Device Protocol, Status, IP Address, Copy, and Super Copy. The first record is CSFhphone2, which is registered with IP 192.168.1.116. The second record is IPCC-1, which is unknown. The third record is IPCC-1, which is unknown. The fourth record is HQ Phone 1, which is registered with IP 10.10.120.19. The fifth record is HQ Phone 2, which is registered with IP 10.10.120.17. The table also shows records for SEPOCD996912474 and SEPECC8821098E9.

BR1:

For BR1 and BR2, we can view the DHCP bindings made by each router, confirming that the IP addresses handed out to the IP phones meet the task requirements.

BR1#**show ip dhcp binding**

Bindings from all pools not associated with VRF:

IP address	Client-ID/ Hardware address/ User name	Lease expiration	Type
10.10.140.10	010c.d996.919a.55	Sep 03 2015 09:17 AM	Automatic
10.10.140.12	0100.24c4.0d8c.fc	Sep 03 2015 09:43 AM	Automatic

BR2:

BR2#**show ip dhcp binding**

Bindings from all pools not associated with VRF:

IP address	Client-ID/ Hardware address/ User name	Lease expiration	Type
10.10.160.10	010c.d996.9026.a3	Infinite	Manual
10.10.160.11	0100.1c58.fb76.01	Infinite	Manual

Task 1.3 Network Time Protocol (NTP) Configuration

We can compare the time displayed on our IP phones with the time displayed on our lab computer (adjusted for the time zone of our lab location).

In this lab, BR1 IP phones should be 3 hours later than HQ IP phones, and BR2 IP phones should be 17 hours later than HQ IP phones.

If we're having an issue, we can check NTP synchronization on a router with the **show ntp status** command.

HQ#**show ntp status**

Clock is synchronized, stratum 3, reference is 192.168.1.78
nominal freq is 250.0000 Hz, actual freq is 249.9980 Hz, precision is 2**21
ntp uptime is 1804500 (1/100 of seconds), resolution is 4016
reference time is D991BDDF.15F05F2D (11:26:39.085 PDT Wed Sep 2 2015)
clock offset is 2.0550 msec, root delay is 31.35 msec
root dispersion is 10.56 msec, peer dispersion is 3.44 msec
loopfilter state is 'CTRL' (Normal Controlled Loop), drift is 0.000007870 s/s
system poll interval is 64, last update was 142 sec ago.

Task 2.1 CUCM SIP (Model 9971)

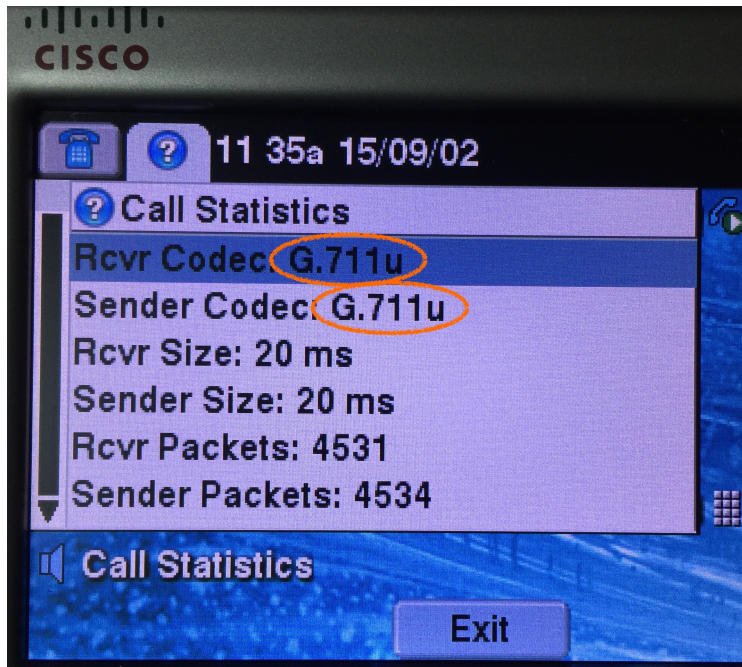
Visually inspect the display of the Cisco 9971 IP phones to make sure the date format, time format, and the number displayed at the top of the screen match the lab guide.

Task 2.2 CUCM SCCP (Model 7965)

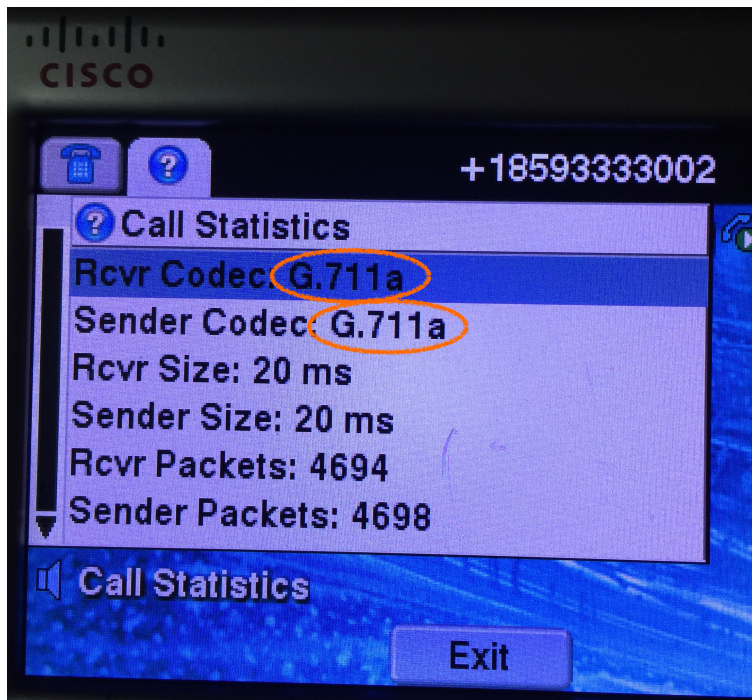
Visually inspect the display of the Cisco 7965 IP phones to make sure the date format, time format, and the number displayed in the upper right corner of the screen match the lab guide.

Task 2.3 CUCM IP Phone Customization

Place a call from HQ Phone 2 to HQ Phone 1. Press the **?** button on HQ Phone 2 twice, and observe the codec is **G.711 uLaw**.



Place a call from BR1 Phone 2 to BR1 Phone 1. Press the **?** button on HQ Phone 2 twice, and observe the codec is **G.711 aLaw**.



Task 2.4 CUCME SIP IP Phone Registration

This was verified as we performed the task.

Task 2.5 CUCME SCCP IP Phone Registration

This was verified as we performed the task.

Task 2.6 CUCME IP Phone Customization

This was verified as we performed the task.

Task 3.1 HQ Cisco IOS H.323 T1-PRI Gateway

We know the gateway works, based on the calls we've placed. However, we want to verify that we've bound the signaling to the appropriate interface.

```
HQ#show run | i h323
  allow-connections h323 to h323
  allow-connections h323 to sip
  allow-connections sip to h323
voice class h323 1
  h323-gateway voip interface
  h323-gateway voip bind srcaddr 10.10.120.1
voice-class h323 1
voice-class h323 1
```

This output confirms that H.323 communication is bound to **10.10.120.1**.

Task 3.2 BR1 Cisco IOS MGCP T1-PRI Gateway

Similar to the previous task's verification, let's just verify our MGCP bindings.

```
BR1#show run | i bind
  isdn bind-l3 ccm-manager
mgcp bind control source-interface Vlan400
mgcp bind media source-interface Vlan400
```

Task 3.3 BR2 CUCM H.323 GW

Again, we know the gateway functions. Also, this task did not require any specific protocol bindings.

Task 3.4 SIP Trunk

This was verified as we performed the task.

Task 3.5 Troubleshooting Video Call Between HQ and BR1

This was verified as we performed the task.

Task 3.6 SIP Gateway

This was verified as we performed the task.

Task 4.1 HQ Call Routing

Since this was a core call routing task, we verified it after we configured it.

Task 4.2 BR1 Call Routing

Since this was a core call routing task, we verified it after we configured it.

Task 4.3 BR2 Call Routing

If we've not yet verified BR2 call routing, let's issue the **debug isdn q931** command on BR2, and place a series of test calls to make sure the ANI/TON match the lab requirements.

```
BR2#debug isdn q931
```

Emergency:

```
Bearer Capability i = 0x8090A3
  Standard = CCITT
  Transfer Capability = Speech
  Transfer Mode = Circuit
```



```
Transfer Rate = 64 kbit/s
Channel ID i = 0xA98383
Exclusive, Channel 3
Progress Ind i = 0x8183 - Origination address is non-ISDN
Display i = 'BR2 Phone 2'
Calling Party Number i = 0x4180, '44444002'
Plan:ISDN, Type:Subscriber(local)
Called Party Number i = 0x81, '999'
Plan:ISDN, Type:Unknown
```

Local:

```
Bearer Capability i = 0x8090A3
Standard = CCITT
Transfer Capability = Speech
Transfer Mode = Circuit
Transfer Rate = 64 kbit/s
Channel ID i = 0xA98383
Exclusi
BR2#ve, Channel 3
Progress Ind i = 0x8183 - Origination address is non-ISDN
Display i = 'BR2 Phone 2'
Calling Party Number i = 0x4180, '44444002'
Plan:ISDN, Type:Subscriber(local)
Called Party Number i = 0xC1, '44444040'
Plan:ISDN, Type:Subscriber(local)
```

International:

```
Bearer Capability i = 0x8090A3
Standard = CCITT
Transfer Capability = Speech
Transfer Mode = Circuit
Transfer Rate = 64 kbit/s
Channel ID i = 0xA98383
Excl
BR2#usive, Channel 3
Progress Ind i = 0x8183 - Origination address is non-ISDN
Display i = 'BR2 Phone 2'
Calling Party Number i = 0x1180, '+8144444002'
Plan:ISDN, Type:International
Called Party Number i = 0x91, '14082222020'
Plan:ISDN, Type:International
```

Task 4.4 Number Globalization and Localization

This was verified as we performed the task.

Task 4.5 URI Dialing

This was verified as we performed the task.

Task 5.1 Video Conference Bridge

This was verified as we performed the task.

Task 5.2 Ad-Hoc Conferencing

- Call into BR2 Phone 1 from a PSTN phone.
- Answer the call.
- Press the **Confrn** softkey on BR2 Phone 1.
- From BR2 Phone 1, dial BR2 Phone 2.
- Answer the call.
- On BR2 Phone 1, press the **Confrn** softkey again.
- A three-party ad-hoc conference should now be established.

Task 6.1 Cisco Catalyst Switch QoS

This task can best be verified by reexamining our configuration against the lab guide.

```
mls qos map cos-dscp 0 8 16 24 34 46 48 56
mls qos srr-queue output cos-map queue 1 threshold 3 4 5
mls qos srr-queue output cos-map queue 2 threshold 3 6 7
mls qos srr-queue output cos-map queue 3 threshold 3 3
mls qos srr-queue output cos-map queue 4 threshold 1 0
mls qos srr-queue output cos-map queue 4 threshold 2 1
mls qos srr-queue output cos-map queue 4 threshold 3 2
mls qos queue-set output 2 threshold 4 25 50 100 100
mls qos
!
interface FastEthernet1/0/7
  switchport access vlan 300
  switchport voice vlan 200
  queue-set 2
  priority-queue out
  mls qos trust device cisco-phone
  mls qos trust cos
  spanning-tree portfast
!
interface FastEthernet1/0/8
  switchport access vlan 300
  switchport voice vlan 200
  queue-set 2
  priority-queue out
  mls qos trust device cisco-phone
  mls qos trust cos
  spanning-tree portfast
!
interface FastEthernet1/0/24
  switchport trunk encapsulation dot1q
  switchport trunk native vlan 300
  switchport mode trunk
  srr-queue bandwidth share 1 40 20 40
  srr-queue bandwidth shape 8 0 0 0
  priority-queue out
  mls qos trust dscp
```

Task 6.2 Router QoS

First, let's examine the configuration on HQ to make sure MLP is configured. One way to do this is to examine our IP routing table to see if we have a **Virtual-Access** interface acting as an egress interface.

```
HQ#show ip route
```

```
Codes: L - local, C - connected, S - static, R - RIP, M - mobile, B - BGP
       D - EIGRP, EX - EIGRP external, O - OSPF, IA - OSPF inter area
       N1 - OSPF NSSA external type 1, N2 - OSPF NSSA external type 2
       E1 - OSPF external type 1, E2 - OSPF external type 2
       i - IS-IS, su - IS-IS summary, L1 - IS-IS level-1, L2 - IS-IS level-2
       ia - IS-IS inter area, * - candidate default, U - per-user static
route
      o - ODR, P - periodic downloaded static route, H - NHRP, l - LISP
      + - replicated route, % - next hop override
```

Gateway of last resort is 192.168.1.1 to network 0.0.0.0

```
S*    0.0.0.0/0 [1/0] via 192.168.1.1
      10.0.0.0/8 is variably subnetted, 16 subnets, 3 masks
C      10.10.32.1/32 is directly connected, Loopback0
O      10.10.32.2/32 [110/65] via 10.10.33.2, 05:59:02, Serial0/1/0.1
O      10.10.32.3/32 [110/131] via 10.10.33.130, 05:59:02, Virtual-Access3
C      10.10.33.0/25 is directly connected, Serial0/1/0.1
L      10.10.33.1/32 is directly connected, Serial0/1/0.1
C      10.10.33.128/25 is directly connected, Virtual-Access3
L      10.10.33.129/32 is directly connected, Virtual-Access3
C      10.10.33.130/32 is directly connected, Virtual-Access3
C      10.10.120.0/24 is directly connected, GigabitEthernet0/0.200
L      10.10.120.1/32 is directly connected, GigabitEthernet0/0.200
C      10.10.130.0/24 is directly connected, GigabitEthernet0/0.300
L      10.10.130.1/32 is directly connected, GigabitEthernet0/0.300
O      10.10.140.0/24 [110/65] via 10.10.33.2, 05:59:02, Serial0/1/0.1
O      10.10.150.0/24 [110/65] via 10.10.33.2, 05:59:02, Serial0/1/0.1
O      10.10.160.0/24 [110/131] via 10.10.33.130, 05:59:02, Virtual-Access3
O      10.10.170.0/24 [110/131] via 10.10.33.130, 05:59:02, Virtual-Access3
      192.168.1.0/24 is variably subnetted, 2 subnets, 2 masks
C      192.168.1.0/24 is directly connected, GigabitEthernet0/0.100
L      192.168.1.77/32 is directly connected, GigabitEthernet0/0.100
```

We do. This implies that we're using MLP. It also implies that BR2 is correctly configured for MLP. Otherwise, we would not have IP connectivity between these two routers.

Next, let's make sure we don't have any cRTP configuration in HQ or BR2, left over from our running of AutoQoS

```
HQ#show run | i header
```

```
HQ#
```

```
BR2#show run | i header
```

```
BR2#
```

Finally, let's check to make sure our LLQ configuration is specifying the correct bandwidth amount on both routers.

```
HQ#show policy-map
Policy Map AutoQoS-Policy-Trust
  Class AutoQoS-VoIP-RTP-Trust
    priority 176 (kbps)
  Class AutoQoS-VoIP-Control-Trust
    bandwidth 5 (%)
  Class class-default
    fair-queue

BR2#show policy-map
Policy Map AutoQoS-Policy-Trust
  Class AutoQoS-VoIP-RTP-Trust
    priority 176 (kbps)
  Class AutoQoS-VoIP-Control-Trust
    bandwidth 5 (%)
  Class class-default
    fair-queue
```

Task 7.1 Cisco Unity Connection (CUC) SCCP Integration Configuration

Instead of checking both HQ IP phones, to save time, we'll just check one phone to spot-check. For example, let's choose HQ Phone 1.

- Call into HQ Phone 1, and verify the incoming call diverts to voicemail after 10 seconds.
- Go off-hook on HQ Phone 1, and then call into HQ Phone 1 from another phone to verify the call immediate diverts to voicemail.
- Leave a message, and make sure the MWI light illuminates.
- Delete the messages, and make sure the MWI light goes off.

Task 7.2 Cisco Unity Connection (CUC) SIP Integration Configuration

Instead of checking both BR1 IP phones, to save time, we'll just check one phone to spot-check. For example, let's choose BR1 Phone 1.

- Call into BR1 Phone 1, and verify the incoming call diverts to voicemail after 10 seconds.
- Go off-hook on BR1 Phone 1, and then call into BR1 Phone 1 from another phone to verify the call immediate diverts to voicemail.
- Leave a message, and make sure the MWI light illuminates.
- Delete the messages, and make sure the MWI light goes off.

Task 7.3 CUC Customization

This was verified as we performed the task.

Task 7.4 Cisco Unity Express (CUE) Module Initialization

This was verified as we performed the task.

Task 7.5 Integrating CUE with CUCME

Instead of checking both BR2 IP phones, to save time, we'll just check one phone to spot-check. For example, let's choose BR2 Phone 1.

- Call into BR2 Phone 1, and verify the incoming call diverts to voicemail after 10 seconds.
- Go off-hook on BR2 Phone 1, and then call into BR1 Phone 1 from another phone to verify the call immediate diverts to voicemail.
- Leave a message, and make sure the MWI light illuminates.
- Delete the messages, and make sure the MWI light goes off.

In our example, MWI was not functioning correctly at the BR2 site. The issue was the SIP gateway configured on the CUE module was set to 10.10.160.1. While 10.10.160.1 is the default gateway for routing IP traffic, the SIP gateway should point to the IP address to which SIP is bound on the BR2 router (i.e. the Loopback 0 interface's IP address of 10.10.32.3). To resolve this issue, we changed the SIP gateway from 10.10.160.1 to 10.10.32.3.

```
CUE# show ccn subsystem sip
SIP Gateway: 10.10.160.1
SIP Port Number: 5060
DTMF Relay: sip-notify
MWI Notification: unsolicited
MWI Envelope Info: enabled
Transfer Mode: blind(REFER)
SIP RFC Compliance: Pre-RFC3261
CUE# conf term
Enter configuration commands, one per line. End with CNTL/Z.
CUE(config)# ccn subsystem sip
CUE(config-sip)# gateway address 10.10.32.3
CUE(config-sip)# end
CUE(config)# end
CUE# copy run star
CUE#
```

Task 7.6 CUE Customization

This was verified as we performed the task.

Task 8.1 UCCX Integration

This was verified as we performed the task.

Task 8.2 UCCX Scripting

This was verified as we performed the task.

Task 9.1 IM and Presence Integration with CUCM

This was verified as we performed the task.

Task 9.2 Cisco Jabber for Windows

This was verified as we performed the task.

Initial Configurations

Initial Configuration of SW1

```
SW1#show run
Building configuration...

Current configuration : 3433 bytes
!
version 12.2
no service pad
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname SW1
!
boot-start-marker
boot-end-marker
!
!
!
!
no aaa new-model
switch 1 provision ws-c3750-24p
system mtu routing 1500
no ip domain-lookup
!
!
!
!
crypto pki trustpoint TP-self-signed-4221020544
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-4221020544
  revocation-check none
  rsakeypair TP-self-signed-4221020544
!
!
crypto pki certificate chain TP-self-signed-4221020544
  certificate self-signed 01
    3082023C 308201A5 A0030201 02020101 300D0609 2A864886 F70D0101 04050030
    31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
    69666963 6174652D 34323231 30323035 3434301E 170D3933 30333031 30303031
    34385A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
    4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D34 32323130
    32303534 3430819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281
    8100A943 8A08D7A8 998E5940 E7FBD4DE 5739FDE1 E41961A9 2E8EE36B 45CC0E11
    F171CCED 5DE38ACE ACA32380 4080A5B5 2431AFA5 8050A20D A596C258 84C6B5AD
```

```

08A8F62D 25119AF2 F3DA4265 E988DB7A C8B7B70A B3B7AF33 F1C42992 9C296B04
882F2617 431B4704 7CCE46BF 72306494 B48F137E 1B034566 E37DF34C 15DEAD66
C22F0203 010001A3 64306230 0F060355 1D130101 FF040530 030101FF 300F0603
551D1104 08300682 04535731 2E301F06 03551D23 04183016 8014CD56 AB802995
8477117C BAA7F6C4 0ADC59E3 A458301D 0603551D 0E041604 14CD56AB 80299584
77117CBA A7F6C40A DC59E3A4 58300D06 092A8648 86F70D01 01040500 03818100
18782CD8 878941D7 5442B651 AC5DC14E EDC041D5 DE38C126 689FB661 9192199D
B9FE3186 A6E5C4F1 CCD652BC 680C99E4 83843ECE 417B3F62 60F9FFB9 1EAF85CA
3775A60D 39DABB98 9AEE3CC3 1250DE7E C2F5E894 E185962E AEB809BF E62D506A
120D8B9E 74654187 DE561036 1B9E7880 A00ED7D4 A7587DE7 EC8AA43B 8610D678
quit
!
!
!
spanning-tree mode pvst
spanning-tree extend system-id
!
vlan internal allocation policy ascending
!
!
!
interface FastEthernet1/0/1
    switchport access vlan 100
!
interface FastEthernet1/0/2
    switchport access vlan 100
!
interface FastEthernet1/0/3
!
interface FastEthernet1/0/4
!
interface FastEthernet1/0/5
!
interface FastEthernet1/0/6
!
interface FastEthernet1/0/7
    switchport access vlan 300
    switchport voice vlan 200
    spanning-tree portfast
!
interface FastEthernet1/0/8
    switchport access vlan 300
    switchport voice vlan 200
    spanning-tree portfast
!
interface FastEthernet1/0/9
!
interface FastEthernet1/0/10
!
interface FastEthernet1/0/11
!
interface FastEthernet1/0/12
!
interface FastEthernet1/0/13
!
interface FastEthernet1/0/14
!

```

```

interface FastEthernet1/0/15
!
interface FastEthernet1/0/16
!
interface FastEthernet1/0/17
!
interface FastEthernet1/0/18
!
interface FastEthernet1/0/19
!
interface FastEthernet1/0/20
!
interface FastEthernet1/0/21
!
interface FastEthernet1/0/22
!
interface FastEthernet1/0/23
!
interface FastEthernet1/0/24
    switchport trunk encapsulation dot1q
    switchport mode trunk
!
interface GigabitEthernet1/0/1
!
interface GigabitEthernet1/0/2
!
interface Vlan1
    no ip address
    shutdown
!
interface Vlan100
    ip address 192.168.1.80 255.255.255.0
!
ip classless
ip http server
ip http secure-server
!
ip sla enable reaction-alerts
no cdp advertise-v2
cdp timer 5
!
!
line con 0
    exec-timeout 0 0
    logging synchronous
line vty 0 4
    login
line vty 5 15
    login
!
end

```

Initial Configuration of HQ

```

HQ#show run
Building configuration...

```

```

Current configuration : 4381 bytes
!
! Last configuration change at 20:22:18 UTC Thu Sep 3 2015 by cisco
version 15.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname HQ
!
boot-start-marker
boot-end-marker
!
!
card type t1 0 0
logging buffered 51200 warnings
enable secret 4 tnhtc92DXBhelxjYk8LWJrPV36S2i4ntXrpb4RFmfqY
!
no aaa new-model
no network-clock-participate wic 0
!
ip cef
!
!
!
ip dhcp excluded-address 10.10.10.1
!
!
!
no ip domain lookup
no ipv6 cef
multilink bundle-name authenticated
!
!
!
!
!
!
crypto pki trustpoint TP-self-signed-2136404343
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-2136404343
  revocation-check none
  rsakeypair TP-self-signed-2136404343
!
!
crypto pki certificate chain TP-self-signed-2136404343
  certificate self-signed 01
    3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
    31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
    69666963 6174652D 32313336 34303433 3433301E 170D3133 30363238 30323434
    30395A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
    4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D32 31333634
    30343334 3330819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281
    81008A62 2482A40E F3E36150 A11F7CF6 E55D8662 840672FC B7B4B14B F99D5AC2
    7937B260 F8AD2D0E 638655D8 F96C01E0 D2B22FAB C5112ED8 597D4507 7379A50B

```



```

9EB48762 BA0474F0 E7883B86 E201B4CC 9EB100CA BDC889A9 3022615C CA2EAB77
9DD51DEA 956D6BB9 792836C5 0556EA75 F6518C79 62C133F5 3881B58A C64E1EF0
31570203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603
551D2304 18301680 149B0E84 AEF12589 6E46FB18 38FE80DC 7971F815 A5301D06
03551D0E 04160414 9B0E84AE F125896E 46FB1838 FE80DC79 71F815A5 300D0609
2A864886 F70D0101 05050003 81810088 EC6CD455 80619269 7FAA646B 7155B671
69AA31AF 486EF9C7 BA2C0BAC D956DF54 C44DAD1D 8F8849EC 5B1A5889 3EC6F3E9
112A5801 543D9927 82F46CCB 59C5DC90 3F3A8FE1 11079927 E7A651B3 3789C69C
5D9EE330 CD1493A9 EF6DFBE5 5161A7C9 885CF68A 5EB8FDA5 4DC3F56C 085513EB
FACC6B6D 9A6B5D4D 978DD68E 4671D6
quit
voice-card 0
!
!
!
voice service voip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
!
!
!
!
!
!
license udi pid CISC02911/K9 sn FTX1726AL7J
hw-module pvdm 0/0
!
hw-module pvdm 0/1
!
!
!
username cisco secret 4 tnhtc92DXBhelxjYk8LWJrPV36S2i4ntXrpb4RFmfqY
!
redundancy
!
!
controller T1 0/0/0
cablelength long 0db
!
!
!
!
!
interface Loopback0
ip address 10.10.32.1 255.255.255.255
!
interface Embedded-Service-Engine0/0
no ip address
shutdown
!
interface GigabitEthernet0/0
description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
no ip address
duplex auto
speed auto
!
interface GigabitEthernet0/0.100
encapsulation dot1Q 100

```

```

    ip address 192.168.1.77 255.255.255.0
    !
interface GigabitEthernet0/0.200
    encapsulation dot1Q 200
    ip address 10.10.120.1 255.255.255.0
    !
interface GigabitEthernet0/0.300
    encapsulation dot1Q 300 native
    ip address 10.10.130.1 255.255.255.0
    !
interface GigabitEthernet0/1
    no ip address
    shutdown
    duplex auto
    speed auto
    !
interface GigabitEthernet0/2
    no ip address
    shutdown
    duplex auto
    speed auto
    !
interface Serial0/1/0
    no ip address
    encapsulation frame-relay IETF
    !
interface Serial0/1/0.1 point-to-point
    ip address 10.10.33.1 255.255.255.128
    frame-relay interface-dlci 102
    !
interface Serial0/1/0.2 point-to-point
    ip address 10.10.33.129 255.255.255.128
    frame-relay interface-dlci 103
    !
router ospf 100
    network 0.0.0.0 255.255.255.255 area 0
    !
ip forward-protocol nd
    !
ip http server
ip http access-class 23
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
    !
ip route 0.0.0.0 0.0.0.0 192.168.1.1
    !
access-list 23 permit 10.10.10.0 0.0.0.7
    !
    !
    !
control-plane
    !
    !
    !
    !

```

```

!
!
mgcp profile default
!
!
!
!
!
gatekeeper
shutdown
!
!
!
line con 0
  exec-timeout 0 0
  logging synchronous
  login local
line aux 0
line 2
  no activation-character
  no exec
  transport preferred none
  transport input all
  transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
  stopbits 1
line vty 0 4
  access-class 23 in
  privilege level 15
  login local
  transport input telnet ssh
line vty 5 15
  access-class 23 in
  privilege level 15
  login local
  transport input telnet ssh
!
scheduler allocate 20000 1000
!
end

```

Initial Configuration of BR1

BR1#**show run**

Building configuration...

Current configuration : 4118 bytes

```

!
! Last configuration change at 20:33:03 UTC Thu Sep 3 2015 by cisco
version 15.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname BR1
!

```

```

boot-start-marker
boot-end-marker
!
!
! card type command needed for slot/vwic-slot 0/0
logging buffered 51200 warnings
!
no aaa new-model
!
ip cef
!
!
!
ip dhcp excluded-address 10.10.10.1
!
!
!
no ip domain lookup
ip domain name kwtrain.com
no ipv6 cef
multilink bundle-name authenticated
!
!
!
!
!
!
crypto pki trustpoint TP-self-signed-1996982465
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-1996982465
  revocation-check none
  rsakeypair TP-self-signed-1996982465
!
!
crypto pki certificate chain TP-self-signed-1996982465
  certificate self-signed 01
    3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
    31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
    69666963 6174652D 31393936 39383234 3635301E 170D3133 30363238 30323331
    35395A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
    4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D31 39393639
    38323436 3530819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281
    81009EC2 17DFBEF6 DB9E0D6D 06B06014 0E0BFAE3 95FFB6E3 96E4A204 4D6CA7C6
    2E83BD2A 62B591D6 7CE5624F 0F8C96AF 8989E38A 2D56460D 0B43F94C 68078642
    76650163 495038E1 7844067C 66BDB7E0 2490CE87 C9387A9F EDB39B8C F685703D
    E425CC75 285102BA FF29EA27 3F2B2249 73388397 05957F5D 6AB2DD15 6CF609CD
    1ACB0203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603
    551D2304 18301680 14F1E5A1 F95B139B 5876237C 0A10E112 C3F4AF6D 70301D06
    03551D0E 04160414 F1E5A1F9 5B139B58 76237C0A 10E112C3 F4AF6D70 300D0609
    2A864886 F70D0101 05050003 8181002D 0A22176D 78D7501D C14339E1 89B6ACEC
    6DED275E 9314EE28 FD90D95A 2A84CA03 BAFA17F7 FCF0FE41 32AA2D16 7820A5BE
    5D9872A5 4443184D 892ADF50 78C07ADC 6D4F6AEE 7A969EBE 9ADC78A9 90F86569
    8D7B9E90 E7B14315 3F6501A9 C0C2D2F5 9D2C4168 8B708EBB CBDAE81B E33736EA
    6286940A FE9680A4 6F6E021E C98E1C
  quit
voice-card 0
!

```

```

!
!
!
!
!
!
license udi pid CISCO2911/K9 sn FTX1726AL6K
hw-module pvdm 0/0
!
hw-module pvdm 0/1
!
!
!
username cisco secret 4 tnhtc92DXBhelxjYk8LWJrPV36S2i4ntXrpb4RFmfqY
!
redundancy
!
!
!
!
!
interface Loopback0
 ip address 10.10.32.2 255.255.255.255
!
interface Embedded-Service-Engine0/0
 no ip address
 shutdown
!
interface GigabitEthernet0/0
 description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface GigabitEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface GigabitEthernet0/2
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface Serial0/1/0
 no ip address
 encapsulation frame-relay IETF
!
interface Serial0/1/0.1 point-to-point
 ip address 10.10.33.2 255.255.255.128
 frame-relay interface-dlci 201
!

```

```

interface FastEthernet0/2/0
  no ip address
!
interface FastEthernet0/2/1
  no ip address
!
interface FastEthernet0/2/2
  no ip address
!
interface FastEthernet0/2/3
  switchport access vlan 500
  no ip address
!
interface Vlan1
  no ip address
!
interface Vlan400
  ip address 10.10.140.1 255.255.255.0
!
interface Vlan500
  ip address 10.10.150.1 255.255.255.0
!
router ospf 100
  network 0.0.0.0 255.255.255.255 area 0
!
ip forward-protocol nd
!
ip http server
ip http access-class 23
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
!
access-list 23 permit 10.10.10.0 0.0.0.7
!
!
!
control-plane
!
!
!
!
!
!
!
mgcp profile default
!
!
!
!
!
gatekeeper
  shutdown
!
!
!

```

```

line con 0
  exec-timeout 0 0
  logging synchronous
  login local
line aux 0
line 2
  no activation-character
  no exec
  transport preferred none
  transport input all
  transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
  stopbits 1
line vty 0 4
  access-class 23 in
  privilege level 15
  login local
  transport input telnet ssh
line vty 5 15
  access-class 23 in
  privilege level 15
  login local
  transport input telnet ssh
!
scheduler allocate 20000 1000
!
end

```

Initial Configuration of BR2

BR2#**show run**

Building configuration...

```

Current configuration : 2417 bytes
!
! Last configuration change at 20:54:54 UTC Thu Sep 3 2015
! NVRAM config last updated at 20:55:02 UTC Thu Sep 3 2015
! NVRAM config last updated at 20:55:02 UTC Thu Sep 3 2015
version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname BR2
!
boot-start-marker
boot-end-marker
!
!
! card type command needed for slot/vwic-slot 0/0
!
no aaa new-model
!
!
no ipv6 cef
ip source-route

```

```
ip cef
!
!
!
!
!
no ip domain lookup
!
multilink bundle-name authenticated
!
!
!
!
!
crypto pki token default removal timeout 0
!
!
voice-card 0
!
!
!
voice service voip
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
!
!
!
!
license udi pid CISCO2921/K9 sn FTX1540AHDV
hw-module pvdm 0/0
!
hw-module sm 1
!
!
!
!
redundancy
!
!
!
!
!
!
!
!
!
interface Loopback0
 ip address 10.10.32.3 255.255.255.255
!
interface Embedded-Service-Engine0/0
 no ip address
 shutdown
!
interface GigabitEthernet0/0
 no ip address
```



```

shutdown
duplex auto
speed auto
!
interface GigabitEthernet0/1
no ip address
shutdown
duplex auto
speed auto
!
interface GigabitEthernet0/2
no ip address
shutdown
duplex auto
speed auto
!
interface Serial0/1/0
no ip address
encapsulation frame-relay IETF
!
interface Serial0/1/0.1 point-to-point
ip address 10.10.33.130 255.255.255.128
frame-relay interface-dlci 301
!
interface FastEthernet0/2/0
no ip address
!
interface FastEthernet0/2/1
no ip address
!
interface FastEthernet0/2/2
no ip address
!
interface FastEthernet0/2/3
no ip address
!
interface SM1/0
no ip address
shutdown
!Application: CUE Running on SM
!
interface SM1/1
no ip address
shutdown
!
interface Vlan1
no ip address
!
!
router ospf 100
network 0.0.0.0 255.255.255.255 area 0
!
ip forward-protocol nd
!
no ip http server
no ip http secure-server
!

```

```

!
!
!
!
!
!
control-plane
!
!
!
!
mgcp profile default
!
!
!
!
!
gatekeeper
  shutdown
!
!
!
line con 0
  exec-timeout 0 0
  logging synchronous
line aux 0
line 2
  no activation-character
  no exec
  transport preferred none
  transport input all
  transport output lat pad telnet rlogin lapb-ta mop udptn v120 ssh
  stopbits 1
line 67
  no activation-character
  no exec
  transport preferred none
  transport input all
  transport output lat pad telnet rlogin lapb-ta mop udptn v120 ssh
  stopbits 1
line vty 0 4
  login
  transport input all
!
scheduler allocate 20000 1000
end

```

Final Configurations

Final Configuration of SW1

```

SW1#show run
Building configuration...

```

```

Current configuration : 4173 bytes
!
version 12.2
no service pad
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname SW1
!
boot-start-marker
boot-end-marker
!
!
!
!
no aaa new-model
switch 1 provision ws-c3750-24p
system mtu routing 1500
no ip domain-lookup
!
!
!
mls qos map cos-dscp 0 8 16 24 34 46 48 56
mls qos srr-queue output cos-map queue 1 threshold 3 4 5
mls qos srr-queue output cos-map queue 2 threshold 3 6 7
mls qos srr-queue output cos-map queue 3 threshold 3 3
mls qos srr-queue output cos-map queue 4 threshold 1 0
mls qos srr-queue output cos-map queue 4 threshold 2 1
mls qos srr-queue output cos-map queue 4 threshold 3 2
mls qos queue-set output 2 threshold 4 25 50 100 100
mls qos
!
crypto pki trustpoint TP-self-signed-4221020544
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-4221020544
  revocation-check none
  rsakeypair TP-self-signed-4221020544
!
!
crypto pki certificate chain TP-self-signed-4221020544
certificate self-signed 01
  3082023C 308201A5 A0030201 02020101 300D0609 2A864886 F70D0101 04050030
  31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
  69666963 6174652D 34323231 30323035 3434301E 170D3933 30333031 30303031
  34385A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D34 32323130
  32303534 3430819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281
  8100A943 8A08D7A8 998E5940 E7FBD4DE 5739FDE1 E41961A9 2E8EE36B 45CC0E11
  F171CCED 5DE38ACE ACA32380 4080A5B5 2431AFA5 8050A20D A596C258 84C6B5AD
  08A8F62D 25119AF2 F3DA4265 E988DB7A C8B7B70A B3B7AF33 F1C42992 9C296B04
  882F2617 431B4704 7CCE46BF 72306494 B48F137E 1B034566 E37DF34C 15DEAD66
  C22F0203 010001A3 64306230 0F060355 1D130101 FF040530 030101FF 300F0603
  551D1104 08300682 04535731 2E301F06 03551D23 04183016 8014CD56 AB802995
  8477117C BAA7F6C4 0ADC59E3 A458301D 0603551D 0E041604 14CD56AB 80299584
  77117CBA A7F6C40A DC59E3A4 58300D06 092A8648 86F70D01 01040500 03818100
  18782CD8 878941D7 5442B651 AC5DC14E EDC041D5 DE38C126 689FB661 9192199D

```

```

B9FE3186 A6E5C4F1 CCD652BC 680C99E4 83843ECE 417B3F62 60F9FFB9 1EAF85CA
3775A60D 39DABB98 9AEE3CC3 1250DE7E C2F5E894 E185962E AEB809BF E62D506A
120D8B9E 74654187 DE561036 1B9E7880 A00ED7D4 A7587DE7 EC8AA43B 8610D678
quit
!
!
!
spanning-tree mode pvst
spanning-tree extend system-id
!
vlan internal allocation policy ascending
!
!
!
interface FastEthernet1/0/1
    switchport access vlan 100
!
interface FastEthernet1/0/2
    switchport access vlan 100
!
interface FastEthernet1/0/3
!
interface FastEthernet1/0/4
!
interface FastEthernet1/0/5
!
interface FastEthernet1/0/6
!
interface FastEthernet1/0/7
    switchport access vlan 300
    switchport voice vlan 200
    queue-set 2
    priority-queue out
    mls qos trust device cisco-phone
    mls qos trust cos
    spanning-tree portfast
!
interface FastEthernet1/0/8
    switchport access vlan 300
    switchport voice vlan 200
    queue-set 2
    priority-queue out
    mls qos trust device cisco-phone
    mls qos trust cos
    spanning-tree portfast
!
interface FastEthernet1/0/9
!
interface FastEthernet1/0/10
!
interface FastEthernet1/0/11
!
interface FastEthernet1/0/12
!
interface FastEthernet1/0/13
!
interface FastEthernet1/0/14

```

```

!
interface FastEthernet1/0/15
!
interface FastEthernet1/0/16
!
interface FastEthernet1/0/17
!
interface FastEthernet1/0/18
!
interface FastEthernet1/0/19
!
interface FastEthernet1/0/20
!
interface FastEthernet1/0/21
!
interface FastEthernet1/0/22
!
interface FastEthernet1/0/23
!
interface FastEthernet1/0/24
    switchport trunk encapsulation dot1q
    switchport trunk native vlan 300
    switchport mode trunk
    srr-queue bandwidth share 1 40 20 40
    srr-queue bandwidth shape 8 0 0 0
    priority-queue out
    mls qos trust dscp
!
interface GigabitEthernet1/0/1
!
interface GigabitEthernet1/0/2
!
interface Vlan1
    no ip address
    shutdown
!
interface Vlan100
    ip address 192.168.1.80 255.255.255.0
!
ip classless
ip http server
ip http secure-server
!
ip sla enable reaction-alerts
cdp timer 5
!
!
line con 0
    exec-timeout 0 0
    logging synchronous
line vty 0 4
    login
line vty 5 15
    login
!
end

```

Final Configuration of HQ

HQ#**show run**

Building configuration...

Current configuration : 8471 bytes

```
!  
! Last configuration change at 05:27:38 PDT Thu Sep 3 2015  
version 15.2  
service timestamps debug datetime msec  
service timestamps log datetime msec  
no service password-encryption  
!  
hostname HQ  
!  
boot-start-marker  
boot-end-marker  
!  
!  
card type t1 0 0  
logging buffered 51200 warnings  
enable secret 4 tnhtc92DXBhelxjYk8LWJrPV36S2i4ntXrpb4RFmfqY  
!  
no aaa new-model  
clock timezone PST -8 0  
clock summer-time PDT recurring  
network-clock-participate wic 0  
network-clock-select 1 T1 0/0/0  
!  
ip cef  
!  
!  
!  
ip dhcp excluded-address 10.10.10.1  
!  
ip dhcp pool ccp-pool  
    import all  
    network 10.10.10.0 255.255.255.248  
    default-router 10.10.10.1  
    lease 0 2  
!  
!  
!  
no ip domain lookup  
ip domain name kwtrain.com  
no ipv6 cef  
multilink bundle-name authenticated  
!  
!  
!  
!  
isdn switch-type primary-ni  
!  
!
```

```

crypto pki trustpoint TP-self-signed-2136404343
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-2136404343
  revocation-check none
  rsakeypair TP-self-signed-2136404343
!
!
crypto pki certificate chain TP-self-signed-2136404343
  certificate self-signed 01
    3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
    31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
    69666963 6174652D 32313336 34303433 3433301E 170D3133 30363238 30323434
    30395A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
    4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D32 31333634
    30343334 3330819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281
    81008A62 2482A40E F3E36150 A11F7CF6 E55D8662 840672FC B7B4B14B F99D5AC2
    7937B260 F8AD2D0E 638655D8 F96C01E0 D2B22FAB C5112ED8 597D4507 7379A50B
    9EB48762 BA0474F0 E7883B86 E201B4CC 9EB100CA BDC889A9 3022615C CA2EAB77
    9DD51DEA 956D6BB9 792836C5 0556EA75 F6518C79 62C133F5 3881B58A C64E1EF0
    31570203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603
    551D2304 18301680 149B0E84 AEF12589 6E46FB18 38FE80DC 7971F815 A5301D06
    03551D0E 04160414 9B0E84AE F125896E 46FB1838 FE80DC79 71F815A5 300D0609
    2A864886 F70D0101 05050003 81810088 EC6CD455 80619269 7FAA646B 7155B671
    69AA31AF 486EF9C7 BA2C0BAC D956DF54 C44DAD1D 8F8849EC 5B1A5889 3EC6F3E9
    112A5801 543D9927 82F46CCB 59C5DC90 3F3A8FE1 11079927 E7A651B3 3789C69C
    5D9EE330 CD1493A9 EF6DFBE5 5161A7C9 885CF68A 5EB8FDA5 4DC3F56C 085513EB
    FACC6B6D 9A6B5D4D 978DD68E 4671D6
  quit
voice-card 0
  voice-service dsp-reservation 30
  dsp services dspfarm
!
!
!
voice service voip
  no ip address trusted authenticate
  mode border-element
  allow-connections h323 to h323
  allow-connections h323 to sip
  allow-connections sip to h323
  allow-connections sip to sip
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
  sip
    bind control source-interface GigabitEthernet0/0.200
    bind media source-interface GigabitEthernet0/0.200
    pass-thru content sdp
!
voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 ilbc
!
voice class h323 1
  h225 timeout tcp establish 3
!
!
!
!

```

```

voice translation-rule 1
  rule 1 /.*\ (2...\$\) / /\1/
!
voice translation-rule 2
  rule 1 // /\0/ type international international
!
!
voice translation-profile PREFIX_PLUS
  translate calling 2
!
voice translation-profile STRIP
  translate called 1
!
!
!
license udi pid CISCO2911/K9 sn FTX1726AL7J
hw-module pvdn 0/0
!
hw-module pvdn 0/1
!
!
!
username cisco secret 4 tnhtc92DXBhelxjYk8LWJrPV36S2i4ntXrpb4RFmfqY
!
redundancy
!
!
controller T1 0/0/0
  cablelength long 0db
  pri-group timeslots 1-3,24
!
!
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 182
  class AutoQoS-VoIP-Control-Trust
    bandwidth percent 5
  class class-default
    fair-queue
!
!
!
!
!
interface Loopback0
  ip address 10.10.32.1 255.255.255.255
!
interface Embedded-Service-Engine0/0
  no ip address
  shutdown
!

```



```

interface GigabitEthernet0/0
  description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
  no ip address
  duplex auto
  speed auto
!
interface GigabitEthernet0/0.100
  encapsulation dot1Q 100
  ip address 192.168.1.77 255.255.255.0
!
interface GigabitEthernet0/0.200
  encapsulation dot1Q 200
  ip address 10.10.120.1 255.255.255.0
  ip helper-address 192.168.1.71
  h323-gateway voip interface
  h323-gateway voip bind srcaddr 10.10.120.1
!
interface GigabitEthernet0/0.300
  encapsulation dot1Q 300 native
  ip address 10.10.130.1 255.255.255.0
!
interface GigabitEthernet0/1
  no ip address
  shutdown
  duplex auto
  speed auto
!
interface GigabitEthernet0/2
  no ip address
  shutdown
  duplex auto
  speed auto
!
interface Serial0/0/0:23
  no ip address
  encapsulation hdlc
  isdn switch-type primary-ni
  isdn incoming-voice voice
  isdn outgoing display-ie
  isdn outgoing ie redirecting-number
  no cdp enable
!
interface Serial0/1/0
  no ip address
  encapsulation frame-relay IETF
!
interface Serial0/1/0.1 point-to-point
  ip address 10.10.33.1 255.255.255.128
  frame-relay interface-dlci 102
!
interface Serial0/1/0.2 point-to-point
  bandwidth 1544
  frame-relay interface-dlci 103 ppp Virtual-Template200
  class AutoQoS-FR-Se0/1/0-103
  auto qos voip trust fr-atm
!
interface Virtual-Template200

```

```

bandwidth 768
ip address 10.10.33.129 255.255.255.128
ppp multilink
ppp multilink interleave
ppp multilink fragment delay 10
service-policy output AutoQoS-Policy-Trust
!
router ospf 100
 network 0.0.0.0 255.255.255.255 area 0
!
ip forward-protocol nd
!
ip http server
ip http access-class 23
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
ip route 0.0.0.0 0.0.0.0 192.168.1.1
!
!
map-class frame-relay AutoQoS-FR-Se0/1/0-103
 frame-relay cir 768000
 frame-relay bc 7680
 frame-relay be 0
 frame-relay mincir 768000
access-list 23 permit 10.10.10.0 0.0.0.7
!
!
!
control-plane
!
!
voice-port 0/0/0:23
 translation-profile incoming STRIP
!
!
!
!
ccm-manager music-on-hold
!
!
mgcp profile default
!
sccp local GigabitEthernet0/0.200
sccp ccm 192.168.1.72 identifier 1 version 7.0
sccp ccm 192.168.1.71 identifier 2 version 7.0
sccp
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate ccm 2 priority 2
 associate profile 1 register HQ-VID-CFB
!
dspfarm profile 1 conference video homogeneous
 codec g729br8
 codec g729r8

```

```

codec g729abr8
codec g729ar8
codec g711alaw
codec g711ulaw
codec ilbc
codec h264 cif frame-rate 30 bitrate 320kbps
maximum conference-participants 8
maximum sessions 4
associate application SCCP
!
dial-peer voice 1 pots
    incoming called-number .
    direct-inward-dial
!
dial-peer voice 2 voip
    destination-pattern 2...$
    session target ipv4:192.168.1.72
    voice-class codec 1
    voice-class h323 1
    dtmf-relay h245-alphanumeric
    no vad
!
dial-peer voice 3 voip
    preference 1
    destination-pattern 2...$
    session target ipv4:192.168.1.71
    voice-class codec 1
    voice-class h323 1
    dtmf-relay h245-alphanumeric
    no vad
!
dial-peer voice 4 pots
    translation-profile outgoing PREFIX_PLUS
    destination-pattern .T
    port 0/0/0:23
!
dial-peer voice 5 voip
    session protocol sipv2
    incoming called-number [23]...$
    dtmf-relay rtp-nte sip-notify sip-kpml
    codec transparent
    no vad
!
dial-peer voice 6 voip
    destination-pattern 3...$
    session protocol sipv2
    session target ipv4:10.10.150.71
    dtmf-relay rtp-nte sip-notify sip-kpml
    codec transparent
    no vad
!
dial-peer voice 7 voip
    destination-pattern 2...$
    session protocol sipv2
    session target ipv4:192.168.1.72
    dtmf-relay rtp-nte sip-notify sip-kpml
    codec transparent

```

```

    no vad
!
dial-peer voice 8 voip
  preference 1
  destination-pattern 2...$
  session protocol sipv2
  session target ipv4:192.168.1.71
  dtmf-relay rtp-nte sip-notify sip-kpml
  codec transparent
  no vad
!
dial-peer voice 9 voip
  destination-pattern +1408[2-9].....$
  session protocol sipv2
  session target ipv4:10.10.150.71
  dtmf-relay sip-kpml
  codec transparent
  no vad
!
!
!
!
gatekeeper
  shutdown
!
!
!
line con 0
  exec-timeout 0 0
  logging synchronous
  login local
line aux 0
line 2
  no activation-character
  no exec
  transport preferred none
  transport input all
  transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
  stopbits 1
line vty 0 4
  access-class 23 in
  login local
  length 0
  transport input telnet ssh
line vty 5 15
  access-class 23 in
  privilege level 15
  login local
  transport input telnet ssh
!
scheduler allocate 20000 1000
ntp source Loopback0
ntp server 192.168.1.78
!
end

```

Final Configuration of BR1

BR1#**show run**

Building configuration...

```
Current configuration : 5466 bytes
!
! No configuration change since last restart
version 15.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname BR1
!
boot-start-marker
boot-end-marker
!
!
card type t1 0 0
logging buffered 51200 warnings
!
no aaa new-model
clock timezone EST -5 0
clock summer-time EDT recurring
network-clock-participate wic 0
!
ip cef
!
!
!
ip dhcp excluded-address 10.10.10.1
ip dhcp excluded-address 10.10.140.1 10.10.140.9
ip dhcp excluded-address 10.10.140.21 10.10.140.254
!
ip dhcp pool IPPHONES
    network 10.10.140.0 255.255.255.0
    option 150 ip 10.10.150.71
    default-router 10.10.140.1
!
!
!
no ip domain lookup
ip domain name kwtrain.com
no ipv6 cef
multilink bundle-name authenticated
!
!
!
!
!
isdn switch-type primary-ni
!
!
crypto pki trustpoint TP-self-signed-1996982465
    enrollment selfsigned
```

```

subject-name cn=IOS-Self-Signed-Certificate-1996982465
revocation-check none
rsakeypair TP-self-signed-1996982465
!
!
crypto pki certificate chain TP-self-signed-1996982465
certificate self-signed 01
  3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
  31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
  69666963 6174652D 31393936 39383234 3635301E 170D3133 30363238 30323331
  35395A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D31 39393639
  38323436 3530819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281
  81009EC2 17DFBEF6 DB9E0D6D 06B06014 0E0BFAE3 95FFB6E3 96E4A204 4D6CA7C6
  2E83BD2A 62B591D6 7CE5624F 0F8C96AF 8989E38A 2D56460D 0B43F94C 68078642
  76650163 495038E1 7844067C 66BDB7E0 2490CE87 C9387A9F EDB39B8C F685703D
  E425CC75 285102BA FF29EA27 3F2B2249 73388397 05957F5D 6AB2DD15 6CF609CD
  1ACB0203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603
  551D2304 18301680 14F1E5A1 F95B139B 5876237C 0A10E112 C3F4AF6D 70301D06
  03551D0E 04160414 F1E5A1F9 5B139B58 76237C0A 10E112C3 F4AF6D70 300D0609
  2A864886 F70D0101 05050003 8181002D 0A22176D 78D7501D C14339E1 89B6ACEC
  6DED275E 9314EE28 FD90D95A 2A84CA03 BAF17F7 FCF0FE41 32AA2D16 7820A5BE
  5D9872A5 4443184D 892ADF50 78C07ADC 6D4F6AEE 7A969EBE 9ADC78A9 90F86569
  8D7B9E90 E7B14315 3F6501A9 C0C2D2F5 9D2C4168 8B708EBB CBDAE81B E33736EA
  6286940A FE9680A4 6F6E021E C98E1C
quit
voice-card 0
!
!
!
!
!
!
!
!
license udi pid CISCO2911/K9 sn FTX1726AL6K
hw-module pvdm 0/0
!
hw-module pvdm 0/1
!
!
!
username cisco secret 4 tnhtc92DXBhelxjYk8LWJrPV36S2i4ntXrpb4RFmfqY
!
redundancy
!
!
controller T1 0/0/0
  cablelength long 0db
  pri-group timeslots 1-3,24 service mgcp
!
controller T1 0/0/1
  cablelength long 0db
!
!
!
!

```

```

!
interface Loopback0
 ip address 10.10.32.2 255.255.255.255
!
interface Embedded-Service-Engine0/0
 no ip address
 shutdown
!
interface GigabitEthernet0/0
 description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface GigabitEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface GigabitEthernet0/2
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface Serial0/0/0:23
 no ip address
 encapsulation hdlc
 isdn switch-type primary-ni
 isdn incoming-voice voice
 isdn bind-l3 ccm-manager
 no cdp enable
!
interface Serial0/1/0
 no ip address
 encapsulation frame-relay IETF
!
interface Serial0/1/0.1 point-to-point
 ip address 10.10.33.2 255.255.255.128
 frame-relay interface-dlci 201
!
interface FastEthernet0/2/0
 switchport access vlan 500
 switchport voice vlan 400
 no ip address
 spanning-tree portfast
!
interface FastEthernet0/2/1
 switchport access vlan 500
 switchport voice vlan 400
 no ip address
 spanning-tree portfast
!
interface FastEthernet0/2/2
 no ip address

```

```

!
interface FastEthernet0/2/3
  switchport access vlan 500
  no ip address
!
interface Vlan1
  no ip address
!
interface Vlan400
  ip address 10.10.140.1 255.255.255.0
!
interface Vlan500
  ip address 10.10.150.1 255.255.255.0
!
router ospf 100
  network 0.0.0.0 255.255.255.255 area 0
!
ip forward-protocol nd
!
ip http server
ip http access-class 23
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
!
access-list 23 permit 10.10.10.0 0.0.0.7
cdp timer 5
!
!
!
control-plane
!
!
voice-port 0/0/0:23
!
!
!
!
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config server 10.10.150.71
!
mgcp
mgcp call-agent 10.10.150.71 2427 service-type mgcp version 0.1
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability pre-package
no mgcp package-capability res-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 inhibit
mgcp bind control source-interface Vlan400
mgcp bind media source-interface Vlan400

```



```

!
mgcp profile default
!
!
!
!
!
gatekeeper
shutdown
!
!
!
line con 0
exec-timeout 0 0
logging synchronous
login local
line aux 0
line 2
no activation-character
no exec
transport preferred none
transport input all
transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
stopbits 1
line vty 0 4
access-class 23 in
privilege level 15
login local
transport input telnet ssh
line vty 5 15
access-class 23 in
privilege level 15
login local
transport input telnet ssh
!
scheduler allocate 20000 1000
ntp server 10.10.32.1
!
end

```

Final Configuration of BR2

BR2#**show run**

Building configuration...

Current configuration : 11131 bytes

```

!
! Last configuration change at 01:53:42 TKY Wed Oct 28 2015
version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname BR2
!

```

```

boot-start-marker
boot-end-marker
!
!
card type e1 0 0
!
no aaa new-model
!
clock timezone TKY 9 0
network-clock-participate wic 0
network-clock-select 1 E1 0/0/0
!
no ipv6 cef
ip source-route
ip cef
!
!
!
!
ip dhcp pool BR2PH1
  host 10.10.160.10 255.255.255.0
  client-identifier 010c.d996.9026.a3
  default-router 10.10.160.1
  option 150 ip 10.10.32.3
!
ip dhcp pool BR2PH2
  host 10.10.160.11 255.255.255.0
  client-identifier 0100.1c58.fb76.01
  default-router 10.10.160.1
  option 150 ip 10.10.32.3
!
!
no ip domain lookup
!
multilink bundle-name authenticated
!
!
!
!
!
isdn switch-type primary-net5
!
crypto pki token default removal timeout 0
!
!
voice-card 0
  dsp services dspfarm
!
!
!
voice service voip
  allow-connections h323 to h323
  allow-connections h323 to sip
  allow-connections sip to h323
  allow-connections sip to sip
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
  sip
    bind control source-interface Loopback0

```

```

    bind media source-interface Loopback0
    registrar server
!
voice class codec 1
    codec preference 1 ilbc
    codec preference 2 g711ulaw
!
voice class h323 1
    h225 timeout tcp establish 3
!
!
voice register global
    mode cme
    source-address 10.10.32.3 port 5060
    bandwidth video tias-modifier 512000 negotiate end-to-end
    max-dn 20
    max-pool 10
    load 9971 sip9971.9-2-2SR1-9.loads
    timezone 44
    time-format 24
    voicemail 4500
    tftp-path flash:
    create profile sync 0000663007350425
    ntp-server 10.10.32.1 mode directedbroadcast
    camera
    video
!
voice register dn 1
    number 4001
    call-forward b2bua busy 4500
    call-forward b2bua noan 4500 timeout 10
    name BR2 Phone 1
    huntstop channel 1
!
voice register pool 1
    id mac OCD9.9690.26A3
    type 9971
    number 1 dn 1
    cor outgoing SIP default
    dtmf-relay sip-kpml
    description +8144444001
    codec g711ulaw
    no vad
!
!
!
voice translation-rule 1
    rule 1 /.*\ (4...\$)\ / /\1/
!
voice translation-rule 2
    rule 1 /^4...\$/ /4444\0/ type any subscriber plan any isdn
    rule 2 // // type any unknown plan any isdn
!
voice translation-rule 3
    rule 1 /^4...\$/ /4444\0/ type any subscriber plan any isdn
    rule 2 // // type any subscriber plan any isdn
!

```

```

voice translation-rule 4
  rule 1 /^4...$/ /+814444\0/ type any international plan any isdn
  rule 2 // // type any international plan any isdn
!
!
voice translation-profile 999
  translate calling 2
  translate called 2
!
voice translation-profile INTL
  translate calling 4
  translate called 4
!
voice translation-profile LOCAL
  translate calling 3
  translate called 3
!
voice translation-profile STRIP
  translate called 1
!
!
license udi pid CISCO2921/K9 sn FTX1540AHDV
hw-module pvdn 0/0
!
hw-module sm 1
!
!
!
!
redundancy
!
!
!
!
controller E1 0/0/0
  pri-group timeslots 1-3,16
!
controller E1 0/0/1
!
!
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 176
  class AutoQoS-VoIP-Control-Trust
    bandwidth percent 5
  class class-default
    fair-queue
!
!
!

```

```

!
!
!
!
!
interface Loopback0
 ip address 10.10.32.3 255.255.255.255
!
interface Embedded-Service-Engine0/0
 no ip address
 shutdown
!
interface GigabitEthernet0/0
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface GigabitEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface GigabitEthernet0/2
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface Serial0/0/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5
 isdn incoming-voice voice
 isdn outgoing display-ie
 isdn outgoing ie redirecting-number
 no cdp enable
!
interface Serial0/1/0
 no ip address
 encapsulation frame-relay IETF
 frame-relay traffic-shaping
!
interface Serial0/1/0.1 point-to-point
 bandwidth 1544
 frame-relay interface-dlci 301 ppp Virtual-Template200
 class AutoQoS-FR-Se0/1/0-301
 auto qos voip trust fr-atm
!
interface FastEthernet0/2/0
 switchport access vlan 700
 switchport voice vlan 600
 no ip address
 spanning-tree portfast
!
interface FastEthernet0/2/1

```

```

switchport access vlan 700
switchport voice vlan 600
no ip address
spanning-tree portfast
!
interface FastEthernet0/2/2
no ip address
!
interface FastEthernet0/2/3
no ip address
!
interface SM1/0
ip unnumbered Vlan600
service-module ip address 10.10.160.2 255.255.255.0
!Application: CUE Running on SM
service-module ip default-gateway 10.10.160.1
!
interface SM1/1
no ip address
shutdown
!
interface Virtual-Template200
bandwidth 768
ip address 10.10.33.130 255.255.255.128
ppp multilink
ppp multilink interleave
ppp multilink fragment delay 10
service-policy output AutoQoS-Policy-Trust
!
interface Vlan1
no ip address
!
interface Vlan600
ip address 10.10.160.1 255.255.255.0
h323-gateway voip interface
h323-gateway voip bind srcaddr 10.10.160.1
!
interface Vlan700
ip address 10.10.170.1 255.255.255.0
!
!
router ospf 100
network 0.0.0.0 255.255.255.255 area 0
!
ip forward-protocol nd
!
no ip http server
no ip http secure-server
!
ip route 10.10.160.2 255.255.255.255 SM1/0
!
!
map-class frame-relay AutoQoS-FR-Se0/1/0-301
frame-relay cir 768000
frame-relay bc 7680
frame-relay be 0
frame-relay mincir 768000

```

```

!
cdp timer 5
!
!
!
!
tftp-server flash:phones/9971/kern9971.9-2-2SR1-9.sebn alias kern9971.9-2-
2SR1-9.sebn
tftp-server flash:phones/9971/rootfs9971.9-2-2SR1-9.sebn alias rootfs9971.9-
2-2SR1-9.sebn
tftp-server flash:phones/9971/sboot9971.031610R1-9-2-2SR1-9.sebn alias
sboot9971.031610R1-9-2-2SR1-9.sebn
tftp-server flash:phones/9971/skern9971.022809R2-9-2-2SR1-9.sebn alias
skern9971.022809R2-9-2-2SR1-9.sebn
tftp-server flash:phones/9971/sip9971.9-2-2SR1-9.loads alias sip9971.9-2-
2SR1-9.loads
tftp-server flash:phones/7945-7965/apps45.9-2-1TH1-13.sbn alias apps45.9-2-
1TH1-13.sbn
tftp-server flash:phones/7945-7965/cnu45.9-2-1TH1-13.sbn alias cnu45.9-2-
1TH1-13.sbn
tftp-server flash:phones/7945-7965/cvm45sccp.9-2-1TH1-13.sbn alias
cvm45sccp.9-2-1TH1-13.sbn
tftp-server flash:phones/7945-7965/dsp45.9-2-1TH1-13.sbn alias dsp45.9-2-
1TH1-13.sbn
tftp-server flash:phones/7945-7965/jar45sccp.9-2-1TH1-13.sbn alias
jar45sccp.9-2-1TH1-13.sbn
tftp-server flash:phones/7945-7965/SCCP45.9-2-1S.loads alias SCCP45.9-2-
1S.loads
tftp-server flash:phones/7945-7965/term45.default.loads alias
term45.default.loads
tftp-server flash:phones/7945-7965/term65.default.loads alias
term65.default.loads
tftp-server flash:phones/9971/dkern9971.100609R2-9-2-2SR1-9.sebn alias
dkern9971.100609R2-9-2-2SR1-9.sebn
!
control-plane
!
!
voice-port 0/0/0:15
 translation-profile incoming STRIP
!
!
!
mgcp profile default
!
sccp local Loopback0
sccp ccm 10.10.32.3 identifier 1 version 7.0
sccp
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 1 register BR2-CFB
!
dspfarm profile 1 conference
 codec g729br8
 codec g729r8
 codec g729abr8

```

```

codec g729ar8
codec g711alaw
codec g711ulaw
codec ilbc
maximum sessions 1
associate application SCCP
!
dial-peer cor custom
  name SIP
  name SCCP
!
!
dial-peer cor list SIP
  member SIP
!
dial-peer cor list SCCP
  member SCCP
!
!
dial-peer voice 1 pots
  incoming called-number .
  direct-inward-dial
!
dial-peer voice 2 pots
  translation-profile outgoing 999
  destination-pattern 999$
  no digit-strip
  port 0/0/0:15
!
dial-peer voice 3 pots
  translation-profile outgoing LOCAL
  destination-pattern 9[1-9].....$
  port 0/0/0:15
!
dial-peer voice 4 pots
  translation-profile outgoing INTL
  destination-pattern 900T
  port 0/0/0:15
!
dial-peer voice 5 voip
  corlist outgoing SCCP
  destination-pattern 2...$
  session protocol sipv2
  session target ipv4:192.168.1.72
  voice-class codec 1
  dtmf-relay sip-kpml
  no vad
!
dial-peer voice 6 voip
  corlist outgoing SCCP
  preference 1
  destination-pattern 2...$
  session protocol sipv2
  session target ipv4:192.168.1.71
  voice-class codec 1
  dtmf-relay sip-kpml
  no vad

```



```

!
dial-peer voice 7 voip
  corlist outgoing SCCP
  destination-pattern 3...$
  session protocol sipv2
  session target ipv4:10.10.150.71
  voice-class codec 1
  dtmf-relay sip-kpml
  no vad
!
dial-peer voice 4500 voip
  destination-pattern 45..
  session protocol sipv2
  session target ipv4:10.10.160.2
  dtmf-relay sip-notify
  codec g711ulaw
  no vad
!
dial-peer voice 8 voip
  corlist outgoing SIP
  destination-pattern 2...$
  rtp payload-type cisco-codec-fax-ack 98
  rtp payload-type cisco-codec-video-h264 97
  session protocol sipv2
  session target ipv4:192.168.1.72
  dtmf-relay sip-kpml
  codec transparent
  no vad
!
dial-peer voice 9 voip
  corlist outgoing SIP
  preference 1
  destination-pattern 2...$
  rtp payload-type cisco-codec-fax-ack 98
  rtp payload-type cisco-codec-video-h264 97
  session protocol sipv2
  session target ipv4:192.168.1.71
  dtmf-relay sip-kpml
  codec transparent
  no vad
!
dial-peer voice 10 voip
  corlist outgoing SIP
  destination-pattern 3...$
  rtp payload-type cisco-codec-fax-ack 98
  rtp payload-type cisco-codec-video-h264 97
  session protocol sipv2
  session target ipv4:10.10.150.71
  dtmf-relay sip-kpml
  codec transparent
  no vad
!
dial-peer voice 11 voip
  corlist incoming SIP
  rtp payload-type cisco-codec-fax-ack 98
  rtp payload-type cisco-codec-video-h264 97
  session protocol sipv2

```

```

incoming called-number 4001
dtmf-relay sip-kpml
codec transparent
no vad
!
dial-peer voice 12 voip
corlist incoming SCCP
session protocol sipv2
incoming called-number 4002
voice-class codec 1
dtmf-relay sip-kpml
no vad
!
!
sip-ua
mwi-server ipv4:10.10.160.2 expires 3600 port 5060 transport udp unsolicited
!
!
!
gatekeeper
shutdown
!
!
telephony-service
sdspfarm units 2
sdspfarm tag 1 BR2-CFB
conference hardware
max-ephones 10
max-dn 20 no-reg both
ip source-address 10.10.32.3 port 2000
system message CUCME
load 7965 SCCP45.9-2-1S.loads
time-zone 44
date-format yy-mm-dd
voicemail 4500
mwi relay
max-conferences 8 gain -6
transfer-system full-consult
create cnf-files version-stamp 7960 Oct 28 2015 01:53:47
!
!
ephone-dn 1 octo-line
number 4002 no-reg both
description +8144444002
name BR2 Phone 2
call-forward busy 4500
call-forward noan 4500 timeout 10
corlist outgoing SCCP
huntstop channel 1
mwi sip
!
!
ephone-dn 2 octo-line
number 4444
conference ad-hoc
!
!

```

```

ephone-dn 3
  number 4510.... no-reg both
  mwi on
!
!
ephone-dn 4
  number 4511.... no-reg both
  mwi off
!
!
ephone 1
  privacy-button
  device-security-mode none
  mac-address 001C.58FB.7601
  codec ilbc
  type 7965
  button 1:1
!
!
!
!
line con 0
  exec-timeout 0 0
  logging synchronous
line aux 0
line 2
  no activation-character
  no exec
  transport preferred none
  transport input all
  transport output lat pad telnet rlogin lapb-ta mop udptn v120 ssh
  stopbits 1
line 67
  no activation-character
  no exec
  transport preferred none
  transport input all
  transport output lat pad telnet rlogin lapb-ta mop udptn v120 ssh
  stopbits 1
line vty 0 4
  login
  length 0
  transport input all
!
scheduler allocate 20000 1000
ntp server 10.10.32.1
end

```

Final Configuration of CUE Module

CUE# **show run**

Generating configuration:

```
clock timezone Asia/Tokyo
```

```

hostname CUE

line console

system language preferred "en_US"

ntp server 192.168.1.78 prefer

software download server url "ftp://127.0.0.1/ftp" credentials hidden
"6u/dKTN/hsEuSAEfw40XlF2eFHnZfyUTSd8ZZNgd+Y9J3xlk2B35j0nfGWTYHfmPSd8ZZNgd+Y9J
3xlk2B35j0nfGWTYHfmPSd8ZZNgd+Y9J3xlk2B35j0nfGWTYHfmP"

log trace local enable

site name local
end site

license agent max-sessions 9

privilege vm-imap create
privilege local-broadcast create
privilege broadcast create
privilege ManagePrompts create
privilege manage-passwords create
privilege manage-users create
privilege ViewRealTimeReports create
privilege ViewPrivateList create
privilege ManagePublicList create
privilege ViewHistoricalReports create

groupname Broadcasters create

username br2phone1 create
username administrator create
username br2phone2 create

privilege vm-imap description "Privilege to manage personal voicemail via
IMAP client"
privilege local-broadcast description "Privilege to send local broadcast
messages"
privilege broadcast description "Privilege to send local or remote broadcast
messages"
privilege ManagePrompts description "Privilege to create, modify, or delete
system prompts"
privilege manage-passwords description "Privilege to reset user passwords"
privilege manage-users description "Privilege to create, modify, and delete
users and groups"
privilege ViewRealTimeReports description "Privilege to view realtime
reports"
privilege ViewPrivateList description "Privilege to view private list"
privilege ManagePublicList description "Privilege to manage public lists"
privilege ViewHistoricalReports description "Privilege to view historical
reports"
privilege vm-imap operation voicemail.imap.user
privilege local-broadcast operation system.debug
privilege local-broadcast operation broadcast.local
privilege broadcast operation broadcast.remote

```

```

privilege broadcast operation system.debug
privilege broadcast operation broadcast.local
privilege ManagePrompts operation system.debug
privilege ManagePrompts operation prompt.modify
privilege manage-passwords operation user.pin
privilege manage-passwords operation user.password
privilege manage-passwords operation system.debug
privilege manage-users operation user.pin
privilege manage-users operation user.password
privilege manage-users operation system.debug
privilege manage-users operation user.configuration
privilege manage-users operation user.notification
privilege manage-users operation user.remote
privilege manage-users operation user.mailbox
privilege manage-users operation group.configuration
privilege ViewRealTimeReports operation report.realtime
privilege ViewPrivateList operation voicemail.lists.private.view
privilege ManagePublicList operation system.debug
privilege ManagePublicList operation voicemail.lists.public
privilege ViewHistoricalReports operation report.historical.view

groupname Administrators member br2phone1
groupname Administrators member administrator
groupname Broadcasters privilege broadcast

username br2phone1 phonenumber "4001"
username br2phone2 phonenumber "4002"

restriction msg-notification create
restriction msg-notification min-digits 1
restriction msg-notification max-digits 30
restriction msg-notification dial-string preference 1 pattern * allowed

backup server url "ftp://127.0.0.1/ftp" credentials hidden
"EWlTygcMhYmjazXhE/VNXHCkplVV4KjesCbDaLa4f14WLSPFvvlrWUnfGWTYHfmPSd8ZZNgd+Y9J
3x1k2B35j0nfGWTYHfmPSd8ZZNgd+Y9J3x1k2B35j0nfGWTYHfmP"

calendar biz-schedule systemschedule
  open day 1 from 00:00 to 24:00
  open day 2 from 00:00 to 24:00
  open day 3 from 00:00 to 24:00
  open day 4 from 00:00 to 24:00
  open day 5 from 00:00 to 24:00
  open day 6 from 00:00 to 24:00
  open day 7 from 00:00 to 24:00
end schedule

ccn application autoattendant aa
  description "autoattendant"
  enabled
  maxsessions 32
  script "aa.aef"
  parameter "dialByExtnAnytime" "false"
  parameter "busOpenPrompt" "AABusinessOpen.wav"
  parameter "dialByExtnAnytimeInputLength" "4"
  parameter "operExtn" ""
  parameter "welcomePrompt" "AAWelcome.wav"

```

```

parameter "disconnectAfterMenu" "false"
parameter "dialByFirstName" "false"
parameter "busClosedPrompt" "AABusinessClosed.wav"
parameter "allowExternalTransfers" "false"
parameter "holidayPrompt" "AAHolidayPrompt.wav"
parameter "businessSchedule" "systemschedule"
parameter "MaxRetry" "3"
end application

ccn application ciscomwiapplication aa
description "ciscomwiapplication"
enabled
maxsessions 32
script "setmwi.aef"
parameter "CallControlGroupID" "0"
parameter "strMWI_OFF_DN" "4511"
parameter "strMWI_ON_DN" "4510"
end application

ccn application msgnotification aa
description "msgnotification"
enabled
maxsessions 32
script "msgnotify.aef"
parameter "logoutUri" "http://localhost/voicemail/vxmlscripts/mbxLogout.jsp"
parameter "DelayBeforeSendDTMF" "1"
end application

ccn application promptmgmt aa
description "promptmgmt"
enabled
maxsessions 1
script "promptmgmt.aef"
parameter "appManagementScript" ""
end application

ccn application voicemail aa
description "voicemail"
enabled
maxsessions 32
script "voicebrowser.aef"
parameter "uri" "http://localhost/voicemail/vxmlscripts/login.vxml"
parameter "logoutUri" "http://localhost/voicemail/vxmlscripts/mbxLogout.jsp"
end application

ccn engine
maxsteps -1
end engine

ccn reporting historical
database local
description "se-10-10-160-2"
end reporting

ccn subsystem sip
gateway address "10.10.32.3"
dtmf-relay sip-notify

```

```

mwi sip unsolicited
mwi envelope-info
transfer-mode blind
end subsystem

ccn trigger http urlname msgnotifytrg
application "msgnotification"
enabled
maxsessions 2
end trigger

ccn trigger http urlname mwiapp
application "ciscomwiapplication"
enabled
maxsessions 1
end trigger

ccn trigger sip phonenummer 4500
application "voicemail"
enabled
maxsessions 6
end trigger

ccn trigger sip phonenummer 4550
application "autoattendant"
enabled
maxsessions 2
end trigger

ccn trigger sip phonenummer 4551
application "promptmgmt"
enabled
maxsessions 1
end trigger

service phone-authentication
end phone-authentication

service voiceview
enable
end voiceview

voicemail broadcast recording time 300
voicemail default messagesize 240
voicemail notification restriction msg-notification
voicemail mailbox owner "br2phone1" size 86400
end mailbox

voicemail mailbox owner "br2phone2" size 86400
end mailbox

end

```

PSTN Configuration

```
PSTN#show run
```

Building configuration...

```
Current configuration : 9752 bytes
!
! No configuration change since last restart
version 15.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname PSTN
!
boot-start-marker
boot-end-marker
!
!
card type t1 0 0
card type e1 0 1
logging buffered 51200 warnings
!
no aaa new-model
network-clock-participate wic 0
network-clock-participate wic 1
no network-clock-participate wic 2
!
ip cef
!
!
!
ip dhcp excluded-address 10.10.10.1
ip dhcp excluded-address 10.10.200.1 10.10.200.9
ip dhcp excluded-address 10.10.200.20 10.10.200.254
ip dhcp excluded-address 10.10.201.1 10.10.201.9
ip dhcp excluded-address 10.10.201.20 10.10.201.254
!
ip dhcp pool PSTN1
 network 10.1.200.0 255.255.255.0
 default-router 10.1.200.229
 option 150 ip 10.1.200.229
!
ip dhcp pool PSTN2
 network 10.1.201.0 255.255.255.0
 default-router 10.1.201.229
 option 150 ip 10.1.200.229
!
!
!
no ip domain lookup
ip domain name yourdomain.com
no ipv6 cef
multilink bundle-name authenticated
!
frame-relay switching
!
!
!
```



```

isdn switch-type primary-net5
!
!
crypto pki trustpoint TP-self-signed-2614227737
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-2614227737
  revocation-check none
  rsakeypair TP-self-signed-2614227737
!
!
crypto pki certificate chain TP-self-signed-2614227737
  certificate self-signed 01
    3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
    31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
    69666963 6174652D 32363134 32323737 3337301E 170D3133 30363237 31363332
    32375A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
    4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D32 36313432
    32373733 3730819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281
    8100C0FC CE7D46A0 C00064FF 130DEBB8 39A8C098 6930748B 8F49FEAE 5CBCE56A
    80BEC6AE 1F014CA0 DE315E57 0D6B683A 35623964 A030F40B EC9F3FBB 133E541C
    5A239627 19048F3B EDCB8CE4 918BE484 DAE449E FB76F7FD 60B89E9A CC752FCB
    D9F6B41F 98D76727 39DB59CF 5AE171E2 A8BB1DF0 10E6319A 9A98D2F2 6D014AC6
    D0E70203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603
    551D2304 18301680 141DD495 6B6F3EEB 425A1DFB 822A6DDB 162762D6 A2301D06
    03551D0E 04160414 1DD4956B 6F3EEB42 5A1DFB82 2A6DDB16 2762D6A2 300D0609
    2A864886 F70D0101 05050003 8181000C C61A7D45 ED9A73F8 7B32FDAA 2540DBB8
    9359DEBE F7CB8E9A 75C80BA0 5CC03372 FBE24C0C 1D303EAF 1F34C903 A6584CFC
    18A52377 9846CB10 2EE9810E 596A9FEE D2029703 C8F084FC 118E5BAF 34811E87
    2C1D825D 55D8A0DE ED9A9A23 E60A36A8 0FC03D48 2076D191 FC48C496 94951F6B
    A9ADA15F ABF7A54E 6B1C3D21 BFCFB4
  quit
voice-card 0
!
!
!
voice service voip
  allow-connections h323 to sip
  allow-connections sip to h323
  allow-connections sip to sip
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
  sip
    bind control source-interface gig0/0
    bind media source-interface gig0/0
    registrar server
!
!
voice register global
  mode cme
  source-address 10.1.200.229 port 5060
  bandwidth video tias-modifier 512000 negotiate end-to-end
  max-dn 20
  max-pool 10
  load 9971 sip9971.9-2-2SR1-9.loads
  tftp-path flash:
  create profile sync 0017344407105603
  camera
  video

```

```

!
voice register dn 1
  number 6065555
!
voice register pool 1
  id mac 001D.70FC.8BF3
  type 9971
  number 1 dn 1
  dtmf-relay sip-notify
  codec g711ulaw
!
!
!
voice translation-rule 1
  rule 1 /\(^001\)\(.*\)/ /\2/
  rule 2 /\(^1\)\(.*\)/ /\2/
  rule 3 /\(^00\)\(.*\)/ /\2/
  rule 4 /\(^011\)\(.*\)/ /\2/
!
voice translation-rule 2
  rule 1 /^408/ /1&/ type any international
  rule 2 /^81\(.*\)/ /\1/ type any subscriber
  rule 3 /^4444.* /&/ type any subscriber
  rule 4 /^859/ /1&/ type any international
!
voice translation-rule 3
  rule 1 /^859\(.*\)/ /\1/ type any subscriber
  rule 2 /^408/ /&/ type any national
  rule 3 /^81/ /&/ type any international
  rule 4 /^333/ /&/ type any subscriber
  rule 5 /^4444/ /81&/ type any international
!
voice translation-rule 4
  rule 1 /^408\(.*\)/ /\1/ type any subscriber
  rule 2 /^859/ /&/ type any national
  rule 3 /^81/ /&/ type any international
  rule 4 /^222/ /\0/ type any subscriber
  rule 5 /^4444/ /81&/ type any international
!
!
voice translation-profile BR1
  translate calling 3
!
voice translation-profile BR2
  translate calling 2
!
voice translation-profile HQ
  translate calling 4
!
voice translation-profile PSTN
  translate called 1
!
!
!
license udi pid CISCO2911/K9 sn FTX1726AKML
hw-module pvdm 0/0
!

```

```

!
!
username cisco secret 4 tnhtc92DXBhelxjYk8LWJrPV36S2i4ntXrpb4RFmfqY
!
redundancy
!
!
controller T1 0/0/0
    clock source internal
    cablelength long 0db
    pri-group timeslots 1-3,24
!
controller T1 0/0/1
    clock source internal
    cablelength long 0db
    pri-group timeslots 1-3,24
!
controller E1 0/1/0
    clock source internal
    pri-group timeslots 1-10,16
!
!
!
!
!
interface Embedded-Service-Engine0/0
    no ip address
    shutdown
!
interface GigabitEthernet0/0
    description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
    ip address 192.168.1.78 255.255.255.0
    duplex auto
    speed auto
!
interface GigabitEthernet0/1
    ip address 10.1.200.229 255.255.255.0
    duplex auto
    speed auto
!
interface GigabitEthernet0/2
    ip address 10.1.201.229 255.255.255.0
    duplex auto
    speed auto
!
interface Serial0/0/0:23
    no ip address
    encapsulation hdlc
    isdn switch-type primary-ni
    isdn protocol-emulate network
    isdn incoming-voice voice
    isdn outgoing display-ie
    isdn outgoing ie redirecting-number
    no cdp enable
!
interface Serial0/0/1:23
    no ip address

```

```

encapsulation hdlc
isdn switch-type primary-ni
isdn protocol-emulate network
isdn incoming-voice voice
isdn outgoing display-ie
isdn outgoing ie redirecting-number
no cdp enable
!
interface Serial0/1/0:15
no ip address
encapsulation hdlc
isdn switch-type primary-net5
isdn protocol-emulate network
isdn incoming-voice voice
isdn outgoing display-ie
isdn outgoing ie redirecting-number
no cdp enable
!
interface Serial0/2/0
description FRSW to HQ
no ip address
encapsulation frame-relay IETF
clock rate 8064000
frame-relay lmi-type ansi
frame-relay intf-type dce
frame-relay route 102 interface Serial0/2/1 201
frame-relay route 103 interface Serial0/2/2 301
!
interface Serial0/2/1
no ip address
encapsulation frame-relay IETF
clock rate 8064000
frame-relay lmi-type ansi
frame-relay intf-type dce
frame-relay route 201 interface Serial0/2/0 102
!
interface Serial0/2/2
no ip address
encapsulation frame-relay IETF
clock rate 8064000
frame-relay lmi-type ansi
frame-relay intf-type dce
frame-relay route 301 interface Serial0/2/0 103
!
interface Serial0/2/3
no ip address
shutdown
clock rate 2016000
!
ip forward-protocol nd
!
ip http server
ip http access-class 23
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!

```

```

ip route 0.0.0.0 0.0.0.0 192.168.1.1
ip route 10.0.0.0 255.0.0.0 192.168.1.77
!
access-list 23 permit 10.10.10.0 0.0.0.7
!
!
tftp-server flash:dkern9971.100609R2-9-2-2SR1-9.sebn
tftp-server flash:kern9971.9-2-2SR1-9.sebn
tftp-server flash:rootfs9971.9-2-2SR1-9.sebn
tftp-server flash:sboot9971.031610R1-9-2-2SR1-9.sebn
tftp-server flash:sip9971.9-2-2SR1-9.loads
tftp-server flash:skern9971.022809R2-9-2-2SR1-9.sebn
tftp-server flash:term65.default.loads
tftp-server flash:term45.default.loads
tftp-server flash:SCCP45.9-2-1S.loads
tftp-server flash:jar45sccp.9-2-1TH1-13.sbn
tftp-server flash:dsp45.9-2-1TH1-13.sbn
tftp-server flash:cvm45sccp.9-2-1TH1-13.sbn
tftp-server flash:cnu45.9-2-1TH1-13.sbn
tftp-server flash:apps45.9-2-1TH1-13.sbn
!
control-plane
!
!
voice-port 0/0/0:23
 translation-profile incoming PSTN
 translation-profile outgoing HQ
!
voice-port 0/1/0:15
 translation-profile incoming PSTN
 translation-profile outgoing BR2
!
voice-port 0/0/1:23
 translation-profile incoming PSTN
 translation-profile outgoing BR1
!
!
!
!
!
mgcp profile default
!
!
dial-peer voice 11 pots
 description PSTN Phone
 destination-pattern 2222...
 port 0/0/0:23
 forward-digits all
!
dial-peer voice 1 pots
 incoming called-number .
 direct-inward-dial
!
dial-peer voice 10 pots
 destination-pattern 4082222...
 port 0/0/0:23

```

```

    forward-digits all
!
dial-peer voice 12 pots
    destination-pattern 8144444...
    port 0/1/0:15
    forward-digits 8
!
dial-peer voice 13 pots
    destination-pattern 8593333...
    port 0/0/1:23
    forward-digits all
!
dial-peer voice 14 pots
    destination-pattern 3333...
    port 0/0/1:23
    forward-digits all
!
dial-peer voice 15 pots
    destination-pattern 4444....
    port 0/1/0:15
    forward-digits all
!
!
!
!
gatekeeper
    shutdown
!
!
telephony-service
    no auto-reg-ephone
    max-ephones 2
    max-dn 10 no-reg both
    ip source-address 10.1.200.229 port 2000
    system message CUCME
    load 7965 SCCP45.9-2-1S
    time-zone 44
    date-format yy-mm-dd
    max-conferences 8 gain -6
    transfer-system full-consult
    secondary-dialtone 9
    create cnf-files version-stamp 7960 Jul 13 2015 12:39:52
!
!
ephone-dn 1
    number 911 secondary 999 no-reg both
    label 911+999
!
!
ephone-dn 2
    number 4082222020 secondary 2222020 no-reg both
!
!
ephone-dn 3
    number 8593333030 secondary 3333030 no-reg both
!
!

```

```

ephone-dn 4
  number 8144444040 secondary 44444040 no-reg both
!
!
ephone 1
  mac-address EC44.761E.B52C
  type 7965
  button 1:1 2:2 3:3 4:4
!
!
!
!
line con 0
  exec-timeout 0 0
  logging synchronous
  login local
line aux 0
line 2
  no activation-character
  no exec
  transport preferred none
  transport input all
  transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
  stopbits 1
line vty 0 4
  access-class 23 in
  privilege level 15
  login local
  transport input telnet ssh
line vty 5 15
  access-class 23 in
  privilege level 15
  login local
  transport input telnet ssh
!
scheduler allocate 20000 1000
ntp server 64.113.32.5
!
end

```

Hardware Inventory

SW1

```

SW1#show inv
NAME: "1", DESCR: "WS-C3750-24P"
PID: WS-C3750-24PS-S , VID: V08 , SN: FDO1408Z0RM

```

HQ

```

HQ#show inv
NAME: "CISCO2911/K9", DESCR: "CISCO2911/K9 chassis, Hw Serial#: FTX1726AL7J,
Hw Revision: 1.0"

```

PID: CISCO2911/K9 , VID: V07 , SN: FTX1726AL7J

NAME: "VWIC2-1MFT-T1/E1 - 1-Port RJ-48 Multiflex Trunk - T1/E1 on Slot 0 SubSlot 0", DESCR: "VWIC2-1MFT-T1/E1 - 1-Port RJ-48 Multiflex Trunk - T1/E1"
 PID: VWIC2-1MFT-T1/E1 , VID: V01 , SN: FOC14283BL9

NAME: "WAN Interface Card - HWIC Serial 1T on Slot 0 SubSlot 1", DESCR: "WAN Interface Card - HWIC Serial 1T"
 PID: HWIC-1T , VID: V05 , SN: FOC18018N2D

NAME: "PVDM3 DSP DIMM with 128 Channels on Slot 0 SubSlot 4", DESCR: "PVDM3 DSP DIMM with 128 Channels"
 PID: PVDM3-128 , VID: V01 , SN: FOC1633524N

NAME: "PVDM3 DSP DIMM with 64 Channels on Slot 0 SubSlot 5", DESCR: "PVDM3 DSP DIMM with 64 Channels"
 PID: PVDM3-64 , VID: V01 , SN: FOC17405EWP

NAME: "C2911 AC Power Supply", DESCR: "C2911 AC Power Supply"
 PID: PWR-2911-AC , VID: V05 , SN: DCA1708R0L1

BR1

BR1#**show inv**

NAME: "CISCO2911/K9", DESCR: "CISCO2911/K9 chassis, Hw Serial#: FTX1726AL6K, Hw Revision: 1.0"
 PID: CISCO2911/K9 , VID: V07 , SN: FTX1726AL6K

NAME: "VWIC2-2MFT-T1/E1 - 2-Port RJ-48 Multiflex Trunk - T1/E1 on Slot 0 SubSlot 0", DESCR: "VWIC2-2MFT-T1/E1 - 2-Port RJ-48 Multiflex Trunk - T1/E1"
 PID: VWIC2-2MFT-T1/E1 , VID: V01 , SN: FOC11160M6C

NAME: "WAN Interface Card - HWIC Serial 1T on Slot 0 SubSlot 1", DESCR: "WAN Interface Card - HWIC Serial 1T"
 PID: HWIC-1T , VID: V02 , SN: FOC14092KZN

NAME: "4 Port FE Switch on Slot 0 SubSlot 2", DESCR: "4 Port FE Switch"
 PID: HWIC-4ESW , VID: VN/A, SN: FOC10150TYW

NAME: "PVDM3 DSP DIMM with 16 Channels on Slot 0 SubSlot 4", DESCR: "PVDM3 DSP DIMM with 16 Channels"
 PID: PVDM3-16 , VID: V01 , SN: FOC17235E3M

NAME: "PVDM3 DSP DIMM with 16 Channels on Slot 0 SubSlot 5", DESCR: "PVDM3 DSP DIMM with 16 Channels"
 PID: PVDM3-16 , VID: V01 , SN: FOC17244F0V

NAME: "C2911 AC Power Supply", DESCR: "C2911 AC Power Supply"
 PID: PWR-2911-AC , VID: V05 , SN: DCA1708R0N6

BR2

BR2#**show inv**

NAME: "CISCO2921/K9 chassis", DESCR: "CISCO2921/K9 chassis"
 PID: CISCO2921/K9 , VID: V05 , SN: FTX1540AHDV

NAME: "VWIC2-2MFT-T1/E1 - 2-Port RJ-48 Multiflex Trunk - T1/E1 on Slot 0 SubSlot 0", DESCR: "VWIC2-2MFT-T1/E1 - 2-Port RJ-48 Multiflex Trunk - T1/E1"
PID: VWIC2-2MFT-T1/E1 , VID: V01 , SN: FOC11164MVN

NAME: "WAN Interface Card - HWIC Serial 1T on Slot 0 SubSlot 1", DESCR: "WAN Interface Card - HWIC Serial 1T"
PID: HWIC-1T , VID: V04 , SN: FOC15170V4S

NAME: "4 Port FE Switch on Slot 0 SubSlot 2", DESCR: "4 Port FE Switch"
PID: HWIC-4ESW , VID: VN/A, SN: FOC10073BT7

NAME: "PVDM3 DSP DIMM with 32 Channels on Slot 0 SubSlot 4", DESCR: "PVDM3 DSP DIMM with 32 Channels"
PID: PVDM3-32 , VID: V01 , SN: FOC15381A1Z

NAME: "Services Module with Services Ready Engine on Slot 1", DESCR: "Services Module with Services Ready Engine"
PID: SM-SRE-700-K9 , VID: V03 , SN: FOC15334LCJ

NAME: "C2921/C2951 AC Power Supply", DESCR: "C2921/C2951 AC Power Supply"
PID: PWR-2921-51-AC , VID: V02 , SN: DCA1534K0FU

PSTN

PSTN#show inv

NAME: "CISCO2911/K9", DESCR: "CISCO2911/K9 chassis, Hw Serial#: FTX1726AKML, Hw Revision: 1.0"
PID: CISCO2911/K9 , VID: V07 , SN: FTX1726AKML

NAME: "VWIC2-2MFT-T1/E1 - 2-Port RJ-48 Multiflex Trunk - T1/E1 on Slot 0 SubSlot 0", DESCR: "VWIC2-2MFT-T1/E1 - 2-Port RJ-48 Multiflex Trunk - T1/E1"
PID: VWIC2-2MFT-T1/E1 , VID: V01 , SN: FOC11160M5K

NAME: "VWIC2-1MFT-T1/E1 - 1-Port RJ-48 Multiflex Trunk - T1/E1 on Slot 0 SubSlot 1", DESCR: "VWIC2-1MFT-T1/E1 - 1-Port RJ-48 Multiflex Trunk - T1/E1"
PID: VWIC2-1MFT-T1/E1 , VID: V01 , SN: FOC110238J8

NAME: "High Speed Wan Interface Card with 4 serial ports(HWIC-4T) on Slot 0 SubSlot 2", DESCR: "High Speed Wan Interface Card with 4 serial ports(HWIC-4T)"
PID: HWIC-4T , VID: V03 , SN: FOC18114QHR

NAME: "PVDM3 DSP DIMM with 16 Channels on Slot 0 SubSlot 4", DESCR: "PVDM3 DSP DIMM with 16 Channels"
PID: PVDM3-16 , VID: V01 , SN: FOC1723400H

NAME: "C2911 AC Power Supply", DESCR: "C2911 AC Power Supply"
PID: PWR-2911-AC , VID: V05 , SN: DCA1708R021