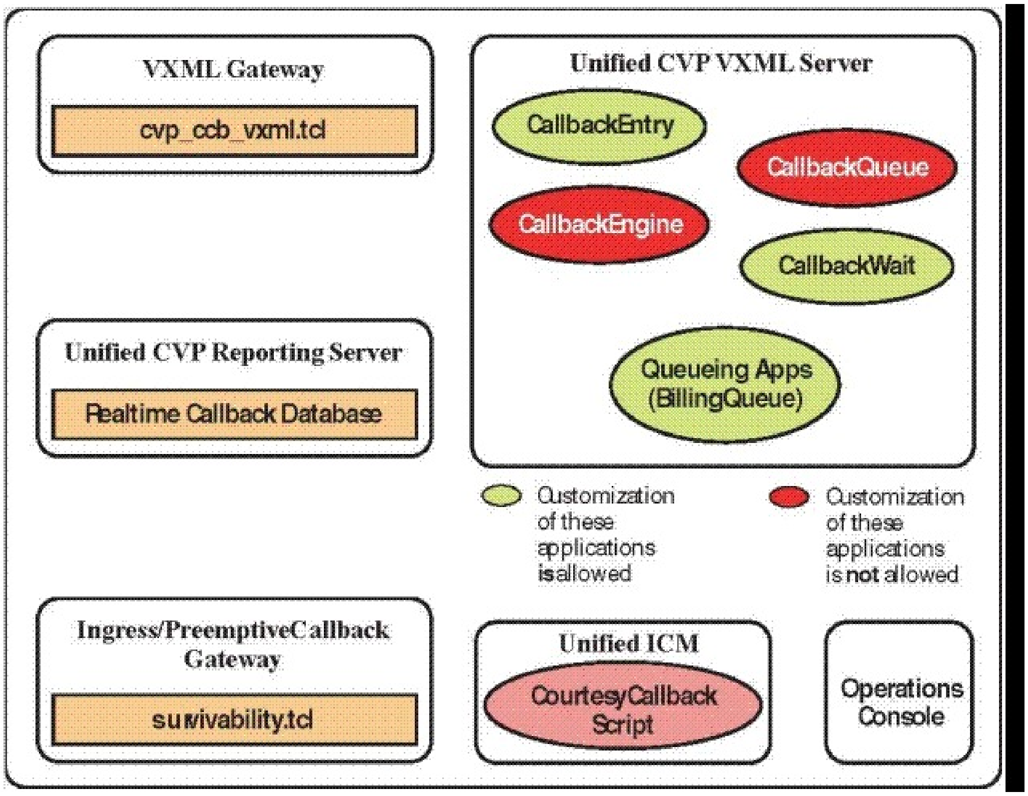
Configure Core Component Integrated Options

Configure Courtesy Callback

The following diagram shows the components that you must configure for Courtesy Callback.

Figure 1: Courtesy Callback components



Complete the following procedures for Courtesy Callback configurations:

- Configure Gateway, on page 2
- Configure Unified CVP, on page 5
- Configure Unified CCE, on page 9

Configure Gateway

Configure the VXML Gateway for Courtesy Callback

Complete the following procedure to configure the VXML gateway for Courtesy Callback:

Procedure

Step 1 Copy **cvp_ccb_vxml.tcl** from the CVP Operations Console to the flash memory of the gateway, as follows:

a) Select **Bulk Administration** > **File Transfer** > **Scripts and Media**.
b) In Device Association, select **Gateway** for Device Type.
c) Select the required gateway from the Available list.
d) Click the right arrow icon to move the available gateway to the Selected list.
e) From the default gateway files, highlight **cvp_ccb_vxml.tcl**.

f) Click **Transfer**.

Step 2 Log on to VXML gateway.

Step 3 Add the cvp_cc service to the configuration **service cvp_cc flash:cvp_ccb_vxml.tcl**.

This service does not require any parameters.

Step 4 Enter the following command to load the application:

call application voice load cvp_cc

Step 5 On the VoIP dial-peer that defines the VRU from Unified CCE, verify that the codec can be used for recording.

Example:

The following example verifies that g711ulaw can be used for recording in Courtesy Callback:

```
dial-peer voice 123 voip
 service bootstrap
 incoming called-number 123T
 dtmf-relay rte-nte
 h245-signal
 h245-alphanumeric
 codec g711ulaw
 no vad!
```

Step 6 Configure the following to ensure that SIP is setup to forward SIP INFO messaging:

```
voice service voip
 signaling forward unconditional
```

Step 7 To play the beep to prompt the caller to record their name in the BillingQueue example script add the following text to the configuration:

vxml version 2.0

Note Whenever you enable vxml version 2.0 on the gateway, vxml audioerror is **off** by default. When an audio file cannot be played, error.badfetch will **not** generate an audio error event.

To generate an error in the gateway, enable vxmlaudioerror.

Example:

The following example uses config terminal mode to add both commands:

```
config t
vxml version 2.0
vxml audioerror
exit
```

Configure the Ingress Gateway for Courtesy Callback

Complete the following procedure to configure the ingress gateway for courtesy callback:

Procedure

Step 1 Copy `surviability.tcl` from the Operations Console to the flash memory of the gateway, as follows:

a) Select **Bulk Administration** > **File Transfer** > **Scripts and Media**.

b) In Device Association, select **Gateway** for Device Type.
c) Select the required gateway from the Available list.
d) Click the right arrow icon to move the available gateway to the Selected list.
e) From the default gateway files, highlight **survivability.tcl**.
f) Click **Transfer**.

Step 2 Log onto the ingress gateway.

Step 3 Add the following to the survivability service:

```
param ccb id:<host name or ip of this gateway>;loc:<location name>;trunks:<number of callback
trunks>
```

- **id -** A unique identifier for this gateway and is logged to the database to show which gateway processed the original callback request.
- **loc -** An arbitrary location name specifying the location of this gateway.
- **Trunks -** The number of DS0's reserved for callbacks on this gateway. Limit the number of T1/E1 trunks to enable the system to limit the resources allowed for callbacks.

Example:

The following example shows a basic configuration:

```
service cvp-survivability flash:survivability.tcl
param ccb id:10.86.132.177;loc:doclab;trunks:1!
```

Step 4 Create the incoming POTS dial peer, or verify that the survivability service is being used on your incoming POTS dial peer.

Example:

For example,

```
dial-peer voice 978555 pots
service cvp-survivability
incoming called-number 9785551234
direct-inward-dial!
```

Step 5 Create outgoing POTS dial peers for the callbacks. These are the dial peers that place the actual call back out to the PSTN.

Example:

For example,

```
dial-peer voice 978555 pots
destination-pattern 978555....
no digit-strip port 0/0/1:23!
```

Step 6 Use the following configuration to ensure that SIP is set up to forward SIP INFO messaging:

voice service voip signaling forward unconditional

Configure CUBE-E for Courtesy Callback

> **Note** If you are using CUBE-E then you need sip profile configuration and apply it on outgoing dial-peer through cvp. See the below the example:

A "sip-profile¨ configuration is needed on ISR CUBE E for the courtesy callback feature. To configure the "sip-profile", the following must be added

```
voice class sip-profiles 103

request INVITE sip-header Call-Info add "X-Cisco-CCBProbe: <ccb param>"
```

where "<ccb param>" is the "ccb" parameter defined in the survivability service. Add this "sip-profile" to the outgoing dial-peer to the CVP.

The following is a configuration example

```
voice class sip-profiles 103

request INVITE sip-header Call-Info add "X-Cisco-CCBProbe: id:10.10.10.180;sydlab;trunks:4"

dial-peer voice 5001 voip

description Comprehensive outbound route to CVP

destination-pattern 5001

session protocol sipv2

session target ipv4:10.10.10.10

dtmf-relay rtp-nte

voice-class sip profiles 103

codec g711ulaw

no vad
```

In the above example, `10.10.10.180` is the CUBE IP and `10.10.10.10` is the CVP Call Server IP.

> **Note** If CUBE E is used for Courtesy Call Back then under voice service voip class in CUBE E must have media flow-through for Courtesy Call Back to work.

Configure Unified CVP

Configure the Reporting Server for Courtesy Callback

A reporting server is required for the Courtesy Callback feature. Complete the following procedure to configure a reporting server for Courtesy Callback:

Before you begin

Install and configure the Reporting Server.

Procedure

Step 1 In the Operations Console, select **System > Courtesy Callback**.

The *Courtesy Callback Configuration* page displays.

Step 2 Choose the **General** tab.

Step 3 Click the **Unified CVP Reporting Server** drop-down, and select the Reporting Server to use for storing Courtesy Callback data.

Step 4 If required, select **Enable secure communication with the Courtesy Callback database**.

Step 5 Configure allowed and disabled dialed numbers.

These are the numbers that the system should and should not call when it is making a Courtesy Callback to a caller.

Note Initially, there are no allowed dialed numbers for the Courtesy Callback feature. Allow Unmatched Dialed Numbers is de-selected and, the Allowed Dialed Numbers window is empty.

Step 6 Adjust the Maximum Number of Calls per Calling Number to the desired number.

By default, this is set to 0 and no limit is imposed. This setting allows you to limit the number of calls that are eligible to receive a callback from the same calling number.

If this field is set to a positive number (X), then the Courtesy Callback Validate element only allows X callbacks per calling number to go through the preemptive exit state at any time.

If there are already X callbacks offered for a calling number, new calls go through the none exit state of the Validate element.

In addition, if no calling number is available for a call, the call always goes through the none exit state of the Validate element.

Step 7 Choose the **Call Server Deployment** tab and move the Call Server you want to use for Courtesy Callbacks from the Available box to the Selected box.

Step 8 Click **Save**.

The configuration becomes active (is deployed) the next time the Reporting Server is restarted.

Step 9 You can also deploy the new Reporting Server configuration immediately by clicking **Save & Deploy**.

Note After all the updates are configured, restart the Reporting Server to update the configuration.

Configure the Call Studio Scripts for Courtesy Callback

The Courtesy Callback feature is controlled by a combination of Call Studio scripts and ICM scripts. Complete the following procedure to configure the Call Studio scripts:

Procedure

Step 1 Access the .zip file from the CVP OAMP machine from the location `C:\Cisco\CVP\OPSConsoleServer\StudioDownloads\CourtesyCallbackStudioScripts.zip`.

Step 2 Extract the example Call Studio Courtesy Callback scripts contained in CourtesyCallbackStudioScripts.zip to a folder of your choice on the computer running CallStudio.

Each folder contains a Call Studio project having the same name as the folder. The five individual project comprise the Courtesy Callback feature.

Note Do not modify the scripts CallbackEngine and CallbackQueue.

Step 3 Modify the scripts **BillingQueue, CallbackEntry,** and **CallbackWait** to suit your business needs.

Step 4 Start Call Studio by selecting **Start > All Programs > Cisco > Cisco Unified Call Studio**.

Step 5 Select **File > Import**.

The Import dialog box displays.

Step 6 Expand the **Call Studio** folder and select **Existing Call Studio Project Into Workspace**.

Step 7 Click **Next** .

The Import Call Studio Project From File System displays.

Step 8 Browse to the location where you extracted the call studio projects. For each of the folders that were unzipped, select the folder (for example BillingQueue) and select **Finish**.

The project is imported into Call Studio.

Step 9 Repeat the action in previous step for each of the five folders.

The five projects display in the upper-left of the Navigator window.

Step 10 Update the Default Audio Path URI field in Call Studio to contain the IP address and port value for your media server.

Step 11 For each of the Call Studio projects previously unzipped, complete the following steps:

a) Select the project in the Navigator window of Call Studio.
b) Choose **Project > Properties > Call Studio > Audio Settings**.
c) On the Audio Settings window, modify the Default Audio Path URI field to http://<media-server >/en-us/VL/.
d) Click **Apply** then click **OK**.

Step 12 Under **BillingQueue Project**, if required, change the music played to the caller while on hold.

a) Expand the tree structure of the project and click **app.callflow**.
b) Click the node **Audio_01**.
c) Navigate to **Element Configuration** > **Audio** > **Audio Groups** expand the tree structure and click **audio item 1**, Use **Default Audio Path** to change the .wav file to be played.

Step 13 Under CallbackEntry Project, if required, modify the caller interaction settings in the **SetQueueDefault_01** node.

a) In the Call Studio Navigator panel, open the **CallBackEntry** project and double-click **app.callflow** to display the application elements in the script window.
b) Open the Start of Call page of the script using the tab at the bottom of the script display window.
c) Select the **SetQueueDefault_01** node.
d) In the Element Configuration panel, choose the **Setting** tab and modify the default settings as required.

Step 14 In the CallbackEntry project, on the Wants Callback page, configure the following:

a) Highlight the Record Name node and choose the **Settings** tab.

b) In the Path setting, change the path to the location where you want to store the recorded names of the callers.
c) Highlight the **Add Callback to DB 1**node.
d) Change the Recorded name file setting to match the location of the recording folder that you created in the previous step.
e) Ensure the **keepalive Interval**(in seconds) is greater than the length of the queue music being played. In the **Start of Call** page.

   The default is 120 seconds for the **SetQueueDefaults_01 node**.

f) Save the CallbackEntry project.
g) In the CallbackWait Project, modify values in the CallbackWait application.

   In this application, you can change the IVR interaction that the caller receives at the time of the actual callback. The caller interaction elements in CallbackWait > AskIfCallerReady page may be modified. Save the project after you modify it.

h) Validate each of the five projects associated with the Courtesy Callback feature and deploy them to your VXML Server.

Step 15 Right-click each Courtesy Callback project in the **Navigator** window and select **Validate**.

Step 16 Right-click on one of the project and click **Deploy**.

Step 17 Check the check box against each project to select the required projects.

Step 18 In the Deploy Destination area, select **Archive File** and click **Browse**.

Step 19 Navigate to the archive folder that you have set up.

Example:

C:\Users\Administrator\Desktop\Sample.

Step 20 Enter the name of the file.

Example:

For example Samplefile.zip.

Step 21 Click **Save**.

Step 22 In the Deploy Destination area click **Finish**.

Step 23 Log in to OAMP and choose **Bulk Administration\File Transfer\VXMLApplications**.

Step 24 Select the VXML Server to which you want to deploy the applications.

Step 25 Select the zip file that contains the applications.

Example:

Samplefile.zip.

Step 26 Click **Transfer**.

Step 27 Right-click each of the projects and click **Deploy**, then click **Finish**.

Step 28 Using windows explorer, navigate to `%CVP_HOME%\VXMLServer\applications`.

Step 29 For each of the five Courtesy Callback applications, open the project's admin folder, in%CVP_Home%\VXMLServer\applications, and double-click **deployApp.bat** to deploy the application to the VXML Server.

Step 30 Verify that all the applications are running by going into `%CVP_HOME%\VXMLServer\admin` and double-clicking **status.bat**. All five applications should display under Application Name and with the status Running.

Configure the Media Server for Courtesy Callback

Several Courtesy Callback specific media files are included with the sample scripts for Courtesy Callback. Complete the procedure to configure the Media Server for Courtesy Callback:

Procedure

Step 1 During the Unified CVP installation, the media files are copied as: `%CVP_HOME%\OPSConsoleServer\CCBDownloads\CCBAudioFiles.zip`.

Step 2 Unzip the special audio files and copy to your media server `VXMLServer\Tomcat\webapps\CVP\audio`.

The sample scripts are set up to use the default location "\CVP\audio" for the audio files.

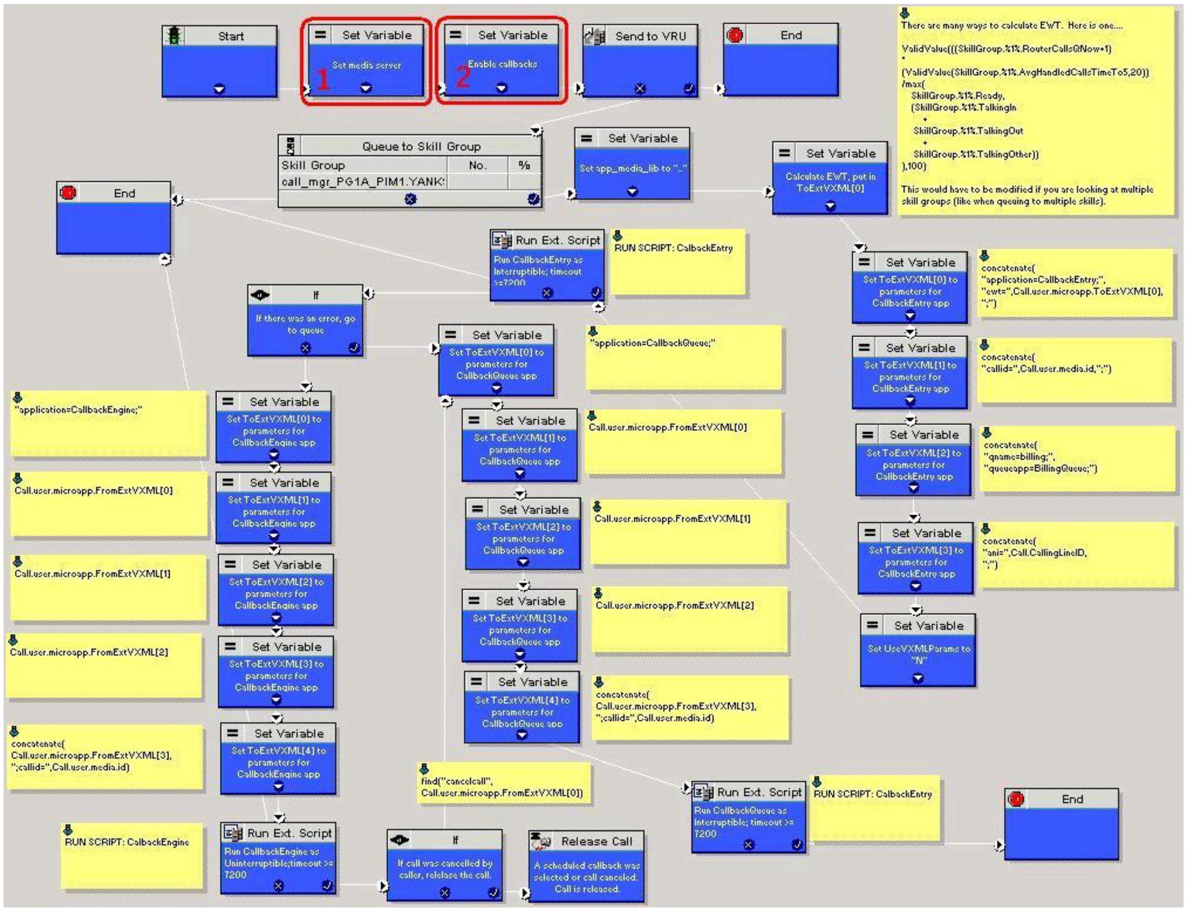
Step 3 Change the default location of the audio files in the sample scripts to be your media server path.

Configure Unified CCE

Configure the ICM Script for Courtesy Callback

Following figure shows the sample Courtesy Callback ICM script.

Figure 2: Sample Courtesy Callback ICM script



Complete the following procedure to configure ICM to use the sample Courtesy Callback ICM script:

Procedure

Step 1 Copy the CCE example script, **CourtesyCallback.ICMS** to the CCE Admin Workstation.
The example CCE script is available in the following locations:

- On the CVP install media in \CVP\Downloads and Samples\.
- From the Operations Console in **%CVP_HOME%\OPSConsoleServer\ICMDownloads**.
- In the Import Script - Manual Object Mapping window, map the route and skill group to the route and skill group available for courtesy callback.

Note For Small Contact Center Deployment Model, copy the CourtesyCallback.ICMS Routing Script on the desktop where Internet Script editor is installed.

Step 2 In Script Editor, select **File > Import Script...**

Note For Small Contact Center Deployment Model follow the below steps.

a. Log In to ISE by sub customer user and Click on File>Import Script.

b. Select the Routing script which is copied in the desktop **CourtesyCallback.ICMS**.

Step 3 In the script location dialog, select the **CourtesyCallback.ICMS** script and click **Open**. You can bypass the set variable "**Set media server**" Highlighted as number 1 node in the Figure 2: Sample Courtesy Callback ICM script, on page 10, as VXML Server, Call Server, and Media Server are collocated.

Step 4 Define a new ECC variable for courtesy callback.

A new ECC variable is used to determine if a caller is in a queue and can be offered a callback.

Step 5 Navigate to **ICM Admin Workstation > ICM Configuration Manager > Expanded Call Variable List tool** to create the ECC Variable **user.CourtesyCallbackEnabled** specific to Courtesy Callback.

Step 6 Set up the following parameters that are passed to CallbackEntry (VXML application):

Example:

- ToExtVXML[0] =concatenate("application=CallbackEntry",";ewt=",Call.user.microapp.ToExtVXML[0])
- ToExtVXML[1] = "qname=billing";
- ToExtVXML[2] = "queueapp=BillingQueue;"
- ToExtVXML[3] = concatenate("ani=",Call.CallingLineID,";");

CallbackEntry is the name of the VXML Server application that is run:

ewt is calculated in **Block #2**.

qname is the name of the VXML Server queue into which the call will be placed. There must be a unique qname for each unique resource pool queue.

queueapp is the name of the VXML Server queuing application that is run for this queue.

ani is the caller's calling Line Identifier.

Step 7 Create Network VRU Scripts.

Step 8 Navigate to **ICM Configuration Manager > Network VRU Script List tool**, create the following Interruptible Script Network VRU Scripts.

Name: **VXML_Server_Interruptible**

Network VRU: Select your Type 10 CVP VRU

VRU Script Name: **GS,Server,V,interrupt**

Timeout: **9000 seconds**

Interruptible: **Checked**

Step 9 Choose **ICM Configuration Manager > Network VRU Script List tool** to create the following Non-Interruptible Script Network VRU Scripts.

Name - **VXML_Server_NonInterruptible**

Network VRU - Select your Type 10 CVP VRU

VRU Script Name - **GS,Server,V, nointerrupt**

Timeout - **9000 seconds (must be greater than the maximum possible call life in Unified CVP)**

Interruptible: **Not Checked**

Step 10 Verify that the user.microapp.ToExtVXMLECC variable is set up for an array of five items with a minimum size of 60 characters and the user.microapp.FromExtVXML variable is set up for an array of four with a minimum size of 60 characters.

Note

Verify that you have at least one available route and skill group to map to the route and skillgroup in the example script.

Step 11 Save the script, then associate the call type and schedule the script.

Note For Small Contact Center Deployment Model ensure the resources used in this Routing Script, like Network VRU Scripts , ECC variables etc are specific to the sub customer.

Configure Agent Greeting

To use Agent Greeting, your phone must meet the following requirements:

- The phones must have the BiB feature.
- The phones must use the firmware version delivered with Unified CM 8.5(1) or greater.

  (In most cases phone firmware is upgraded automatically when you upgrade Unified CM installation.)

Complete the following procedures for Agent Greeting configuration:

- Configure Gateway, on page 12
- Configure Unified CVP, on page 13
- Configure Unified CCE, on page 17
- Configure Unified Communications Manager, on page 22

Configure Gateway

Republish the tcl scripts to VXML Gateway

The .tcl script files that ship with Unified CVP include updates to support Agent Greeting. You must republish these updated files to your VXML Gateway.

Republishing scripts to the VXML Gateways is a standard task in CVP upgrades. You must republish the scripts before you can use Agent Greeting.

Procedure

Step 1 In the Unified CVP Operation Console, select **Bulk Administration** > **File Transfer** > **Scripts and Media**.

Step 2 Set Device to Gateway.

Step 3 Select the gateways you want to update. Typically you would select all of them unless you have a specific reason not to.

Step 4 Select **Default Gateway Files**.

Step 5 Click **Transfer**.

Set Cache Size on VXML Gateway

To ensure adequate performance, set the size of the cache on the VXML Gateway to the maximum allowed. The maximum size is 100 megabytes; the default is 15 kilobytes. Failure to set the VXML Gateway cache to its maximum can result in slowed performance to increased traffic to the media server.

Use the following Cisco IOS commands on the VXML Gateway to reset the cache size:

```
conf t
http client cache memory pool 100000
exit
wr
```

For more information about configuring the cache size, see the *Configuration Guide for Cisco Unified Customer Voice Portal* at https://www.cisco.com/c/en/us/support/customer-collaboration/unified-customer-voice-portal/tsd-products-support-series-home.html.

Configure Unified CVP

Complete the following procedures for Unified CVP configuration:

- Configure FTP Enabled in Server Manager, on page 13
- Configure Unified CVP Media Server
- Configure the Call Studio Scripts for Record Agent Greeting, on page 15

Configure FTP Enabled in Server Manager

Complete the following procedure to configure the FTP enabled in server manager.

Procedure

Step 1 Right- Click **Roles** in the left navigation page of server manager.

Step 2 Select **Add Roles**.

Step 3 Click **Next**.

Step 4 Check the checkbox **Web Server (IIS)** and click **Next**.

Step 5 Check the checkbox **FTP Server** and click **Next**.

Step 6 After the successful installation, click **Close**.

Step 7 Make sure that the FTP and the IIS share the same root directory, because the recording application writes the file to the media server directory structure, and the greeting playback call uses IIS to fetch the file. The en-us/app directory should be under the same root directory for FTP and IIS.

Step 8 Create a dedicated directory on the server to store your greeting files.

This lets you specify a lower cache timeout of 5 minutes for your agent greeting files that does not affect other more static files you may be serving from other directories. By default, the Record Greeting application posts the .wav file to the en-us/app directory under your web/ftp root directory. You may create a dedicated directory such as ag_gr under the en-us/app directory, and then indicate this in the Unified CCE script that invokes the recording application. Use the array for the ECC variable **call.user.microapp.ToExtVXML** to send the ftpPath parameter to the recording application. Make sure the ECC variable length is long enough, or it may get truncated and fail.

Step 9 In IIS Manager, set the cache expiration for the dedicated directory to a value that allows re-recorded greetings to replace their predecessor in a reasonable amount of time, while minimizing requests for data to the media server from the VXML Gateway.

The ideal value varies depending on the number of agents you support and how often they re-record their greetings. Two minutes may be a reasonable starting point.

Step 10 Find the site you are using, go to the agent greeting folder you created (ag_gr), and then select **HTTP Response Headers** .

Step 11 Select **Add**, then **Set Common Headers** .

Create Voice Prompts for Recording Greetings

You must create audio files for each of the voice prompts that agents hear as they record a greeting. The number of prompts you require can vary, but a typical set can consist of:

- A welcome followed by a prompt to select which greeting to work with (this assumes you support multiple greetings per agent)
- A prompt to select whether they want to hear the current version, record a new one, or return to the main menu
- A prompt to play if a current greeting is not found.

To create voice prompts for recording greetings:

Procedure

Step 1 Create the files using the recording tool of your choice. When you record your files:

- The media files must be in `.wav` format. Your `.wav` files must match Unified CVP encoding and format requirements (G.711, CCITT A-Law 8 kHz, 8 bit, mono).
- Test your audio files. Ensure that they are not clipped and that they are consistent in volume and tone.

Step 2 After recording, deploy the files to your Unified CVP media server. The default deployment location is to the *`<web_server_root>`*`\en-us\app` directory.

Step 3 Note the names of the files and the location where you deployed them on the media server. Your script authors need this information for the Agent Greeting scripts.

Built-In Recording Prompts

The Unified CVP Get Speech micro-application used to record Agent Greetings includes the following built-in prompts:

- A prompt that agents can use to play back what they recorded

- A prompt to save the greeting, record it again, or return to the main menu
- A prompt that confirms the save, with an option to end the call or return to the main menu

You can replace these `.wav` files with files of your own. For more information, see the Unified Customer Voice Portal Call Studio documentation at https://www.cisco.com/c/en/us/support/unified-communications/unified-call-studio/tsd-products-support-series-home.html.

Configure Unified CVP Media Server

Procedure

Step 1 In the CVP Operations Console, navigate to **Device Management** > **Media Server**.

Step 2 Click **Add New**.

Step 3 On the **General** tab, configure the following.

a) Enter the IP address and the hostname of the Unified CVP server.
b) Check **FTP Enabled**.
c) Either Check **Anonymous Access** or enter the credentials.
d) Click **Test SignIn** to validate the FTP access.

Step 4 Click **Save**.

Step 5 Repeat Step 1 through 4 for all Media Servers.

Step 6 After you configure all Media Servers, click **Deploy**.

Step 7 Click **Deployment Status** to make sure that you applied the configuration.

Step 8 In the CVP Operations Console, navigate to **Device Management** > **Media Server**.

Step 9 Change Default Media Server from **None** to any one of the Unified CVP servers. Then click **Set**.

Step 10 Click **Deploy**.

Configure the Call Studio Scripts for Record Agent Greeting

The Record Agent Greeting is controlled by a combination of Call Studio script and ICM script. Complete the following procedure to configure the Call Studio script:

Procedure

Step 1 Access the .zip file from the CVP OAMP machine from the location `C:\Cisco\CVP\OPSConsoleServer\StudioDownloads\RecordAgentGreeting.zip`.

Step 2 Extract the example Call Studio Record Agent Greeting scripts contained in RecordAgentGreeting.zip to a folder of your choice on the computer running CallStudio. The folder contains a CallStudio project having the same name as the folder.

Step 3 Start Call Studio by selecting **Start** > **Programs** > **Cisco** > **Cisco Unified Call Studio**.

Step 4 Select **File** > **Import**.
The **Import** dialog box displays.

Step 5 Expand the **Call Studio** folder and select **Existing Call Studio** project Into Workspace.

Step 6 Click **Next**.
The Import Call Studio Project From File System displays.

Step 7 Browse to the location where you extracted the call studio projects. Select the folder and select **Finish**.

Example:

RecordAgentGreeting

Step 8 Follow the below steps, to save the file in a defined path:

a) In the **Call Studio Navigator** panel, open the **RecordAgentGreeting** project and double click **app.callflow** to display the application elements in the **script** window.
b) Select the **Record Greeting With Confirm** node.
c) In the **Element Configuration** panel, choose the **Setting** tab and modify the default path settings to `c:\inetpub\wwwroot\en-us\app\ag_gr`. Save the project after you modify it.
d) Validate the project associated with the **Record Agent Greeting** and deploy them to your VXML Server.

Step 9 Right-click on **Record Agent Greeting** project in the **Navigator** window and select **Validate**.

Step 10 Right-click on the **Record Agent Greeting** project and click **Deploy**.

Step 11 In the **Deploy Destination** area, select **Archive File** and click **Browse**.

Step 12 Navigate to the archive folder that you have set up:

Example:

`C:\Users\Administrator\Desktop\Sample`.

Step 13 Enter the name of the file.

Example:

Samplefile.zip

Step 14 Click **Save**.

Step 15 In the **Deploy Destination**area click **Finish**.

Step 16 Log in to **OAMP** and choose **Bulk Administration\File Transfer\VXMLApplications**.

Step 17 Select the **VXML Server** to which you want to deploy the applications.

Step 18 Select the zip file that contains the applications.

Example:

Samplefile.zip

Step 19 Click **Transfer**.

Step 20 Right-click on the project and click **Deploy**, then click **Finish**.

Step 21 Using windows explorer, navigate to `%CVP_HOME%\VXMLServer\applications\RecordAgentGreeting`, open the project's admin folder and double-click deployApp.bat to deploy the application to the VXML Server.

Step 22 Verify that the application is running in the following path `%CVP_HOME%\VXMLServer\applications\RecordAgentGreeting\admin` and double-click **status.bat**. The application should display under Application Name and with the status Running.

Set Content Expiration in IIS (Windows Server) in Media

Complete the following procedure to set content expiration in IIS on a Windows Server:

Procedure

Step 1 Right-click **My Computer** on the desktop and select **Manage**.

Step 2 Select **Server Manager** > **Roles** > **Web Server (IIS)** > **Internet Information Services (IIS) Manager**.

Step 3 Select the default website and navigate to **Features View**.

Step 4 Double-click **HTTP Response Headers**.

Step 5 Under **Actions**, select **Set Common Headers…**.

Step 6 On **Set Common HTTP Response Headers**, select **Enable HTTP keep-alive** and **Expire Web content** and set **After 5** minutes.

Configure Unified CCE

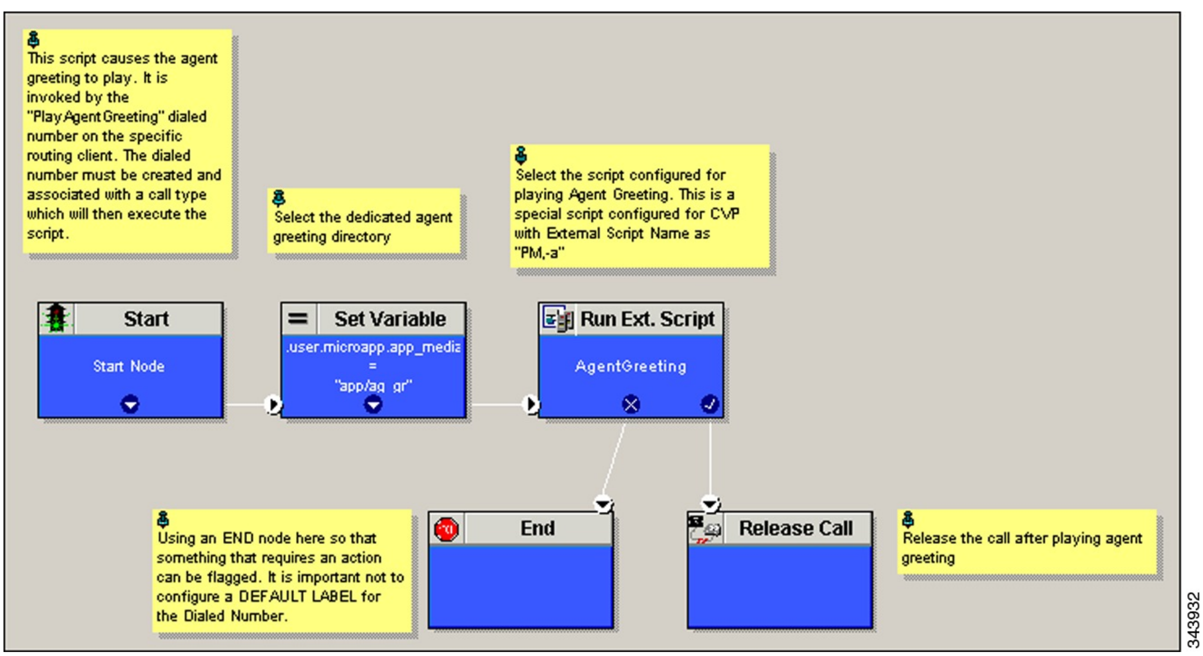
Complete the following procedures forUnified CCE configuration:

- Create Agent Greeting Play Script, on page 17
- Create Agent Greeting Recording Script, on page 18
- Import the Example Agent Greeting Scripts, on page 19

Create Agent Greeting Play Script

A dedicated routing script plays the Agent Greeting. This script is invoked by the PlayAgent Greeting dialed number on the specific routing client. You must create the dialed number and associate it with a call type that runs the script.

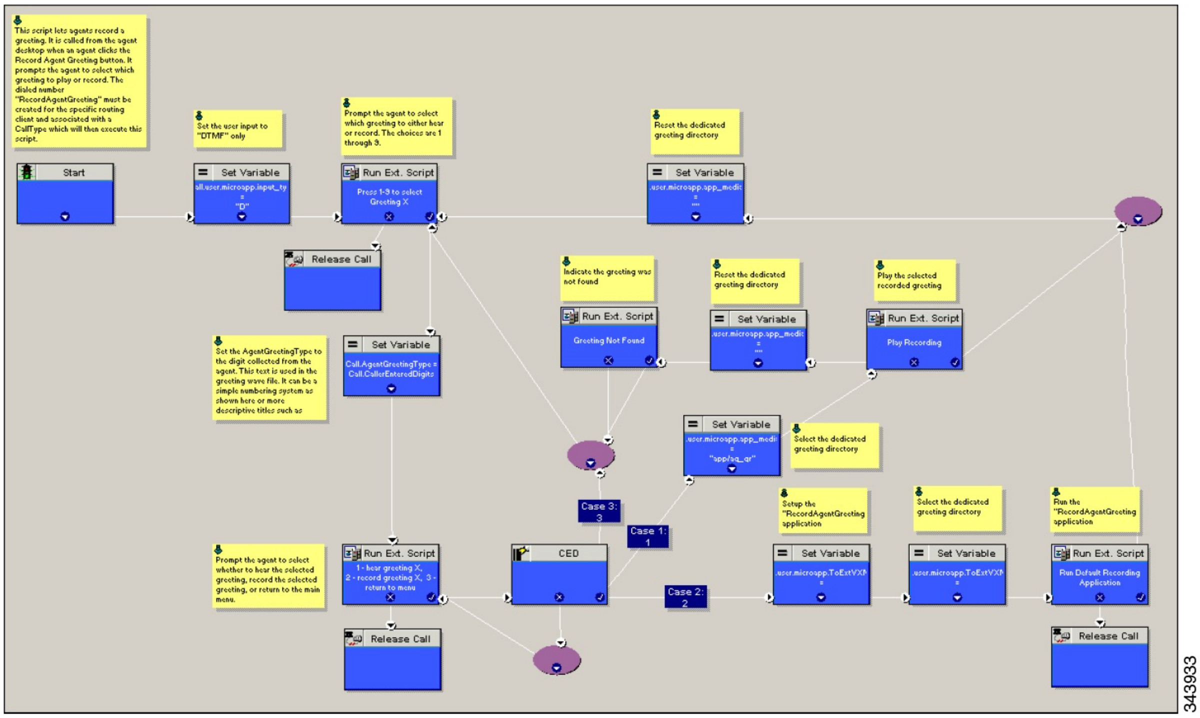
Figure 3: Agent Greeting Play Script



Create Agent Greeting Recording Script

The Agent Greeting Recording script lets agents record a greeting. The agent desktop calls the script when an agent clicks the Record Agent Greeting button, prompting the agent to select which greeting to play or record. Create the dialed number RecordAgentGreeting for the specific routing client and associate it with a call type that then runs this script.

Figure 4: Agent Greeting recording script



Unified CCE Configuration for Record Agent Greeting

- **user.microapp.ToExtVXML** : This is used twice in an Agent Greeting record script, the first time is to queue the Unified CVP Record Agent Greeting application and the second time is to tell the recording application where to save greeting files. Configure it as an array with size 3. Use the Unified CCE Administration tool to ensure this variable includes Maximum Length as 100 and Enabled.
- **user.microapp.app_media_lib** :This is required in Agent Greeting record and play scripts to specify the dedicated directory on the media server where your greeting audio files are stored. Maximum Length is 100 and Enabled.
- **user.microapp.input_type:** This is required in Agent Greeting record scripts to limit the allowable input type to DTMF. Maximum Length is 100 and Enabled.

Note For more information on how to enable the ECC variables, see the *Cisco Hosted Collaboration Solution for Contact Center Configuration Guide* at https://www.cisco.com/c/en/us/support/unified-communications/hosted-collaboration-solution-contact-center/products-installation-and-configuration-guides-list.html.

Import the Example Agent Greeting Scripts

To view or use the example Agent Greeting scripts, you must first import them into theUnified CCE Script Editor. Complete the following procedure to import the example Agent Greeting scripts:

Procedure

Step 1 Launch **Script Editor**.

Step 2 Select **File**>**Import Script** and select the following scripts to import:

a) Agent Greeting Play Script
b) Agent Greeting Recording Script

The scripts will be located in the icm\bin directory on the data server (DS) node.

Step 3 Repeat for the remaining scripts.

Note For Small Contact Center Deployment Model, Default Routing Scripts are available in the partners Community. Download the Routing Scripts to the Desktop where ISE is Installed and Login as the Sub Customer User into the ISE to perform the Step 2 and 3. Download the `Routing Script files for all Deployment models` https://software.cisco.com/download/navigator.html?mdfid=284526699.

Note For Small Contact Center Deployment Model ensure the resources used in this Routing script, like Network VRU Scripts , ECC variables etc are specific to the sub customer.

Configure Call Types

Procedure

Step 1 Sign-in to **Unified CCDM Portal** as Tenant or Sub Customer user.

Step 2 Click the burger icon and select **Provisioning** > **Resource Manager**

Step 3 Select the folder where you want to create the call type.

Step 4 Click **Resource**, then click **Call Types**.

Step 5 Create a call type to record agent greetings and enter **RecordAgentGreeting** as the name.

Step 6 Create a call type to play agent greetings and enter **PlayAgentGreeting** as the name.

Configure Dialed Numbers

Procedure

Step 1 Sign-in to **Unified CCDM Portal** as Tenant or Sub Customer user.

Step 2 Click the burger icon and select **Provisioning** > **Resource Manager**

Step 3 Select the folder where you want to create the dialed number.

Step 4 Click **Resource**, then click **Dialed Number**.

Step 5 Create a dialed number to record agent greetings and enter **RecordAgentGreeting** as the name.

Step 6 Create a dialed number to play agent greetings and enter **PlayAgentGreeting** as the name.

Step 7 Complete the following for each dialed number:

a) Select **Internal Voice** for the Routing type.
b) Retain the default domain value.
c) Select the call type appropriate to the dialed number.

This helps to associate each number to its call type and to a script that runs it.

Schedule the Script

Procedure

Step 1 In the **Script Editor**, select **Script** > **Call Type Manager**.

Step 2 From the Call Type Manager screen, select the **Schedules** tab.

Step 3 From the Call type drop-down list, select the call type to associate with the script; for example, PlayAgentGreeting.

Step 4 Click **Add** and select the script you want from the Scripts box.

Step 5 Click **OK** twice to exit.

Configure Agent Greeting

This section describes how to deploy and configure the Agent Greeting feature.

Agent Greeting Deployment Tasks

Procedure

Step 1 Ensure your system meets the baseline requirements for software, hardware, and configuration described in the System Requirements and Limitations section.

Step 2 Configure IIS and FTP on Media Server.

Step 3 In Unified CVP, add media servers, configure FTP connection information, and deploy the media servers.

Step 4 Configure a Unified CVP media server, if you have not already done so. See Configure Unified CVP Media Server.

Step 5 In Unified CVP Operations Console, republish the VXML Gateway.tcl scripts with updated Agent Greeting support. See Republish the tcl scripts to VXML Gateway, on page 12 for Agent Greeting support.

Step 6 Set the cache size on the VXML Gateway. See Set Cache Size on VXML Gateway, on page 13.

Step 7 Record the voice prompts to play to agents when they record a greeting and to deploy the audio files to your media server, see Create Voice Prompts for Recording Greetings, on page 14.

Step 8 Configure Call Types, on page 19 to record and play agent greetings.

Step 9 Configure Dialed Numbers, on page 19 to record and play agent greetings.

Step 10 Schedule the Script, on page 20

Step 11 In Script Editor:

- To use the installed scripts to record and play agent greetings, see Import the Example Agent Greeting Scripts, on page 19.

Step 12 Modify the Unified CCE call routing scripts to use Play Agent Greeting script, on page 21.

Modify the Unified CCE call routing scripts to use Play Agent Greeting script

For an Agent Greeting play script to run, you must add an AgentGreetingType Set Variable node to your existing Unified CCE call routing scripts: This variable's value is used to select the audio file to play for the greeting. Set the variable before the script node that queues the call to an agent (that is, the Queue [to Skill Group or Precision Queue], Queue Agent, Route Select, or Select node).

Specify AgentGreetingType Call Variable

To include Agent Greeting in a script, insert a Set Variable node that references the AgentGreetingType call variable. The AgentGreetingType variable causes a greeting to play and specifies the audio file it should use. The variable value corresponds to the name of the greeting type for the skill group or Precision Queue. For example, if there is a skill group or Precision Queue for Sales agents and if the greeting type for Sales is '5', then the variable value should be 5.

You can use a single greeting prompt throughout a single call type. As a result, use one AgentGreetingType set node per script. However, as needed, you can set the variable at multiple places in your scripts to allow different greetings to play for different endpoints. For example, if you do skills-based routing, you can specify the variable at each decision point used to select a particular skill group or Precision Queue.

Note Only one greeting can play per call. If a script references and sets the AgentGreetingType variable more than once in any single path through a script, the last value to be set is the one that plays.

Use these settings in the Set Variable node for Agent Greeting:

- Object Type: Call.
- Variable: Must use the AgentGreetingType variable.
- Type: Must use the PersonID_AgentGreetingType type.
- Value: Specify the value that corresponds to the greeting type you want to play. For example: “2” or “French”
    - You must enclose the value in quotes.
    - The value is not case-sensitive.
    - The value cannot include spaces or characters that require URL encoding.

Configure Unified Communications Manager

Built-in-Bridge

Built-in-Bridge (BIB) is not enabled by default for the phones. It is disabled at the system level as it is not used by all the customer by default. It is used only by the customers having Contact Center.

The provider has to perform the following procedures to enable BIB for the customers having contact center.

Note Create a new Field Display Policies at the customer level and add Built-in Bridge to the list.

- Configure the Built-in-Bridge
- Enable or Disable the Built-in-Bridge

Configure the Built-in-Bridge

Procedure

Step 1 Login to **Cisco Unified Communication Domain Manager** as provider.
Step 2 Navigate **Role Management** > **Field Display Policies**.
Step 3 Ensure that hierarchy is set to the appropriate customer.
Step 4 Select the **SubscriberPhoneMenuItemProvider**.
Step 5 In the details page, go to **Action** menu and click **Clone**.
Step 6 Enter **SubscriberPhoneMenuItemProvider** as the name.
Step 7 Select **relation/SubscriberPhone** from the **Target Model Type** drop-down list.
Step 8 Expand **Groups** section and enter **Phone** for Title.
Step 9 Select **builtInBridgeStatus** from the **Available** list and click **Select**.
Step 10 Click **Save**.

Enable or Disable the Built-in-Bridge

Before you begin

Ensure that you configure Built-in-Bridge. See, Configure the Built-in-Bridge.

Procedure

Step 1 Login to **Cisco Unified Communication Domain Manager** as a provider.
Step 2 Ensure that hierarchy is set to the appropriate customer.
Step 3 Navigate **Subscriber Management** > **Phones** and select the appropriate phone.
Step 4 In the **Phone** tab:

- To enable BIB choose **On** from the **Built in Bridge** drop-down list.
- To disable BIB choose **Off** from the **Built in Bridge** drop-down list.

Step 5 Click **Save**.

Configure Whisper Announcement

Complete the following procedures for Whisper Announcement configuration:

- Configure Gateway, on page 23
- Configure Unified CVP, on page 23
- Configure Unified CCE, on page 24

Configure Gateway

Gateway uses two different dialed numbers for Whisper Announcement.

- 91919191 number calls the ring tone that the caller hears while the whisper plays to the agent
- 9191919100 number calls the whisper itself

Configure a dial peer for incoming number 9191919100 and 91919191 as follows:

```
dial-peer voice 919191 voip
description CVP SIP ringtone dial-peer
service ringtone
incoming called-number 9191T
voice-class sip rel1xx disable
dtmf-relay rtp-nte
codec g711ulaw
no vad
```

Configure Unified CVP

Configure the Whisper Announcement Service Dialed Numbers

Unified CVP uses two different dialed numbers for Whisper Announcement:

The first number calls the ring tone service that the caller hears while the whisper plays to the agent. The Unified CVP default for this number is 91919191.

The second number calls the whisper itself. The Unified CVP default for this number is 9191919100.

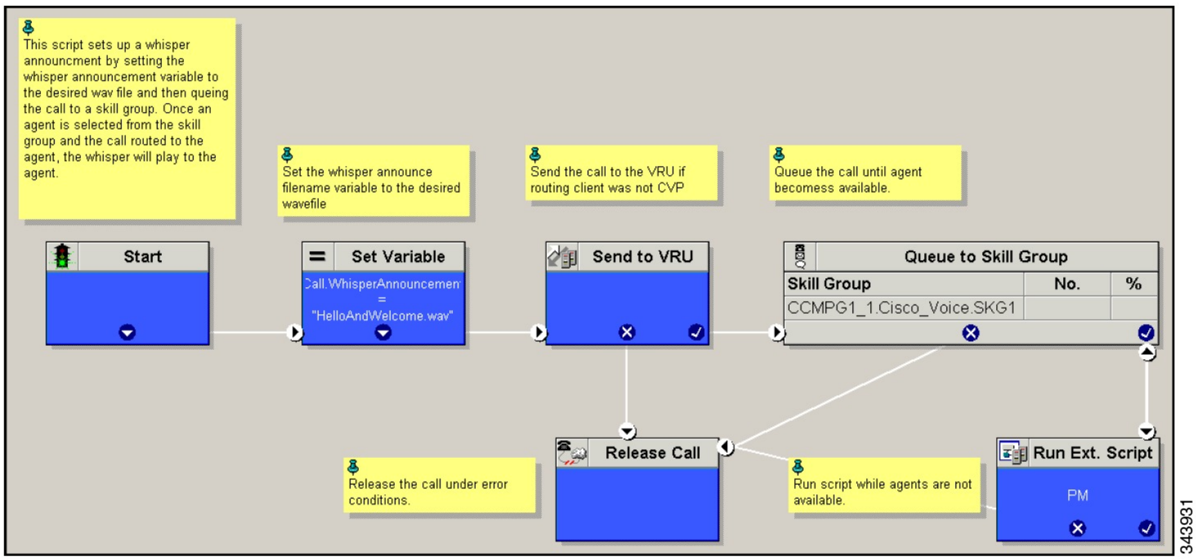
For Whisper Announcement to work, your dial number pattern must cover both of these numbers. The easiest way to ensure coverage is through the use of wild cards such as 9191*. However, if you decide to use an exact dialed number match, then you must specify both 91919191 and 9191919100.

Configure Unified CCE

Create Whisper Announcement Script

It is very important to deploy Whisper Announcement with the Call. Whisper Announcement variable and to set .wav file in your Unified CCE routing scripts.

Figure 5: Whisper Announcement Script



Configure Database Integration

Complete the following procedures for Database Integration configuration:

- Configure Unified CVP, on page 24
- Configure Unified CCE, on page 27

Note Small Contact Center deployment model supports only CVP Database Integration.

Configure Unified CVP

Configure VXML Database Element

You need to configure Java Database Connectivity (JDBC) for VXML Database Element configuration.

Complete the following procedures for JDBC configuration:

- Install JDBC driver, on page 25
- Add JNDI Context, on page 25

• Configure VXML Studio Script, on page 26
• Create ICM Script, on page 26

Install JDBC driver

Complete the following procedure to install the JDBC driver:

Procedure

Step 1 Download the .exe file for Microsoft JDBC Driver for SQL Server

Example:

```
1033\sqljdbc_3.0.1301.101_enu.exe
```

Step 2 Run the executable and install the .exe file in the location `C:\temp\`

Step 3 Copy the file `C:\temp\sqljdbc_3.0\enu\sqljdbc4.jar` to the Unified CVP VXML servers' folder `C:\Cisco\CVP\VXMLServer\Tomcat\common\lib`

Add JNDI Context

Complete the following procedure to add the Java Naming and Directory Interface (JNDI) context configuration:

Procedure

Step 1 Go to the context.xml file located at C:\Cisco\CVP\VXMLServer\Tomcat\conf\context.xml file.

Step 2 Enter the JNDI name, SQL server address, SQL database name, username and password.

The following is an example of the SQL authentication context.xml file:

```
<Context>

<WatchedResource>WEB-INF/web.xml</WatchedResource>

<Manager pathname="" />

<Resource name="jdbc/dblookup"

auth="Container"

type="javax.sql.DataSource"

DriverClassName="com.microsoft.sqlserver.jdbc.SQLServerDriver"

url="jdbc:sqlserver://<dblookupnode_ipaddress>:1433;databaseName=DBLookup;user=sa;password=sa"

>

</Context>
```

Step 3 Perform following steps to restart VXML server services:

a) Goto **Run** window and enter `services.msc` command.
b) Select **Cisco CVP VXML Server** option.
c) Right-click and select **Restart** option.

Note For small contact center agent deployment model , Resource name should be unique for each sub-customers. For example, Sub-cust1 Resource name = "jdbc/dblookup1" and Sub-cust2 Resource name = "jdbc/dblookup2".

Configure VXML Studio Script

Complete the following procedure to configure the VXML studio script:

Procedure

Step 1 Configure the following to create the VXML application with the database element.

a) Select **single** under **Type**.
b) Enter the database lookup name in **JNDI Name**.
c) Query SQL:

```
For example, select AccountNo from AccountInfo where CustomerNo = {CallData.ANI}

Where AccountNo - Value to be retrieved

AccountInfo - Table name

CustomerNo - condition to be queried

Data:

Create a database element with the following values:

Name - AccountNo

Value - {Data.Element.Database_01.AccountNo}
```

Step 2 Deploy the script to the local computer or to the remote computer (VXML call server directly) to create CVP Subdialog return element.

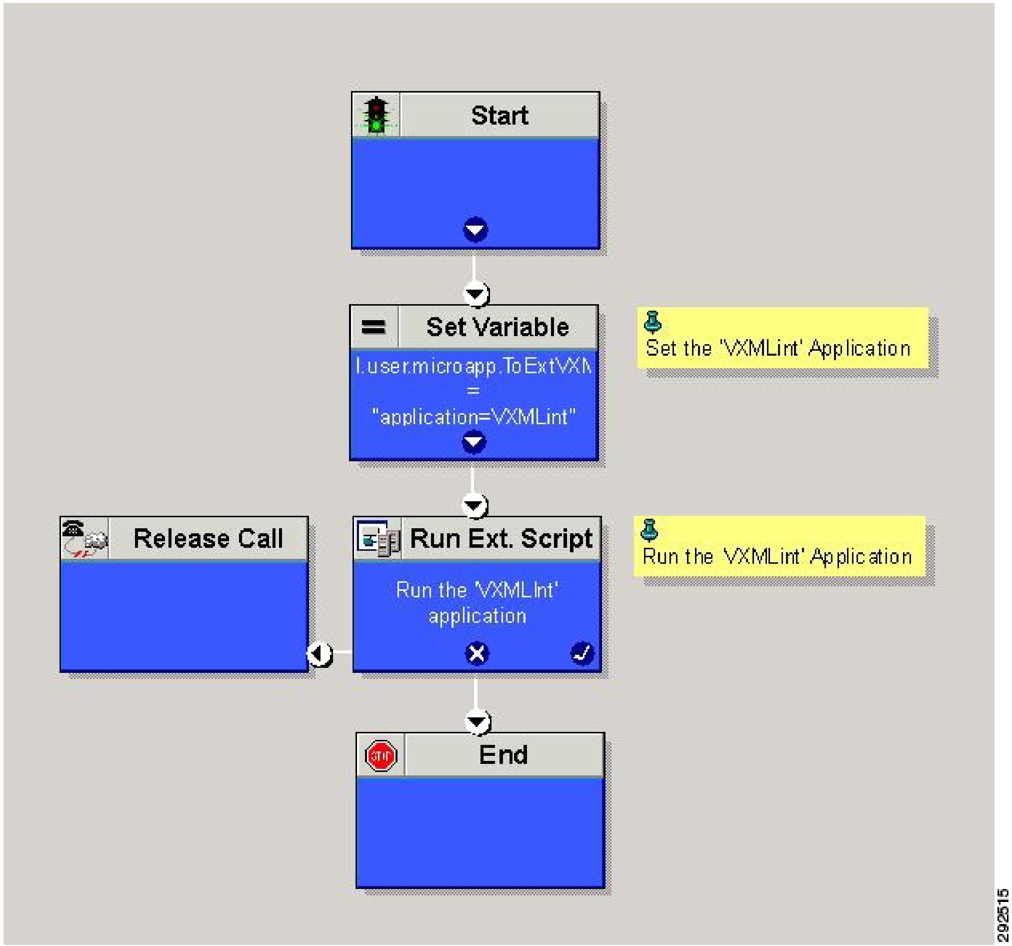
Step 3 If you saved this to the local machine, copy the whole folder to the following location:

<Install dir>:\Cisco\CVP\VXMLServer\applications and deploy it using deployApp windows batch file located inside the admin folder of applications.

Create ICM Script

Create an ICM script similar to the one shown in the following figure:

Figure 6: Sample Script with ICM database Lookup



Configure Unified CCE

Configure ICM Database Lookup

Complete the following procedure to configure ICM Database Lookup.

Procedure

Step 1 Select **Enable Database Routing** in **Router options** to edit Router setup for database lookup changes.

Step 2 Configure Database Lookup explorer:

a) Click **Start** > **All programs** > **Cisco Unified CCE Tools** > **Administration Tools** > **Configuration Manager**.

b) Open **Tools** > **Explorer Tools** > **Database Lookup Explorer.**

c) Configure Script Table and Script Table Column as shown in the following example:

Script Table:

```
Name: AccountInfo

Side A: \\dblookup1\DBLookup.AccountInfo

Side B: <Update Side B of database here>

Description: <Provide description here>
```

dblookup1 is external database server name, DBLookup is external database name, and AccountInfo is the table name.

Script Table Column:

```
Column name: AccountNo

Description: <Provide description here>
```

Step 3 Configure the following to change the registry settings in Unified CCE:

a) Navigate to **HKEY_LOCAL_MACHINE > SOFTWARE > Cisco Systems, Inc. > ICM > <Instance Name> > RouterA > Router > CurrentVersion > Configuration > Database registry**.

   Instance Name is the name of the Instance that is configured.

b) Set the SQLLogin registry key as shown in the following example:

   Example:

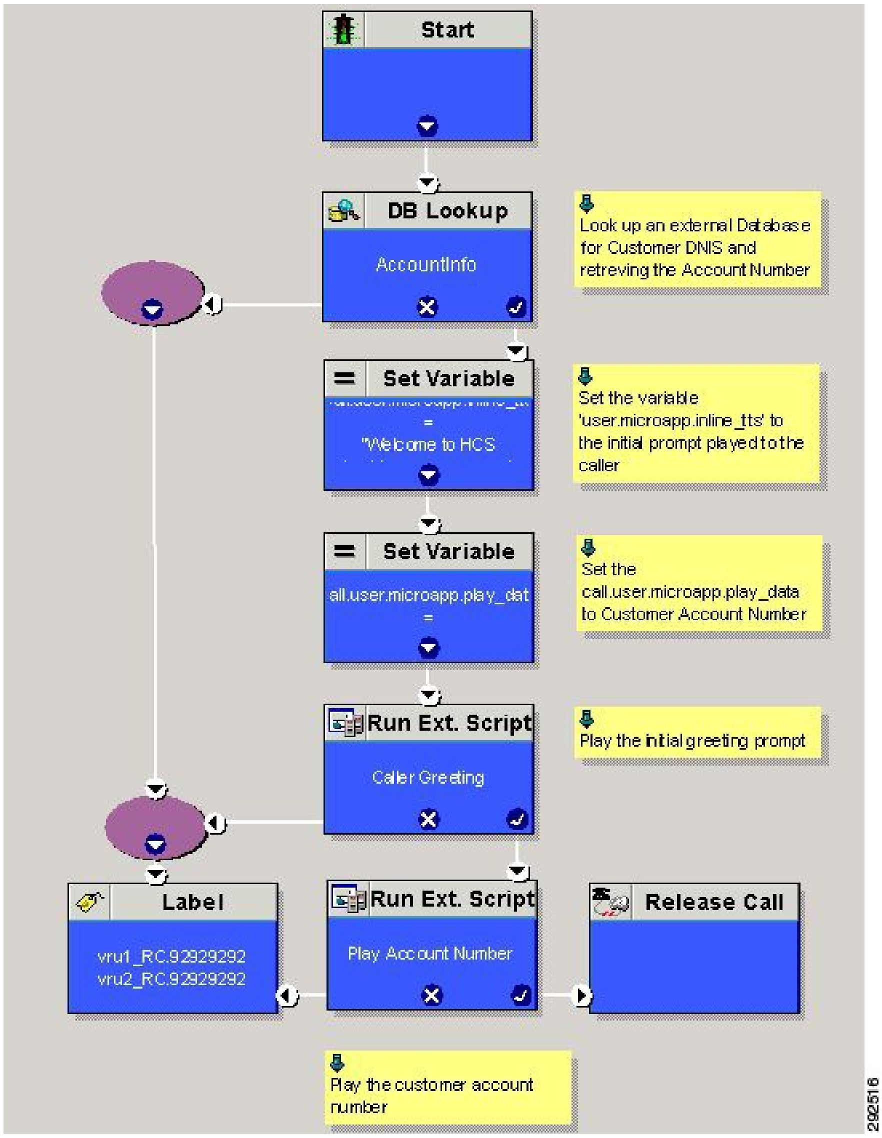
   ```
   \\dblookup1\DBLookup=(sa,sa)
   ```

Where DBLookup is the external database name and (sa,sa) are the SQL server authentication.

Step 4 Create the ICM script with the database lookup node with the respective table and lookup value.

The following figure shows AccountInfo as the table name and Call.CallingLineID as the lookup value.

Figure 7: Example ICM Database Look Up



Configure Unified Mobile Agent

- Configure Gateway for SCC Deployment with VRF, on page 30
- Configure Unified CCE, on page 30
- Configure Unified Communications Manager, on page 31

Configure Gateway for SCC Deployment with VRF

Configure Dial Peer for Sub-Customer1 CUCM

```
dial-peer voice 21011 voip
description from CVP towards VRF1 to Sub-Customer1 for mobileagent
destination-pattern 100.
session protocol sipv2
session target ipv4:20.20.20.31
session transport udp
voice-class codec 1
voice-class sip rel1xx disable
voice-class sip bind control source-interface GigabitEthernet2.100
voice-class sip bind media source-interface GigabitEthernet2.100
dtmf-relay rtp-nte h245-signal h245-alphanumeric
```

Configure Dial Peer for Sub-Customer2 CUCM

```
dial-peer voice 22011 voip
description from CVP towards VRF2 to Sub-Customer2 for mobileagent
destination-pattern 300.
session protocol sipv2
session target ipv4:20.20.20.31
session transport udp
voice-class codec 1
voice-class sip rel1xx disable
voice-class sip bind control source-interface GigabitEthernet2.200
voice-class sip bind media source-interface GigabitEthernet2.200
dtmf-relay rtp-nte h245-signal h245-alphanumeric
```

Configure Unified CCE

Complete the following procedure to configure Mobile Agent in Unified CCE:

Procedure

Step 1 Sign-in to **Unified CCDM Portal** as Tenant or Sub Customer user.

Step 2 Click the burger icon and select **Provisioning** > **Resource Manager**

Step 3 Select the folder where you want to create the agent desktop.

Step 4 Click **Resource**, then click **Agent Desktop**.

Step 5 Enter unique name of up to 32 characters for the record.

This name can use alphanumeric characters, periods, and underscores.

Step 6 Enter the mandatory fields such as **Incoming Work mode**, **Outgoing Work mode**, **Wrap-up time**, and other required fields.

Step 7 Click **Save**.

Enable Mobile Agent Option in CTI OS Server

Complete the following procedure to enable Mobile Agent option in CTI OS server:

Procedure

Step 1 Invoke the CTI OS Server setup.

Step 2 In **Peripheral Identifier** window, check **Enable Mobile Agent** check box, and select **Mobile Agent Mode** from the drop-down list.

Step 3 Repeat the above steps on both sides of CTI OS server.

Configure Unified Communications Manager

Perform the following to configure unified communications manager:

- Configure CTI Port, on page 31
- Tag CTI Ports as Contact Center Agent Lines, on page 33

Configure CTI Port

Ensure that directory numbers are added. See Add Directory Number Inventory.

Unified Mobile Agent needs two configured CTI Port pools on Unified Communications Domain Manager:

- A local CTI port as the agent's virtual extension
- A network CTI port to initiate a call to the Mobile Agent's phone

Note For 12000 agent deployment model, add CTI ports for all three Unified CM clusters.

Complete the following procedure to configure CTI port:

- Configure CTI Port as Provider or Reseller, on page 31
- Configure CTI Port as Customer, on page 32

Configure CTI Port as Provider or Reseller

Procedure

Step 1 Login to Cisco Unified Communication Domain Manager as provider or reseller.

Step 2 Ensure that hierarchy is set to appropriate site

Step 3 Navigate to **Subscriber Management** > **Phones**.

Step 4 Click **Add**.

Step 5 In **Phones** tab:

a) Enter Local CTI Port pool name in **Device Name** field, in *LCPxxxxFyyyy* format.

- LCP - identifies the CTI port as a local device
- xxxx - is the peripheral ID of the Unified Communication Manager PIM
- yyyy - is the local CTI Port

b) Choose **CTI Port** from **Product Type** drop-down list.
c) Choose **Calling Search Space** from the drop-down list.
d) Choose **Device Pool** from the drop-down list.
e) Choose **Location** from the drop-down list.

Step 6 Goto **Lines** tab:

a) Click **Add** icon in **Lines** panel.
b) Choose directory number from **Pattern** drop-down list, in **Drin** Panel.
c) Choose **Route Partition Name** from drop-down list.

Step 7 Click **Save**.

What to do next

Repeat the above steps to create Network CTI port. Enter Network CTI Port pool name in **Device Name** field, in *RCPxxxxFyyyy* format.

- RCP - identifies the CTI port as a network device
- xxxx - is the peripheral ID of the Unified Communication Manager PIM
- yyyy - is the network CTI Port

Note Local CTI port and Network CTI port should be same

Configure CTI Port as Customer

Procedure

Step 1 Login to Cisco Unified Communication Domain Manager as Customer admin.
Step 2 Ensure that hierarchy is set to appropriate site
Step 3 Navigate to **Subscriber Management** > **Phones**.
Step 4 Click **Add**.
Step 5 In **Basic Information** tab:

a) Choose **CTI Port** from **Product Type** drop-down list.
b) Enter Local CTI Port pool name in **Device Name** field, in *LCPxxxxFyyyy* format.

- LCP - identifies the CTI port as a local device

- xxxx - is peripheral ID of the Unified Communication Manager PIM
- yyyy - is the local CTI Port

c) Choose **Calling Search Space** from the drop-down list.

Step 6 Goto **Advanced Infromation** tab:

a) Choose **Device Pool** from the drop-down list.
b) Choose **Location** from the drop-down list.

Step 7 Goto **Lines** tab:

a) Click **Add** icon in **Lines** panel.
b) Choose directory number from **Pattern** drop-down list, in **Drin** Panel.
c) Choose **Route Partition Name** from drop-down list.

Step 8 Click **Save**.

What to do next

Repeat the above steps to create Network CTI port. Enter Network CTI Port pool name in **Device Name** field, in *RCPxxxxFyyyy* format.

- RCP - identifies the CTI port as a network device
- xxxx - is the peripheral ID of the Unified Communication Manager PIM
- yyyy - is the network CTI Port

> **Note** Local CTI port and Network CTI port should be same

Tag CTI Ports as Contact Center Agent Lines

Before you begin

Ensure CTI ports are added. See, Configure CTI Port, on page 31

> **Note** For 12000 agent deployment model, the CTI port for all three CUCM clusters should be tagged.

Perform the below steps for both LCP and RCP CTI ports:

Procedure

Step 1 Login to Cisco Unified Communication Domain Manager as provider, reseller or customer.

Step 2 Ensure that hierarchy is set to appropriate level.

Step 3 Navigate **Subscribe Management** > **Agent Lines**

Step 4 Click **Add**.

Step 5 Choose **Phones** from **Device Types** drop-down list.

Step 6 Choose **CTI Ports** from **Device Name** drop-down list.

Step 7 Choose **Line** from the drop-down list.

Step 8 Choose **Application User** from drop-down list.

Step 9 Click **Save**.

Configure Outbound Dialer

Complete the following procedure to configure Outbound Dialer:

- Configure Gateway, on page 34
- Configure Unified CVP, on page 36
- Configure Unified CCE, on page 36
- Configure Unified Communications Manager, on page 50

Configure Gateway

Note

- In small contact center agent deployment model customer can choose a dedicated or a shared outbound gateway. If it is shared gateway there should be a PSTN connectivity.
- Outbound Dialer do not support A-law, it is not instructed to configure the A-law under inbound dial-peer in the voice gateway.

Follow the below procedure to configure gateway/CUBE(E):

Procedure

Step 1 Create a voice encapsulation type with following voip parameters

Example:

```
voice service voip
     no ip address trusted authenticate
     mode border-element
     allow-connections sip to sip
     no supplementary-service sip refer
     supplementary-service media-renegotiate
     redirect ip2ip
     signaling forward none
sip
     header-passing
     error-passthru
     asymmetric payload full
     options-ping 60
```

```
     midcall-signaling passthru
     !
```

Step 2 Default, CPA is enabled for gateway/CUBE(E). Otherwise, enable CPA for CUBE(E).

Example:

```
voice service voip
 cpa
```

Step 3 Create a voice codec class

Example:

```
voice class codec 1
     codec preference 1 g729r8
     codec preference 2 g711ulaw
```

Step 4 Create dial peer configuration to reach the customer PSTN number.

Example:

```
dial-peer voice 978100 voip
     session protocol sipv2
     incoming called-number <Customer Phone Number Pattern>
     voice-class codec 1
     voice-class sip rel1xx supported "100rel"
     dtmf-relay rtp-nte sip-kpml
     no vad

dial-peer voice 97810 pots
     destination-pattern 97810[1-9]
     port 1/0:23
     forward-digits all
     progress_ind alert enable 8
```

Step 5 Create dial peer configuration to reach the agent extension (VOIP)

Example:

```
dial-peer voice 40000 voip
     description ***To CUCM Agent Extension***
     destination-pattern <Agent Extension Pattern to CUCM>
     session protocol sipv2
     session target ipv4:<CUCM IP Address>
     voice-class codec<Codec Preference number>
     voice-class sip rel1xx supported "100rel"
     dtmf-relay rtp-nte
     no vad
     !
```

Note In 12000 agent deployment model dial peer needs to be created for all 3 CUCM clusters.

Step 6 Create dial peer configuration to reach CVP

Example:

```
dial-peer voice 99995 voip
     description ******To CVP for IVR OB*****
     destination-pattern 9999500T
     session protocol sipv2
     session target ipv4:10.10.10.10
     codec g711ulaw
     voice-class sip rel1xx disable
     dtmf-relay rtp-nte h245-signal h245-alphanumeric
     no vad
```

```
!
!
```

Note

Step 7 Configure Transcoding Profile for CUBE E:

Example:

```
dspfarm profile 4 transcode universal
     codec g729r8
     codec g711ulaw
     codec g711alaw
     codec g729ar8
     codec g729abr8
     maximum sessions 250
     associate application CUBE
     !
```

Configure Unified CVP

Add Outbound Configuration to an Existing Unified CVP Call Server

Complete the following procedure to add Outbound configuration to an existing Unified CVP Call Server.

Procedure

Step 1 Go to Unified CVP OAMP server and login to Operations console page.

Step 2 Click the **Device Management** tab and open Unified CVP Call Server from the menu.

Step 3 Open a Call Server and click the **ICM** tab and add DNIS.

DNIS number should match with the label configured in the Network VRU Explorer for Outbound in Unified CCE.

Step 4 Click **Save** and deploy.

Step 5 Repeat step 3 for each CVP Call Server.

Configure Unified CCE

- Add DNP Host File, on page 49
- Outbound Option Enterprise Data, on page 49

Add Outbound Option Database Using ICMDBA Tool

Note

- For 2000, 4000 agent deployment models and small contact center, perform the configurations on Unified CCE Rogger.
- For 12000 agent deployment model, perform the configurations on Unified CCE Logger.

Procedure

Step 1 Select **Start** > **All Programs** > **Cisco Unified CCE Tools** > **ICMdba**. Click **Yes** at the warnings.

Step 2 Navigate to **Server** > **Instance** > **Logger**. Right-click on the logger that is installed and select **Create** to create the Outbound Option database.

Step 3 In the **Create Database** dialog box, click **Add** to open the Add Device dialog box. Click **Data**. Select the E drive. Leave the DB size with default value and click **OK** to return to the Create Database dialog box.

Step 4 In the **Add Device** dialog box, Click **Log**. Select the E drive. Leave the log size field with default value. Click **OK** to return to the Create Database dialog box.

Step 5 In the **Create Database** dialog box, click **Create**, then click **Start**. When you see the successful creation message, click **OK**, then click **Close**.

Configure the Logger for Outbound Option

Use this procedure to configure the Logger for Outbound Option.

You can (optionally) configure the Logger to enable Outbound Option and Outbound Option High Availability. Outbound Option High Availability facilitates two-way replication between the Outbound Option database on Logger Side A and the Outbound Option database on Logger Side B. Use the ICMDBA tool to create an outbound database on Side A and Side B; then set up the replication by using Web Setup.

Perform the following procedure on both the Side A and Side B Loggers to configure Outbound Option or Outbound Option High Availability. Both Logger machines must be up and operational.

Important Before you configure the Logger for Outbound Option High Availability:

- Confirm that an Outbound Option database exists on Logger Side A and Logger Side B.
- Create a Microsoft SQL Server user and assign that user the sysadmin privilege. Use the same username and password on Logger Side A and Logger Side B. (You use this username and password in the following procedure to configure Outbound Option and enable Outbound Option High Availability.)
- Assign the sysadmin privilege to the NT authority/System user.

Procedure

Step 1 Open the Web Setup tool.

Step 2 Choose **Component Management > Loggers**.

Step 3 Choose the Logger that you want to configure, and click **Edit**.

Step 4 Click **Next** twice.

Step 5 On the Additional Options page, click the **Enable Outbound Option** check box.

Step 6 Click the **Enable High Availability** check box to enable Outbound Option High Availability on the Logger. Checking this check box enables High Availability two-way replication between the Outbound Option database on Logger Side A and the Outbound Option database on Logger Side B. Two-way replication requires that you check this check box on the Additional Options page for both Logger Side A and Side B. If you disable two-way replication on one side, you must also disable it on the other side.

You must enable Outbound Option in order to enable Outbound Option High Availability. Similarly, if you have enabled High Availability, you must disable High Availability (uncheck the **Enable High Availability** check box) before you can disable Outbound Option (uncheck the **Enable Outbound Option** check box).

Step 7 If you enable High Availability, enter a valid public server hostname address for **Logger Side A** and **Logger Side B**. Entering a server IP address instead of a server name is not allowed.

Step 8 If you enable High Availability, enter the **SQL Server Admin Credentials** (**Username** and **Password**) for a user with the SQL Server System Admin privilege to establish two-way replication. Use the same credentials on Logger Side A and Logger Side B.

SQL replication requires that the correct SQL Server system admin username and password be in place when setting up High Availability. If you change the password for that SQL account, replication fails until you disable High Availability and re-enable it with the new username and password. Because of this requirement, be careful about how and when you change the password for that account.

You can use any valid SQL Server system administration account to disable High Availability. Once disabled, you can set any valid SQL Server system administration account when you re-enable High Availability.

Step 9 Select the **Syslog** box to enable the Syslog event feed process (cw2kfeed.exe).

Note The event feed is processed and sent to the Syslog collector only if the Syslog collector is configured. For more information about the Syslog event feed process, see the *Serviceability Guide for Cisco Unified ICM/Contact Center Enterprise* at https://www.cisco.com/c/en/us/support/customer-collaboration/unified-contact-center-enterprise/products-installation-and-configuration-guides-list.html.

Step 10 Click **Next**.

Step 11 Review the Summary page, and click **Finish**.

Configure Outbound Dialer

Procedure

Step 1 On the Unified CCE Admin Workstation Server, navigate to **Start** > **Cisco Unified CCE Tools** > **Administration Tools** > **Configuration Manager**.

Step 2 In the **Configuration Manager** window, select **Outbound** > **Dialer**.

Step 3 For Small Contact Center, click **Retrieve** > **Add** and configure:

a) Enter the Dialer name.
b) Enter the ICM Peripheral Name.
c) Enter Hangup Delay (1-10) value as `1`.
d) Enter Port Throttle value as `10`.
e) Click **Save**.

Step 4 Click the **Port Map Selection** tab to display the port map configuration.

Step 5 Click **Add** to configure a set of ports and their associated extensions.

Step 6 Click **OK**.

Step 7 Click **Save**, then click **Close**.

Note For different sub customers, the port and extension range can be same, because each sub customer has separate dialer.

Create Outbound PIM

Configure Media Routing Peripheral Gateway, then add Outbound PIM. For more information, see Configure MR Peripheral Gateway.

Configure SIP Outbound

Add Import Rule

Procedure

Step 1 Goto **Unified CCE AW-HDS-DDS** machine.

Step 2 Navigate to **Configuration Manager** > **Outbound Option** > **Import Rule** and click **Retrieve**.

Step 3 Click **Add**.

Step 4 In **Import Rule General** tab:

a) Enter **Import Name**.
b) Choose **Import Type** from the drop-down list.
c) Enter **Target Table Name**.
d) Browse **Import File Path**.

Note
- For the import type **Contact**, browse the Contact Import file. See, Create Contact Import File, on page 45
- For the import type **Do Not Call**, browse the Do Not Call List file. See, Create Do Not Call List, on page 46

e) Choose **Comma Delimited** option from **Import Data Type** panel.
f) Check **Overwrite** Table check box.

Note During Campaign, do not use both **Import File Path** and **Overwrite** option. Otherwise, dialer becomes unavailable to access records.

Step 5 Goto **Definition** tab:

a) Click **Add**.
b) Choose **Standard Column Type** from the drop-down list and retain the default values for remaining fields.

Step 6 Click **Save**.

Import Rule Deletion

When you delete an import rule, the corresponding contact table is deleted.

If you are using Outbound Option High Availability and either Side A or Side B is down when the rule is deleted, the corresponding table on that side is not deleted. However, when the side restarts, the table is then automatically deleted.

Add Query Rule

Before you begin

One or more Import rules must be defined. See Add Import Rule, on page 39

Procedure

Step 1 Goto **Unified CCE AW-HDS-DDS** machine.

Step 2 Navigate to **Configuration Manager** > **Outbound Option** > **Query Rule** and click **Retrieve**.

Step 3 Click **Add**.

Step 4 Enter **Query Rule Name**.

Step 5 Choose **Import Rule** from the drop-down list.

Step 6 Enter **Rule Clause**.

Step 7 Click **Save**.

What to do next

1. Goto **Configuration Manager** > **Tools** > **List Tools** > **Call Tye List** and add two call types; one for agent-based and another for IVR-based campaigns.
2. Goto **Configuration Manager** > **Tools** > **List Tools** > **Dialed Number / Script Selector List** and add two dialed numbers under Media routing domain. Map the dial numbers with the call types created in the previous step (one dial number for each call type).
3. Goto **Configuration Manager** > **Tools** > **Explorer Tools** > **Skill Group Explorer** and add a skill group under the call manger peripheral. Add a route for this skill group.
4. Goto **Configuration Manager** > **Tools** > **Explorer Tools** > **Agent Explorer** and add an agent. Associate the agent with the skill group created in the previous step.

Delete a Query Rule

When you delete a query rule, the corresponding Dialing List table is also deleted.

If you are using Outbound Option High Availability and either Side A or Side B is down when the rule is deleted, the corresponding table on that side is not deleted. However, when the side restarts, the table is then automatically deleted.

Add Campaign

- -
- -

Add IVR Based Campaign

Procedure

Step 1 Goto **Unified CCE AW-HDS-DDS** machine.

Step 2 Navigate to **Configuration Manager** > **Outbound Option** > **Campaign** and click **Retrieve**.

Step 3 Click **Add**.

Step 4 Enter **Campaign Name**.

Step 5 Goto **Campaign Purpose** tab:

a) Choose **Tranfer to IVR Campaign** option.
b) Check **Enable IP AMD** check box.
c) Choose **Transfer to IVR Route Point** option.

Step 6 Goto **Query Rule Selection** tab and click **Add**:

a) Choose **Query Rule Name** from the drop-down list and click **OK**.

Step 7 Goto **Skill Group Selection** tab:

a) Choose appropriate CUCM PG from **Peripheral** drop-down list, click **Retrieve**.
b) Choose **Skill Group** from the drop-down list.

c) Enter **Overflow Agents per Skill** value.
d) Enter **Dialed number**.
e) Enter **Records to cache** value.
f) Enter **Number of IVR Ports**.
g) Click **OK**.

Step 8 Goto **Call Target** tab, choose **Daylight Savings Zone** from the drop-down list.

Step 9 Click **Save**.

Add Agent Based Campaign

Procedure

Step 1

Step 2 Navigate to **Configuration Manager** > **Outbound Option** > **Campaign** and click **Retrieve**.

Step 3 Click **Add**.

Step 4 Enter **Campaign Name**.

Step 5 Goto **Campaign Purpose** tab:

a) Choose **Agent Based Campaign** option.
b) Check **Enable IP AMD** check box.
c) Choose **Transfer to Agent** option.

Step 6 Goto **Query Rule Selection** tab and click **Add**:

a) Choose **Query Rule Name** from the drop-down list and click **OK**.

Step 7 Goto **Skill Group Selection** tab:

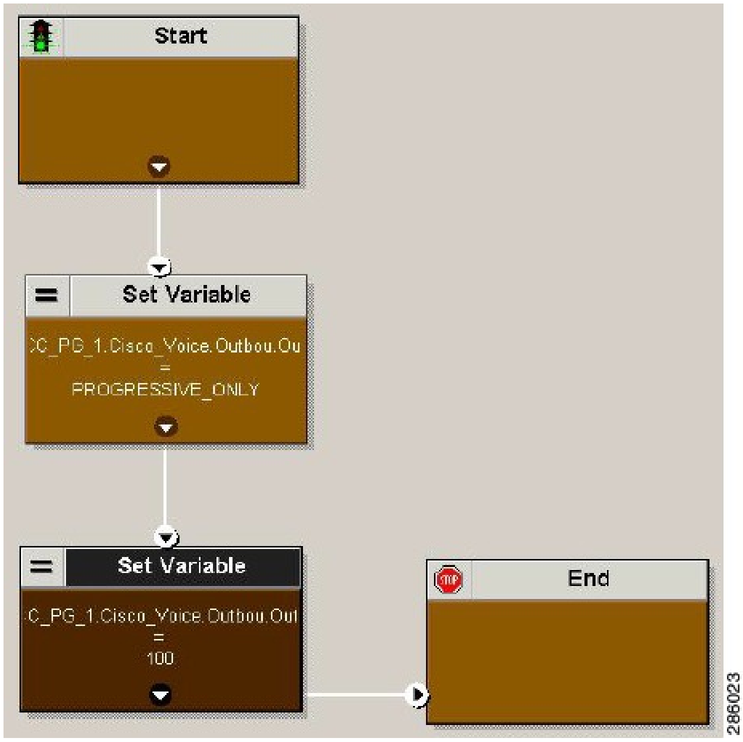
a) Choose appropriate CUCM PG from **Peripheral** drop-down list, click **Retrieve**.
b) Choose **Skill Group** from the drop-down list.
c) Enter **Overflow Agents per Skill** value.
d) Enter **Dialed number**.
e) Enter **Records to cache** value.
f) Enter **Number of IVR Ports**.
g) Click **OK**.

Step 8 Goto **Call Target** tab, choose **Daylight Savings Zone** from the drop-down list.

Step 9 Click **Save**.

Create Admin Script

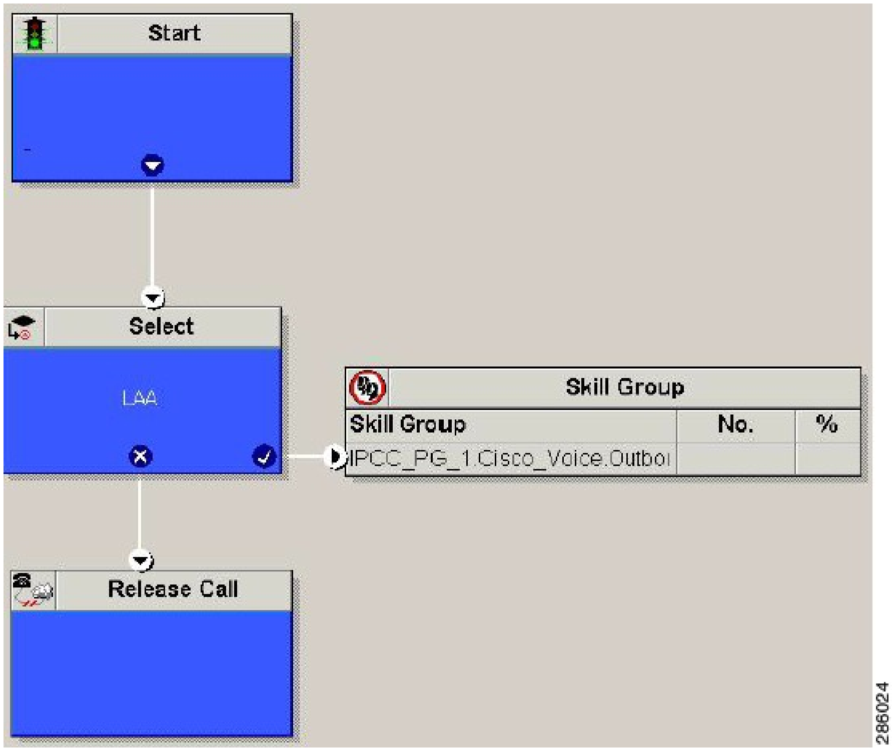
Figure 8: Create Admin Script



For more information, see Outbound Option Guide.

Add Routing Script for Agent Based Campaign

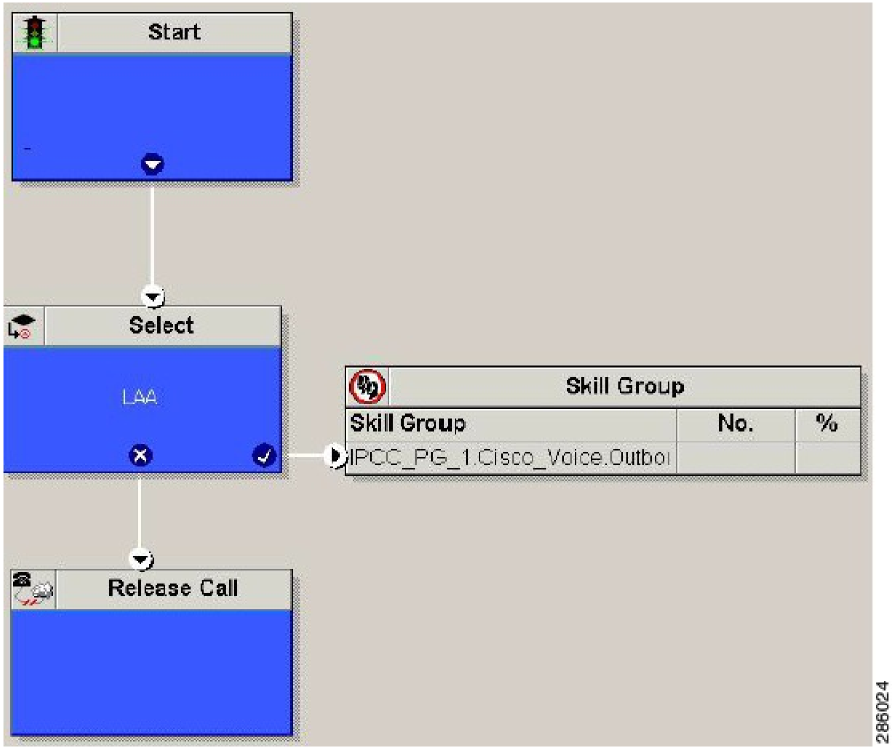
Figure 9: Add Routing Script for Agent Based Campaign



For more information, see Outbound Option Guide.

Add Routing Script for IVR Based Campaign

Figure 10: Add Routing Script for IVR Based Campaign



Configure the following for IVR based campaign:

Procedure

Step 1 Open Network VRU Explorer Tool from Configuration Manager tool. Add a label (label should match with the DNIS value configured in CVP call server) to the existing Network VRU of type 10 and select Media Routing type as "Outbound" from drop down list.

Step 2 Add the IVR based campaign.

What to do next

- Create Contact Import File, on page 45
- Create Do Not Call List, on page 46

Related Topics

Add IVR Based Campaign, on page 41

Create Contact Import File

When creating a contact import file, observe the format you designed according to the database rules set up in Import Rule Definition Tab Page.

The following example assumes that you have contact information with AccountNumber, FirstName, LastName, and Phone column types.

Procedure

Step 1 Using a text editor, create a text file that contains the information for these fields.

Step 2 Enter an account number, first name, last name, and phone number for each entry on a new line.

Use either Comma Delimited, Pipe Delimited, or Fixed Format, as defined on the Import Rule General Tab Page.

Step 3 Save the text file to the local server.

Example

The following is an example of a contact import file in the comma-delimited format:

```
6782,Henry,Martin,2225554444

3456,Michele,Smith,2225559999

4569,Walker,Evans,2225552000
```

The following is the same example in Fixed Format with the following column definitions:

- Custom - VARCHAR(4)
- FirstName - VARCHAR(10)
- LastName - VARCHAR(20)
- Phone - VARCHAR(20)

```
6782Henry     Martin              2225554444

3456Michele   Smith               2225559999

4569Walker    Evans               2225552000
```

Create Do Not Call List

When creating a Do_Not_Call list file, format it correctly using the following instructions.

Procedure

Step 1 Using a text editor, create a text file that contains all the do-not-call phone numbers.

Step 2 Enter a phone number for each Do Not Call entry on a new line.

Step 3 Observe the following characteristics for each Do Not Call entry:

- Each phone number can be a maximum of 20 characters long.
-

Step 4 Save the text file to the local server.

The following is an example of a Do_Not_Call list:

```
2225554444

2225556666

2225559999
```

To add a customer to this list, import a Do Not Call list.

The Campaign Manager reads from the Do_Not_Call table. Dialing List entries are marked as Do Not Call entries only when the Campaign Manager fetches the Dialing List entry *and only when there is an exact, digit-for-digit match*. This allows Do Not Call imports to happen while a Campaign is running without rebuilding the Dialing List.

Note If the Dialing List includes a base number plus extension, this entry must match a Do Not Call entry for that same base number and same extension. The dialer will not dial the extension.

Note To clear the Do Not Call list, import a blank file with the Overwrite table option enabled.

Install SIP Dialer Using Peripheral Gateway Setup

Procedure

Step 1 Stop all ICM Services.

Step 2 On the Unified CCE PG Side A and Side B, run Peripheral Gateway Setup. Select **Start** > **All Programs** > **Cisco Unified CCE Tools** > **Peripheral Gateway Setup**.

Step 3 In the **Cisco Unified ICM/Contact Center Enterprise & Hosted Components Setup** dialog, select an instance from the left column under **Instances**.

Step 4 Click **Add** in the **Instance Components** section.

The **ICM Component Selection** dialog opens.

Step 5 Click **Outbound Option Dialer**.

The **Outbound Option Dialer Properties** dialog opens.

Step 6 Check **Production mode** and **Auto start at system startup**, unless your Unified ICM support provider specifically tells you otherwise. These options set the Dialer Service startup type to Automatic, so the dialer starts automatically when the machine starts up.

The **SIP (Session Initiation Protocol)** Dialer Type is automatically selected.

Step 7 Click **Next**.

Step 8 On the **Outbound Option Dialer Properties** dialog, specify the following information:

- **Outbound Option server**—The hostname or IP address of the Outbound Option server in Unified CCE. This server is typically the same VM where the Outbound Option Campaign Manager (Dataserver Side A) is located.
- **Campaign Manager server A**—If the Campaign Manager is set up as duplex, enter the hostname or IP address of the machine where the Side A Campaign Manager is located. If the Campaign Manager is set up as simplex, enter the same hostname or IP address in this field and the **Campaign Manager server B** field. You must supply a value in this field.
- **Campaign Manager server B**—If the Campaign Manager is set up as duplex, enter the hostname or IP address of the machine where the Side B Campaign Manager is located. If the Campaign Manager is set up as simplex, enter the same hostname or IP address in this field and the **Campaign Manager server A** field. You must supply a value in this field.
- **Enable Secured Connection**— Allows you to establish secured connection between the following:
  - CTI server and dialer
  - MR PIM and dialer

  Check the **Enable Secured Connection** check box to enable secured connection.

  Note If you check the **Enable Secured Connection** check box, secured connection is established between the dialer and the servers, such as MR PIM and CTI server.

  Note Before you enable secured connection between the components, ensure to complete the security certificate management process.

  For more information, see the *Security Guide for Cisco Unified ICM/Contact Center Enterprise* at https://www.cisco.com/c/en/us/support/customer-collaboration/unified-contact-center-enterprise/products-installation-and-configuration-guides-list.html.
- **CTI server A**—The hostname or IP address of the VM with CTI server Side A. This server is typically the same VM where the PG is located (Call Server Side A).
- **CTI server port A**—The port number that the dialer uses to create an interface with CTI server Side A. The default is 42027 for non-secured connection and 42030 for secured connection. Make sure the CTI server port matches with the CG configuration. Locate the CTI OS Server port number by running the **Diagnostic Framework Portico** page from the call server machine, and selecting **ListProcesses**.
- **CTI server B**—The hostname or IP address of the VM with CTI server Side B.
- **CTI server port B**—The port number that the dialer uses to create an interface with CTI server Side B. The default is 43027 for non-secured connection and 43030 for secured connection.
- **Heart beat**—The interval between dialer checks for the connection to the CTI server, in milliseconds. The default value is 500.
- **Media routing port**—The port number that the dialer uses to create an interface with the Media Routing PIM on the Media Routing PG. The default is 38001. Make sure the Media routing port matches that of the MR PG configuration. For example, you can access this registry key: `Computer\HKEY_LOCAL_MACHINE\SOFTWARE\Cisco Systems, Inc.\ICM\mango\PG3A\PG\CurrentVersion\PIMS\pim1\MRData\Config\ApplicationTcpServiceName1`.

Step 9 Click **Next**. A **Summary** screen appears.

Step 10 Click **Next** to begin the dialer installation.

Optional - Edit Dialer Registry Value for AutoAnswer

If you enable auto answer in the CallManager with a zip tone, you must disable auto answer in the Dialer or Dialers, if there are more than one. A zip tone is a tone sent to the agent's phone to signal that a customer is about to be connected.

To disable auto answer in the Dialer, after the Dialer process runs for the first time, change the value of the following registry key to 0:

HKEY_LOCAL_MACHINE\SOFTWARE\Cisco Systems, Inc.\ICM\pra01\Dialer\AutoAnswerCall

For information on other Dialer registry settings, see the *Outbound Option Guide for Unified Contact Center Enterprise*.

Add DNP Host File

Complete this procedure to add DNP Host file.

Procedure

Step 1 In the C drive of the virtual machine where dialer is installed, navigate to \icm\customerInstanceName\Dialer directory.

Step 2 Modify the DNP Host file for static route mapping.

The format for a static route is `wildcard pattern, IP address or hostname of the Gateway that connects to the dialer, description.`

Example : `7????? (Dial pattern), 10.86.227.144 (gateway ip) , calls to agent extensions`

Note Repeat these steps for each sub customer Dialer.

Outbound Option Enterprise Data

In order for Outbound Option enterprise data to appear in the Cisco Agent Desktop Enterprise Data window, the administrator must edit the Default layout to include some or all Outbound Option variables. These variables are prefixed with “BA.” (Edit the default enterprise data layout in the Cisco Desktop Administrator.)

- BAAccountNumber
- BABuddyName
- BACampaign
- BADialedListID
- BAResponse
- BAStatus
- BATimeZone

> **Note**
> To enable the ECC variables, See Configure Expanded Call Variable. The BAStatus field is required. All other BA fields are optional for Progressive and Predictive modes. In Preview mode, the Skip button will not work if BADialedListID is not enabled.

- The BABuddyName field is required, if you want to see the customer's name being called.
- If a call is part of a Preview dialing mode campaign, the first letter in the BAStatus field entry is a P. If a call is part of a Direct Preview dialing mode campaign, the first letter in the BAStatus field entry is a "D."

Configure Unified Communications Manager

Add Normalization Script

This script is needed to disable Ringback during Transfer to Agent for SIP calls.

Procedure

Step 1 Log in to **Unified Communications Manager Administration** page.

Step 2 Navigate to **Devices** > **Device Settings** > **SIP Normalization Scripts**.

Step 3 Click **Add New**.
Displays **SIP Normalization Script** page.

Step 4 Enter **Name** of the script.

Step 5 Enter the following script in **Content** field:

```
M = {}
function M.outbound_180_INVITE(msg)
msg:setResponseCode(183, "Session in Progress")
end
return M
```

Step 6 Keep default values for remaining fields.

Step 7 Click **Save**.

Configure Trunk towards the Outbound Gateway

To configure trunk towards the outbound gateway, see Add SIP Trunks. While updating **SIP info** tab:

Procedure

Step 1 Enter IP address of outbound gateway in **Address IPv4** field.

Step 2 Choose newly added **Normalization Script** from the drop-down list.

Configure Post Call Survey

Complete the following procedures to configure post call survey:

- Configure Post Call Survey in CVP, on page 51
- Configure Unified CCE, on page 51

Configure Post Call Survey in CVP

Complete the following procedure to configure Post Call Survey in Unified CVP.

Procedure

Step 1 Log in to the Unified CVP Operations Console and choose **System** > **Dialed Number Pattern**.

Step 2 Enter the following configuration settings to associate incoming dialed numbers with survey numbers:

- **Dialed Number Pattern** - Enter the appropriate dialed number.

  The incoming Dialed Number for calls being directed to a Post Call Survey Dialed. This is the Dialed Number you want to redirect to the survey.

- **Enable Post Call Survey for Incoming Calls** - Select to enable post call survey for incoming calls.
- **Survey Dialed Number Pattern** - Enter the dialed number of the Post Call Survey. This is the dialed number to which the calls should be transferred to after the call flow is completed.
- Click **Save** to save the Dialed Number Pattern.

Step 3 Click **Deploy** to deploy the configuration to all Unified CVP Call Server devices.

Configure Unified CCE

Configure ECC Variable

You need not configure Unified CCE to use Post Call Survey, however, you can turn the feature off (and then on again) within an ICM script by using the ECC variable **user.microapp.isPostCallSurvey** and a value of n or y (value is case insensitive) to disable and re-enable the feature.

Configure the ECC variable to a value of n or y before the label node or before the Queue to Skillgroup node. This sends the correct value to Unified CVP before the agent transfer. This ECC variable is not needed to initiate a Post Call Survey call, but you can use it to control the feature when the Post Call Survey is configured using the Operations Console.

When the DN is mapped in the Operations Console for Post Call Survey, the call automatically transfers to the configured Post Call Survey DN.

Complete the following procedure to enable or disable the Post Call Survey:

Procedure

Step 1 On the Unified CCE Administration Workstation, using configuration manager, select the **Expanded Call Variable List tool**.

Step 2 Create a new ECC Variable with **Name:user.microapp.isPostCallSurvey**.

Step 3 Set **Maximum Length** to 1.

Step 4 Select the **Enabled** check box then click **Save**.

Configure a-Law Codec

Configure the following in Cisco HCS for CC core components to support a-law codec:

- Configure Gateway, on page 52
- Configure Unified CVP, on page 54
- Configure Unified Communication Manager, on page 56

Configure Gateway

- Configure Ingress Gateway, on page 52
- Configure VXML Gateway, on page 53

Configure Ingress Gateway

Procedure

Step 1 Add the voice class codec 1 to set the codec preference in dial-peer:

Example:

```
voice class codec 1
     codec preference 1 g729r8
     codec preference 2 g711alaw
     codec preference 3 g711ulaw

dial-peer voice 70021 voip
     description Used for Switch leg SIP Direct
     preference 1
     max-conn 225
     destination-pattern xxxx...... # Customer specific destination
     session protocol sipv2
     session target ipv4:###.###.###.### # IP Address for Unified CVP
     session transport tcp
     voice class codec 1
     voice-class sip options-keepalive up-interval 12 down-interval 65 retry 2
     dtmf-relay rtp-nte
     no vad
```

Step 2 Modify the dial-peer to specify the codec explicitly for a dial-peer:

```
dial-peer voice 9 voip
     description For Outbound Call for Customer
     destination-pattern <Customer Phone Number Pattern>
     session protocol sipv2
     session target ipv4:<Customer SIP Cloud IP Address>
     session transport tcp
     voice-class sip rel1xx supported "100rel"
     voice-class sip options-keepalive up-interval 12 down-interval 65 retry 2
     dtmf-relay rtp-nte
     codec g711alaw
     no vad

dial-peer voice 10 voip
     description ***To CUCM Agent Extension For Outbound***
     destination-pattern <Agent Extension Pattern to CUCM>
     session protocol sipv2
     session target ipv4:<CUCM IP Address>
     voice-class sip rel1xx supported "100rel"
     dtmf-relay rtp-nte
     codec g711alaw
```

Configure VXML Gateway

Procedure

Modify the following dial-peer to specify the codec explicitly for a dial-peer:

```
dial-peer voice 919191 voip
     description Unified CVP SIP ringtone dial-peer
     service ringtone
     incoming called-number 9191T
     voice-class sip rel1xx disable
     dtmf-relay rtp-nte
     codec g711alaw
     no vad

dial-peer voice 929292 voip
     description CVP SIP error dial-peer
     service cvperror
     incoming called-number 9292T
     voice-class sip rel1xx disable
     dtmf-relay rtp-nte
     codec g711alaw
     no vad

dial-peer voice 7777 voip
     description Used for VRU leg #Configure VXML leg where the incoming called
     service bootstrap
     incoming called-number 7777T
     dtmf-relay rtp-nte
     codec g711alaw
     no vad

dial-peer voice 5 voip
     description for SIP TTS Media Call
     preference 1
     session protocol sipv2
```

```
     session target ipv4: <ASR primary server IP>
     destination uri tts
     voice-class sip options-keepalive up-interval 12 down-interval 65 retry 2
     dtmf-relay rtp-nte
     codec g711alaw
     no vad

dial-peer voice 6 voip
     description for SIP ASR Media Call
     preference 1
     session protocol sipv2
     session target ipv4: <TTS primary server IP>
     destination uri asr
     voice-class sip options-keepalive up-interval 12 down-interval 65 retry 2
     dtmf-relay rtp-nte
     codec g711alaw
     no vad

dial-peer voice 7 voip
     description for SIP TTS Media Call
     preference 2
     session protocol sipv2
     session target ipv4: <ASR secondary server IP>
     destination uri tts
     voice-class sip options-keepalive up-interval 12 down-interval 65 retry 2
     dtmf-relay rtp-nte
     codec g711alaw
     no vad

dial-peer voice 8 voip
     description for SIP ASR Media Call
     preference 2
     session protocol sipv2
     session target ipv4: <TTS secondary server IP>
     destination uri asr
     voice-class sip options-keepalive up-interval 12 down-interval 65 retry 2
     dtmf-relay rtp-nte
     codec g711alaw
     no vad
```

Configure Unified CVP

Unified CVP does not require any specific configuration in OAMP.

You must convert the following files to A-law:

1. C:\inetpub\wwwroot\en-us\app
2. C:\inetpub\wwwroot\en-us\app\ag_gr
3. C:\inetpub\wwwroot\en-us\sys
4. C:\Cisco\CVP\OPSConsoleServer\GWDownloads in OAMP server
5. C:\Cisco\CVP\VXMLServer\Tomcat\webapps\CVP\audio

Note

- After converting the files in the OAMP server, access the Unified CVP OAMP page to upload the newly converted A-law files to the gateway.
- If gateways are previously used for u-law, then restart the gateway to clear the u-law files in the gateway cache.

Complete the following procedure to convert mu-law audio files to a-law format:

Procedure

Step 1 Copy the wav file from Unified CVP to your local desktop.
Step 2 Go to **All programs** > **Accessories** > **Entertainment**.
Step 3 Open the **Sound Recorder**.
Step 4 Select **File** and click **Open**.
Step 5 Browse for the mu-law audio file and click **Open**.
Step 6 Go to **Properties**.
Step 7 Click **Convert Now**.
Step 8 Select **CCITT A-Law** from **Format**.
Step 9 Click **OK**.
Step 10 Select **Files** > **Save As** and provide a filename.
Step 11 Copy the new a-law format file into the following directory of media server:

```
C:\inetpub\wwwroot\en-us\app
```

Enable Recording for Agent Greeting and Courtesy Callback

Complete the following procedure to enable recording for Agent Greeting and Courtesy Callback.

Procedure

Step 1 Open the call studio and go to the callback entry application.
Step 2 Double-click **app.callflow**.
Step 3 Go to **Record Name** element settings and change the File Type to **other** (default is wav).
Step 4 Set the MIME type to **audio/x-alaw-basic**.
Step 5 Set the File extension as **wav**
Step 6 Open the **RecordAgentGreeting** application and double-click **app.callflow**.
Step 7 Go to **Record Greeting With Confirm** element settings and change the File Type to **other** (default is wav).
Step 8 Set the MIME type to **audio/x-alaw-basic**.
Step 9 Set the File extension as **wav**.
Step 10 Validate, save, and deploy the application.

Step 11 Restart the Unified CVP services.

Configure Unified Communication Manager

Complete the following procedure to provision a-Law through Cisco Unified Communications Manager:

Procedure

Step 1 Login to the **Cisco Unified Communication Manager Administration** page.

Step 2 Navigate to **System** > **Service Parameter**.

Step 3 Choose publisher server from **Server** drop-down list.

Step 4 Choose **Cisco CallManager (Active)** from **Service** drop-down list.

Step 5 In **ClusterWide Parameters (System - Location and region)**, choose **Enabled for All Devices** from **G.711 A-law Codec Enabled** drop-down list.

Step 6 Choose **Disable** from following drop-down lists:

- **G.711 mu-law Codec Enabled**
- **G.722 Codec Enabled**
- **iLBC Codec Enabled**
- **iSAC Codec Enabled**

Step 7 Click **Save**.

Configure Unified CM Based Silent Monitoring

Perform the following steps to configure unified CM based silent monitoring:

- Enable or Disable the Built-in-Bridge. See, Enable or Disable the Built-in-Bridge
- Add Monitoring Calling Search Space for the device

Add Monitoring Calling Search Space for the device

Before you begin

Ensure that agent phones are added. See, Add Phones.

Note During CTIOS Server installation, for **IPCC Silent Monitor Type**, select **CCM Based**.

Procedure

Step 1 Log in to Unified Communication Domain Manager as provider, reseller or customer.

Step 2 Add Calling Search Space for monitoring purpose. See, Add Class of Service.

Step 3 Edit **Lines**, choose newly added **Calling Search Space** from the drop-down list. See, Edit Lines.

Step 4 Click **Save**.

Configure Unified Communication Manager

A Unified Communications Manager Music On Hold (MoH) server can generate MoH stream from an audio file or a fixed source. Either of this can be transmitted as unicast or multicast.

MoH server can be deployed in two modes.

1. Along with Unified CM on the same server for HCS for CC deployments with less than 1250 users in a CM Cluster.
2. As standalone node (TFTP/MoH Server) for HCS for CC deployments with more than 1250 users in a CM Cluster

- Configure Music On Hold Server Audio Source, on page 57
- Set up Service Parameters for Music on Hold, on page 58
- Set up Phone Configuration for Music on Hold, on page 58

Configure Music On Hold Server Audio Source

Procedure

Step 1 Login to **Cisco Unified Communications Manager Administration** page.

Step 2 Select **Media Resources** > **Music On Hold Audio Source**.

Step 3 Retain the default sample audio source.

Step 4 Select **Initial Announcement** from drop down list (optional).

Step 5 Click **Save**.

Step 6 Perform the following steps to create new Audio Source.

a) Click **Add New**.
b) Select MOH audio stream number from the drop down list.
c) Select MOH audio source file from the drop down list.
d) Enter the MOH source name .
e) Choose **Initial Announcement** from the drop-down list.
f) Click **Save**.

Set up Service Parameters for Music on Hold

Procedure

Step 1 Login to **Cisco Unified Communications Manager Administration** page.

Step 2 Select **System** > **Service Parameters**.

Step 3 Select the MoH server from the drop-down list .

Step 4 Select the app service from **Cisco IP Voice Media Streaming App Service** drop-down list.

Step 5 Select the required codec in the **Supported MOH Codecs** field and click **Ok**.

Step 6 Click **Save**.

Set up Phone Configuration for Music on Hold

Procedure

Step 1 Login to **Cisco Unified Communications Manager Administration** page.

Step 2 Select **Device** > **Phone**.

Step 3 Select the phone to configure MOH.

Step 4 Select a audio source from **User Hold MOH Audio Source** drop-down list.

Step 5 Select a audio source from **Network Hold MOH Audio Source** drop-down list.

Step 6 Click **Save** and click **Apply** and reset the phone.