Telephony and Media Provision

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Unified CCX Telephony and Media

The Unified CCX system uses a telephony resource called Computer Telephony Interface (CTI) ports to accept incoming calls and to place outbound calls. The Unified CCX system uses the following media resources to provide interactive services for calls:

- Unified CM Telephony—The Unified CCX Engine uses the Unified CM Telephony subsystem to send and receive calls from the Unified CM by interfacing with the CTI Manager through the Unified CM Telephony client.
- Cisco Media Termination (CMT)—The CMT channels provide media terminations in the Unified CCX for Unified CM Telephony Call Contacts. These channels enable the Unified CCX to play media to the connected party. DTMF digits are received out of band by the Unified CM Telephony subsystem.
- MRCP Automated Speech Recognition (MRCP ASR)—The ASR media resource allows callers to use speech to navigate menus and to provide other information to Unified CCX applications.
- MRCP Text-To-Speech (MRCP TTS)—The TTS media resource enables Unified CCX applications to play back documents to callers as speech.

Note Media resources are licensed and sold as Unified IP IVR ports. Although you can provision more channels than you are licensed for, licensing is enforced at run-time. If more channels are provisioned than licensed, the system will not accept the extra calls, because doing so would violate your licensing agreements.

The Unified CCX system uses *groups* to share telephony and media resources among different applications:

- *Call control groups* allow you to control how the system uses CTI ports. For example, you can reserve more ports for higher-priority applications or provide access to fewer ports for applications with less traffic.
- *Media resource groups* allow you to share media resources among different applications. For example, you can share ASR media resource groups with applications that collect caller information and applications that transfer calls to specific extensions.

The Unified CCX system also uses *triggers*, which are specified signals that invoke application scripts in response to incoming contacts.

Media Termination Groups

Media termination groups are associated with CTI port groups.

Note For Unified CM deployment, you can create and use additional CTI port groups as required.

If a CTI port group is selected to support media termination and if the number of channels are identical to both groups, the CTI port group is automatically created in the background. This auto creation feature eliminates the manual CTI port group creation process.

If you choose to override media termination, the call control channel chooses the media termination automatically. If you want to select a new dialog group, you can have more than one media termination option. The options are used in the order that is displayed in the drop-down list.

Channels Required to Process Calls

Unified CCX needs two types of channels to process calls:

- A *call control channel*, which is provisioned through the Unified CM Telephony subsystem and corresponds to CTI port resources in Unified CM.
- A *media channel*, which is provisioned through either the CMT subsystem or the MRCP subsystem and corresponds to the kernel resources for handling the media voice path with the caller.

Note MRCP channels also correspond to additional resources on the MRCP server for performing speech recognition.

Unified CCX needs access to a channel of each type to successfully process a call. However, the capabilities of the two channel types are not identical.

For example, consider a Unified CCX system provisioned with a single Unified CM Telephony call control channel (that is, a CTI port) and a single CMT channel. The system can handle one call at a time; when that call terminates, the system must reinitialize the channel resources before it can accept another call.

However, the time each channel takes to reinitialize is not equal—CMT channels take more time to reinitialize than CTI ports. For example:

- The Unified CM Telephony call control channel may take approximately 1 millisecond to reinitialize.

- The CMT channel may take approximately 200 milliseconds to reinitialize.

This example implies that the system will not be able to accept a new incoming call for 200 milliseconds after the first call terminates; although the Unified CM Telephony channel is available after one millisecond, the CMT channel is not and Unified CCX needs both channels to process a call.

Such a delay can become an issue when a Unified CCX system is experiencing a high load condition or needs to handle a burst of incoming calls. Consequently, CMT channels require a higher channel count provisioning.

Tip To provision Unified CCX systems to handle burst calls equally among all required resources, you must configure approximately 10 percentage more CMT channels than CTI ports, and approximately 10 percentage more MRCP channels than ASR licenses.

Provision Telephony and Media Resources

To provision telephony and media resources, complete the following tasks:

Procedure

Step 1 Provision the Unified CM Telephony subsystem.

Unified CM Telephony subsystem controls telephony resources for Unified CCX system.

Step 2 Provision the CiscoMedia subsystem.

CiscoMedia subsystem controls CMT media resources for Unified CCX system.

Step 3 Provision the MRCPASR subsystem.

MRCPASR subsystem controls ASR media resources for Unified CCX system.

Step 4 Provision the MRCP TTS subsystem.

MRCP TTS subsystem controls TTS media resources for Unified CCX system.

Provision Unified CM Telephony Subsystem

The Unified CM Telephony subsystem is the subsystem of the Unified CCXEngine that sends and receives call-related messages from the Unified CM CTI Manager through the Unified CM Telephony client. To enable your Unified CCX server to handle Cisco Unified Communications requests, you must provision the Unified CM Telephony subsystem. The Unified CM Telephony subsystem is available in all the Unified CCX license packages.

Note In previous versions of Unified CCX, it was necessary to configure Unified CM Telephony information using Unified CM. In Unified CCX, Unified CM Telephony configuration tasks are performed directly through Unified CCX Administration web pages.

To provision the Unified CM Telephony subsystem, complete the following tasks:

Procedure

Step 1 Configure a Unified CM Telephony Provider, if not already configured. Specify the server on which Unified CM CTI Manager is running, and provide a Unified CM user ID and password.

Step 2 Provision Unified CM Telephony call control groups.

Unified CM Telephony call control groups pool together a series of CTI ports, which the system then uses to serve calls as they arrive at the Unified CCX server.

Step 3 Provision a Unified CM Telephony trigger.

Unified CM Telephony triggers invoke application scripts in response to incoming contacts.

Step 4 Resynchronize Unified CM Telephony versions.

Resynchronize Cisco JTAPI Client

During the resynchronizing process, an additional check ensures that the Unified CM Telephony Client (also known as the Cisco JTAPI Client) are the same between the clients installed on the Unified CCX node and the Cisco Unified CM. If the Unified CCX detects a mismatch, the system downloads and installs the required version of Cisco JTAPI Client.

To resynchronize and view the status of Cisco JTAPI client, complete the following steps.

Procedure

Step 1 Choose **Subsystems** > **Cisco Unified CM Telephony** > **Cisco JTAPI Resync** from the Unified CCX Administration menu bar.

Step 2 The Cisco JTAPI Resync web page opens, displaying the status of Cisco JTAPI Client resynchronization.

At this point, if there is an incompatible version, it automatically downloads the new client.

Resynchronize Unified CM Telephony Data

This resynchronizing process ensures that the Unified CM Telephony user, the call control groups, and the triggers match the data of Unified CM being used.

To resynchronize the Unified CM Telephony data, complete the following steps.

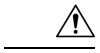
Procedure

From the Unified CCX Administration menu bar, choose **Subsystems** > **Cisco Unified CM Telephony** > **Data Synchronization** .

The Cisco Unified CM Telephony Data Synchronization web page opens after resynchronization, displaying the Data Resync status of Unified CM Telephony Port Groups and Unified CM Telephony Triggers.

Configure Unified CM Telephony Provider

The Unified CM Telephony Provider web page is a read-only page that displays the latest configured information.



Caution Some setups may prevent the Unified CM directory administrator from creating new Unified CM Telephony providers in a multiserver configuration. If this setup applies to you, be sure to delete preexisting Unified CM Telephony providers before creating new Unified CM Telephony providers. For example, if the Unified CM Telephony provider prefix is *cmtelephony* and you have a two-server configuration (*node_id1* and *node_id2*), you must delete both *cmtelephony_<node_id1>* and *cmtelephony_<node_id2>*. If you do not verify and delete preexisting Unified CM Telephony providers, the Unified CM Telephony subsystem issues an error and will not allow you to create Unified CM Telephony providers from the Unified CM Telephony Provider Configuration web page.

Procedure

Step 1 Choose **Subsystems** > **Cisco Unified CM Telephony** > **Provider** from the Unified CCX Administration menu bar.

The Cisco Unified CM Telephony Provider web page opens.

The following table describes the read-only fields displayed in the Unified CM Telephony Provider Configuration web page.

Field Heading	Description
Primary Unified CM Telephony Provider	IP address of the Server, running Unified CM CTI Manager in the cluster. This is usually the first CTI Manager or Cisco Unified CM Telephony Provider selected by the Unified CCX user for Unified CM Telephony subsystem using **System** > **Cisco Unified CM Configuration** web page.

Field Heading	Description
Secondary Unified CM Telephony Provider	IP address of the second Server, running Unified CM CTI Manager in the cluster. This is usually the second CTI Manager or Cisco Unified CM Telephony Provider selected by the Unified CCX user for Unified CM Telephony subsystem using **System** > **Cisco Unified CM Configuration** web page. **Note** If you have selected only one Unified CM Telephony provider, this field will be blank.
User Prefix	User prefix for the Unified CM user IDs created in Unified CM.

Step 2 To modify the Unified CM Telephony subsystem, click **Modify Cisco Unified CM Telephony Provider Information** icon that displays in the tool bar in the upper left corner of the window. The Cisco Unified CM Configuration web page opens.

Add New Call Control Group

The Unified CCX system uses Unified CM Telephony call control groups to create a series of CTI ports. The system uses the CTI ports to serve calls as they arrive at or depart from the Unified CCX server. You can create multiple Unified CM Telephony call control groups to share and limit the resources that are used by specific applications.

To configure a new Unified CM Telephony call control group, complete the following steps.

Procedure

Step 1 From the Unified CCX Administration menu bar, choose **Subsystems** > **Unified CM Telephony** > **Call Control Group**.

The Cisco Unified CM Telephony Call Control Group Configuration web page opens, which displays the existing Unified CM Telephony Call Control Group information, if any.

Step 2 Click the **Add New** icon that is displayed in the tool bar in the upper left corner of the window or the **Add New** button at the bottom of the window to create a new CTI port. The Cisco Unified CM Telephony Call Control Group Configuration web page opens.

Note
You can create only one call control group of the Outbound type, in which the number of CTI ports must be always equal to or greater than the licensed Outbound IVR ports.

Create the CTI ports through the Publisher for both nodes in a HA over WAN deployment.

Step 3 Use this web page to specify the following information:

Page Area	Field	Description
Group Information	Group ID	Corresponds to the trunk group number reported to Cisco Unified Intelligent Contact Management Enterprise (Unified ICME) when the Unified CCX server is part of the Unified ICME solution. The value for this field is automatically generated. **Note** If a Stop icon appears beside the Group ID (on the Cisco Unified CM Call Control Group Configuration list page), it indicates that the data is invalid or out of sync with Unified CM data; if a Head icon appears, the group is valid.
	Description	Description of the Group ID. Press the **Tab** key to automatically populate the Description field.
Group Information (continued)	Number of CTI Ports	Number of CTI ports assigned to the call control group. This is a mandatory field. If you have a Premium license with an Outbound IVR license, you can create only one Outbound call control group with a minimum licensed number of IVR ports or more. The number of CTI ports for an outbound type of call control group can be modified but not below the licensed ports for Outbound IVR. This rule does not apply to inbound type call control groups. You can continue to create more inbound type call control groups. **Note** If this field is set to <*n*>, the system creates <*n*> ports for each Unified CCX Engine node (node in which Unified CCX Engine component is enabled).
	Media Termination Support	Enables the auto-creation of media termination groups. This is a mandatory field. Yes = Provides automatic media termination if the CTI port group is successful. No = Media termination port group is not created (default).
	Group Type	Select the group type for the call control group using this radio button. The choices are Inbound and Outbound. This is a mandatory field and the Inbound radio button is enabled by default. You cannot change the group type from Outbound to Inbound and vice versa. The Outbound type call control group will be displayed only if you have uploaded the Outbound IVR license on top of the premium license in your Unified CCX.
Directory Number Information	Device Name Prefix	The Device Name Prefix (DNP) given to all of the CTI ports in this group. This is a mandatory field. The CTI ports for this port group are restricted to a maximum of 5 characters and has the following format: `<deviceprefix>_<directoryno>` For example, if the Device Name Prefix is CTP and the starting Directory Number is 7000, the CTI port that is created in Unified CM can have the device name CTP_7000.

Page Area	Field	Description
Select **Server for Telephony Port Group Configuration** (displayed only in a HA over WAN deployment).		
	Select Server	This field is displayed only in a HA over WAN deployment and it displays the different Unified CCX nodes that are available in a HA over WAN deployment in a drop-down list. In a HA over WAN setup, you need to configure directory information along with Unified CM-specific information for the ports in each node. Once you select a node, all configuration details displayed below this field will be specific to the selected node only. So, if you update any node-specific parameters (below the **Select Server** field), it will be applicable only to the ports specific to the selected node. But, if you update any configuration data above the **Select Server** field, it will be applicable for the ports in both the nodes except for the **Number of s** field. **Note** You need to ensure that the values in **Number of s** field for both the nodes are the same. If you modify this field, the number of ports is modified for the selected node only as the device pool selection for both nodes could be different in a HA over WAN deployment. If you click **Add** before updating this value for either of the node, the port group for that node will be marked with a red cross in the main Cisco Unified CM Telephony Call Control Group Configuration web page to signify the fact that the number of ports between the two nodes is different and the other node should also be updated. In such a scenario, click the hyperlink for the node that is tagged in red; and from the Cisco Unified CM Telephony Call Control Group Configuration page for the selected node, update the value in the **Number of CTI Ports** field and click **Update** to ensure the number of CTI ports for both the nodes are the same. After you configure the data for the selected node and click **Add** or **Update**, the updated configuration information will be saved. For detailed information on behavior in HA over WAN scenario, refer to the https://www.cisco.com/en/US/products/sw/custcosw/ps1846/products_implementation_design_guides_list.html. In case of LAN deployment, this field is not displayed, as the same configuration data will be applicable for both the nodes in the cluster.

Page Area	Field	Description
Directory Number Information	Starting Directory Number	A unique phone number. The value can include numeric characters and special characters (#) and (*). The specified number of ports will be created starting from the value specified in this field. The Directory Number that you enter can appear in more than one partition. This is a mandatory field. **Note** When a pattern is used as a Directory Number, the phone display and the caller ID display on the dialed phone will contain characters other than digits. To avoid this, provide a value for Display (Internal Caller ID), Line Text Label, and External Phone Number Mask.
	Device Pool	Set of common characteristics for devices, such as region, date/time group, softkey template, and MLPP information to which you want to assign this phone. **Note** The support for having multiple device pools associated with the call control group(s) has been withdrawn in Unified CCX. Manually assign a single device pool to each call control group if you have multiple device pools associated with call control group(s) in an older version of Unified CCX. In a HA over WAN setup, you need to configure directory information along with Unified CM-specific information for the ports in each node. Once you select a node, all configuration details displayed below this field will be specific to the selected node only. So, if you update any node-specific parameters (below the **Select Server** field), it will be applicable only to the ports specific to the selected node. But, if you update any configuration data above the **Select Server** field, it will be applicable for the ports in both the nodes except for the **Number of CTI Ports** field. **Note** You need to ensure that the values in **Number of CTI Ports** field for both the nodes are the same. If you modify this field, the number of ports is modified for the selected node only as the device pool selection for both nodes could be different in a HA over WAN deployment. If you click **Add** before updating this value for either of the node, the port group for that node will be marked with a red cross in the main Cisco Unified CM Telephony Call Control Group Configuration web page to signify the fact that the number of ports between the two nodes is different and the other node should also be updated. In such a scenario, click the hyperlink for the node that is tagged in red; and from the Cisco Unified CM Telephony Call Control Group Configuration page for the selected node, update the value in the **Number of CTI Ports** field and click **Update** to ensure the number of CTI ports for both the nodes are the same.
	DN Calling Search Space	A collection of partitions that are searched to determine how a dialed number should be routed. The calling search space for the device and the calling search space for the directory number get used together. The directory number calling search space takes precedence over the device calling search space. For more information, see the *System Configuration Guide for Cisco Unified Communications Manager*.

Page Area	Field	Description
	Location	The Cisco Unified Communications phone location setting specifies the total bandwidth that is available for calls to and from this location. A location setting of **HUB_NONE** means that the location feature does not keep track of the bandwidth that this Cisco Unified Communications phone consumes.
Advanced Directory Number Information (only available if you click **Show More**)		
Directory Number (continued)	Alerting Name ASCII	This information is automatically populated based on the configuration in the Unified CM setup and displays the ASCII name filed used in one of the following situations: • If the device is not capable of handling the Unicode strings • If the locals on endpoint devices do not match • If the Unicode string is not specified
	Redirect Calling Search Space	A collection of partitions that are searched to determine how a redirected call is routed. Redirect Calling Search Space options: **Note** **DN Calling Search Space** is deprecated. Use **Calling Party** or **Redirect Party** instead. • **DN Calling Search Space—**This option enables the CTI port to use its directory number CSS when performing a redirect transfer. • **Calling Party—**This option enables the CTI port to use the calling party's CSS when performing a redirect / consult transfer. • **Redirect Party—**This option enables the CTI port to use the CTI port's CSS to control the redirect.
	Media Resource Group List	A prioritized grouping of media resource groups. An application chooses the required media resource, such as a Music On Hold server, from the available media resources according to the priority order that is defined in a Media Resource Group List. If you choose <none>, Unified CM uses the Media Resource Group that is defined in the device pool.

Page Area	Field	Description
Directory Number Setting	Voice Mail Profile	A list of profiles defined in the Voice Mail Profile Configuration. The first option is <None>, which is the current default Voice Mail Profile that is configured in the Voice Mail Profile Configuration.
	Presence Group	See the *Cisco Unified Communications Manager Administration Guide* for detailed information on how to configure presence groups.
	Require DTMF Reception	A Unified CM radio button to determine if DTMF reception is required. Yes is selected by default. If you select No, a warning message is displayed.
	AAR Group	Automated Alternate Routing (AAR) group for this device. The AAR group provides the prefix digits that are used to route calls that are otherwise blocked due to insufficient bandwidth. An AAR group setting of <None> specifies that no rerouting of blocked calls will be attempted.
	User Hold Audio Source	Audio source heard by the caller when the Unified CCX Script places the caller on Hold by using the Hold Step (when you press the hold key).
	Network Hold Audio Source	Audio source heard by the caller when Unified CCX performs a Consult Transfer (when Unified CCX calls an agent). Use this entry for the .wav file (for example, a .wav file playing a ringback tone) to be played to the caller during this Consult Transfer.
Call Forward and Pickup Settings	Call Pickup Group	The number that can be dialed to answer calls to this directory number in the specified partition.
	Display	Use a maximum of 30 alphanumeric characters. Typically, use the user name or the directory number (if you use the directory number, the person receiving the call may not see the proper identity of the caller). Leave this field blank to have the system display the extension.
	External Phone Number Mask	Phone number (or mask) that is used to send Caller ID information when a call is placed from this line. You can enter a maximum of 24 number, the international escape character +, *, # and "X" characters. The X characters represent the directory number and must appear at the end of the pattern. For example, if you specify a mask of 972813XXXX, an external call from extension 1234 displays a caller ID number of 9728131234.

Step 4 Click **Add** or **Save**.

The Unified CM Telephony Call Control Group Configuration summary web page opens. The corresponding CTI ports are created in the Unified CM Telephony call control group. The new call control group appears in the list of call control groups in the Cisco Unified CM Telephony Call Control Group Configuration web page.

Add Unified CM Telephony Trigger

You must configure Unified CM Telephony triggers to invoke application scripts in response to incoming contacts. A Unified CM Telephony trigger responds to calls that arrive on a specific route point by selecting telephony and media resources to serve the call and invoking an application script to handle the call. The Unified CM Telephony triggers are available with all Unified CCX license packages.

Unified CM Telephony trigger settings include:

- **Session** information, such as the application to associate with the trigger, Maximum Number of sessions allowed, and the Idle Timeout value.
- **CTI** information, such as a CTI port device and CTI route points for each call Unified CCX simultaneously places or accepts.
- **Directory Number** information, such as the Voice Mail Profile and Calling Search Space.
- **Call Forward and Pickup** instructions.

To add and configure a Unified CM Telephony trigger, complete the following steps.

Procedure

Step 1 From the Unified CCX Administration menu bar, choose **Subsystems** > **Cisco Unified CM Telephony** > **Triggers**.

The Unified CM Telephony Trigger Configuration web page opens displaying the following fields.

Field	Description
Route Point	Available CTI route point, which is the directory number associated with the trigger.
Application	Application name to associate with the trigger.
Sessions	Maximum number of simultaneous calls that the trigger can handle.
Enabled	True if the trigger is enabled; False if the trigger is disabled.

Note
If you try to delete a trigger associated with an outbound call control group, then the campaigns associated with the trigger become invalid and the application also gets deleted. In such cases, when you click the **Delete** icon or button, a dialog box opens to confirm your action. Click **OK** if you want to delete the trigger and disassociate the campaigns associated with it. If you delete a trigger and navigate to the Campaign Configuration web page, you will also see an alert regarding the missing trigger association for that campaign.

Step 2 Click the **Add New** icon that is displayed in the tool bar in the upper left corner of the window or the **Add New** button that is displayed at the bottom of the window.

The Unified CM Telephony Trigger Configuration web page opens.

Step 3 Use this web page to specify the following mandatory fields:

Field	Description
Directory Information	

Field	Description
Directory Number	A unique phone number. To support E.164 compliance, Unified CCX allows you to add a plus sign (+) before the agent extension or a route point directory number followed by 15 characters which consist of numerals and the following special characters: uppercase letter X, hash (#), square brackets ([ ]), hyphen (-), and asterisk (*). • Supports only route point directory numbers and Finesse agent and supervisor extensions. • Does not support directory number for a Call Control Group configuration. • Does not support outbound. **Note** • +1234 and 1234 are two different directory numbers. • The square brackets ([ ]) enclose a range of values. • For more information, see the "Wildcards and Special Characters in Route Patterns and Hunt Pilots" section in the *System Configuration Guide for Cisco Unified Communications Manager*. • For more information, see the "Wildcards and Special Characters in Route Patterns and Hunt Pilots" section in the *System Configuration Guide for Cisco Unified Communications Manager*. **Examples:** • Valid directory numbers—+1223* or *#12#* • Invalid directory numbers—91X+ or +-12345 **Note** Use of two(2) wildcard CTI Route Points that overlap with each other is not supported. For example, Route Point 1: 123XXXX and Route Point 2: 1234XXX overlap with one another and is not supported. However, a wildcard CTI Route point can overlap with a full DID (best match pattern) that doesn't contain a wildcard. For example, Route Point 1: 123XXXX and Route Point 2: 1234567 is supported.
Trigger Information	
Language	Choose the default language to associate with the incoming call when the application is started from this drop-down menu. **Note** To add a Language option, click **Edit** button. The User Prompt dialog box opens. Enter a locale string value and click **OK**. The User Prompt dialog box closes, and the name of the language opens in the Language field in the Unified CM Telephony Configuration web page.
Application Name	From the drop-down menu, choose the application to associate with the trigger.
Device Name	A unique identifier for this device, consisting of alphanumeric characters, dots, dashes, or underscores.

Field	Description
Description	A descriptive name for the CTI route point.
Call Control Group	Choose the call control group to associate with the trigger from this drop-down menu. For Outbound IVR Dialer, you must select the call control group from Outbound type call control group list. The route point should be created on Unified CM. Once you assign the Outbound group for a trigger, you cannot change it to an Inbound group and vice versa.
Advanced Configuration (available only if you click **Show More**).	
Advanced Trigger Information	
Enabled	Radio buttons to choose the required option: • **Yes—**enable the trigger (default) • **No—**disable the trigger.
Maximum Number of Sessions	The maximum number of simultaneous calls that this trigger can handle. The number is actually governed by the Unified CM (10,000 for each separate line). However in Unified CCX, this number is restricted to the maximum number of sessions. Any call after this number is exceeded gets the busy tone.
Idle Timeout (in ms)	The number of milliseconds (ms) the system should wait before rejecting the Unified CM Telephony request for this trigger.
Override Media Termination	Radio buttons to choose the required options: **Yes—**Override media termination. **No—**Enable media termination (default). If you select Yes, two panes open: • Selected Dialog Groups displays the default or selected group. • Available Dialog Groups lists the configured dialog.
CTI Route Point Information	
Alerting Name ASCII	This information is automatically populated based on the configuration in the Unified CM setup and displays the ASCII name filed used in one of the following situations: • If the device is not capable of handling the Unicode strings • If the locals on endpoint devices do not match • If the Unicode string is not specified
Device Pool	The device pool to which you want to assign this route point. A device pool defines sets of common characteristics for devices, such as region, date/time group, softkey template, and MLPP information.

Field	Description
Location	The total bandwidth that is available for calls to/from this location. A location setting of **HUB_NONE** indicates that the locations feature does not keep track of the bandwidth used by this route point.
Directory Number Settings	
Partition	The partition to which the Directory Number belongs. The Directory Number field value must be unique within the partition that you choose. If you do not want to restrict access to the Directory Number, select <**None**> as the partition setting.
Voice Mail Profile	A list of profiles defined in the Voice Mail Profile Configuration. The first option is <**None**>, which is the current default Voice Mail Profile that is configured in the Voice Mail Profile Configuration.
Calling Search Space	A collection of partitions that are searched for numbers that are called from this directory number. The specified value applies to all devices that use this directory number. For example, assume you have two calling search spaces: Building and PSTN. Building only allows users to call within the building, while PSTN allows users to call both in and outside the building. You could assign the phone to the Building calling search space and the line on your phone to the PSTN calling search space. For more information, see the *System Configuration Guide for Cisco Unified Communications Manager*.
Calling Search Space for Redirect	By default, Cisco Unified Communications Manager uses the original calling party's calling search space (CSS) to process the redirected call from a Unified CCX Trigger to a Unified CCX CTI Port. This default behavior requires the partition of the Unified CCX CTI ports to be a member of the original calling party's CSS even if the partition of the CTI Route Point/Unified CCX Trigger is accessible to the calling device's CSS and the CSS of the CTI Route Point/Unified CCX Trigger contains the partition of the Unified CCX CTI Ports. You can modify this behavior using the drop-down list to instruct Cisco Unified Communications Manager which CSS to use when redirecting the call from the CTI Route Point to the CTI Port. Calling Search Space for Redirect options: • **Default Calling Search Space—**CSS of the calling device • **Calling Address Search Space—**CSS of the calling device • **Route Point Address Search Space—**CSS of the CTI Route Point (Trigger)
Presence Group	A list of groups to integrate the device with the iPass server. The device/line information is provided for integrating applications.
Call Forward and Pickup Settings	

Field	Description
Forward Busy	Check one of the following options: **Voice Mail—**Check this box to use settings in the Voice Mail Profile Configuration window. **Note** When this box is checked, Unified CM ignores the settings in the Destination box and Calling Search Space. **Destination—**To use any disable phone number, including an outside destination. **Calling Search Space—**To apply the above setting all devices that are using this directory number. **Note** For limiting the number of calls per application in Unified CCX system, see the Unified CM Telephony Triggers for Unified CCX Queuing, on page 16 section.
Display	Use a maximum of 30 alphanumeric characters. Typically, use the user name or the directory number (if using the directory number, the person receiving the call may not see the proper identity of the caller). Leave this field blank to have the system display an extension.
External Phone Number Mask	Phone number (or mask) that is used to send Caller ID information when a call is placed from this line. You can enter a maximum of 24 number, the international escape character +, *, # and "X" characters. The X characters represent the directory number and must appear at the end of the pattern. For example, if you specify a mask of 972813XXXX, an external call from extension 1234 displays a caller ID number of 9728131234.

Step 4 Click **Add** or **Save** to save the changes. The specified route point is created on the Unified CM.

The Unified CM Telephony Trigger Configuration web page opens and displays the new Unified CM Telephony trigger.

Additional Unified CM Telephony Information

This section includes the following topics:

- Unified CM Telephony Triggers for Unified CCX Queuing, on page 16
- Unified CM Telephony Information Resynchronization, on page 17

Unified CM Telephony Triggers for Unified CCX Queuing

When limiting the number of calls per application in Unified CCX applications, you need to take care to coordinate the Unified CM Telephony trigger Maximum Number of Sessions limit with the Media Group session limit.

For example, if you are using Unified CCX for queuing calls and set the Unified CM Telephony trigger Maximum Number of Sessions limit on Unified CCX to 4 and set the Call Forward and Pickup Settings to

send the fifth call to voice mail. To make this happen, you must configure the Media Group Session Limit to the identical setting (4). This will cause Unified CM to forward the next incoming call to voice mail (once the CTI New Call Accept timer setting expires).

The disadvantage of this approach is that you need to define more media groups for each application and you cannot share the same set of media groups across multiple applications.

Unified CM Telephony Information Resynchronization

If the Unified CM Telephony information (Unified CM Telephony users, CTI ports, triggers, SRTP) in the Unified CM is missing or not in sync with Unified CCX data, choose **Subsystems** > **Cisco Unified CM Telephony** > **Data Resync** from the Unified CCX Administration menu bar. Unified CCX checks whether:

- The Unified CM Telephony users exist in Unified CM.
- All the ports belonging to the Port Group exist in Unified CM.
- The port group's data is in sync with Ports data in Unified CM.
- The ports' association to users are correct.
- The route point exists in Unified CM.
- The triggers data is in sync with the Route Point data in the Unified CM.
- The route points have been associated with all the Unified CM Telephony users in Unified CM.
- The Unified CM configuration is in sync with Unified CCX for SRTP.

Unified CCX synchronizes the data by:

- Creating any missing users
- Creating any missing ports
- Modifying out-of-sync ports
- Associating CTI Ports to Unified CM Telephony users. (For example, associating CTI Ports created for Node 1 to the Unified CM Telephony User for Node 1, and so on.)
- Creating any missing route points
- Modifying out-of-sync route points
- Associating route points to all the Unified CM Telephony users.
- Modifying Unified CM configuration, such as user roles and CAPF profiles for JTAPI telephony users to match SRTP configuration.

Cisco Media Subsystem

The Cisco Media subsystem is a subsystem of the Unified CCX Engine. The Cisco Media subsystem manages the CMT media resource. CMT channels are required for Unified CCX to be able to play or record media.

The Cisco Media subsystem uses *dialog groups* to organize and share resources among applications. A dialog group is a pool of *dialog channels* in which each channel is used to perform *dialog interactions* with a caller, during which the caller responds to automated prompts by pressing buttons on a touch-tone phone.

Note The built-in grammars and grammar options that are supported by Unified CCX when using an MRCP dialog channel is determined by the MRCP speech software you purchase. See the software vendor for information about what built-in grammars and features are supported.

To enable your Unified CCX applications to handle simple DTMF-based dialog interactions with customers, you must provision the Cisco Media subsystem to configure CMT dialog groups.

Caution All media termination strings begin with `auto` and contain the same ID as the call control group—not the CMT dialog group. If the default media termination is configured and the ID differs, follow the procedure provided in the **Add CMT Dialog Control Group**.

Add CMT Dialog Control Group

To add a CMT dialog control group, complete the following steps.

Procedure

Step 1 From the Unified CCX Administration menu bar, choose **Subsystems** > **Cisco Media**.

The Cisco Media Termination Dialog Group Configuration web page opens. Any preconfigured entry is listed on this page with the following information:

Field	Description
GroupID	The unique Group ID associated with the media.
Description	CMT group description. **Note** The ID in this field need not necessarily match the CMT group ID.
Channels	Number of channels associated with the group.

Step 2 Click **Add New** icon at the top or **Add New** button at the bottom of the window. The Cisco Media Termination Dialog Group Configuration web page opens.

Note
By default, a Unified CM Telephony Call Control Group with Group ID 0 is created.

Step 3 Use this web page to specify the following fields.

Field	Description
Group ID	A Group ID value unique within all media group identifiers, including ASR group identifiers. This is a mandatory field.
Description	Description for the Cisco Media Termination Dialog group.
Number of Licensed IVR ports	Number of licensed IVR ports. Display only.
Maximum Number Of Channels	Maximum number of channels associated with this group. This is a mandatory field. **Note** You can specify any value for Maximum Number Of Channels, but restrictions are placed on the system when a call is made. This restriction is imposed by the number of licensed IVR ports on your system. This is a mandatory field.

Step 4 Click **Add** icon that displays in the tool bar in the upper left corner of the window or the **Add** button that displays at the bottom of the window.

The CMT Dialog Group Configuration web page opens, displaying the new CMT dialog group.

You are now ready to provision MRCP ASR and MRCP TTS subsystems.

ASR and TTS in Unified CCX

Unified CCX supports ASR and TTS through two subsystems:

MRCP ASR
This subsystem allows users to navigate through a menu of options by speaking instead of pressing keys on a touch-tone telephone.

MRCP TTS
This subsystem converts plain text (UNICODE) into spoken words to provide a user with information, or prompt a user to respond to an action.

Note Only G.711 codec is supported for ASR/TTS integrations.

Prepare to Provision ASR/TTS

It is the responsibility of the customer to perform the following tasks:

- Order ASR/TTS speech servers from Cisco-supported vendors.

Note For more information on supported speech servers for Unified CCX, see the Unified CCX Compatibility related information, located at: https://www.cisco.com/c/en/us/support/customer-collaboration/unified-contact-center-express/products-device-support-tables-list.html.

- Work with the ASR/TTS vendor to size the solutions.
- Provision, install, and configure the ASR/TTS vendor software on a different server (in the same LAN) and not where the Unified CCX runs. (see the Unified CCX Compatibility related information, located at: https://www.cisco.com/c/en/us/support/customer-collaboration/unified-contact-center-express/products-device-support-tables-list.html.).
- Before uploading a ASR/TTS script to Unified CCX Administration, validate the script against the capabilities and specifications supported by the ASR/TTS vendor.

Provision of MRCP ASR Subsystem

The MRCP ASR subsystem allows users to navigate through a menu of options by speaking instead of pressing keys on a touch-tone telephone. When a user calls local directory assistance, for example, ASR can prompt the user to say the city and state in which to locate the information, then connect the user to an appropriate operator.

To provision the MRCP ASR subsystem, define the following information:

- **MRCP ASR Providers**—Information about the vendor of your speech server, including the number of licenses and the grammar type.
- **MRCP ASR Servers**—Information about the ASR server's name, port location, and available languages.
- **MRCP ASR Groups**— Information about the MRCP ASR dialog control groups and associated locales, which enable Unified CCX applications to use speech recognition.

Provision MRCP ASR Providers

Use the MRCP ASR Provider Configuration web page to specify information about the vendor of your speech server.

Procedure

Step 1 From the Unified CCX Administration menu bar, choose **Subsystem** > **MRCP ASR** > **MRCP ASR Providers**.

The MRCP ASR Provider Configuration web page opens, displaying the list of currently configured MRCP providers, licenses, and the corresponding status.

Step 2 Click **Add New** icon that displays in the tool bar in the upper left corner of the window or the **Add New** button that is displayed at the bottom of the window.

The MRCP ASR Provider Configuration web page opens.

Step 3 Specify the following mandatory fields:

Field	Description
Provider Name	Enter the name of the MRCP ASR provider supported by Unified CCX.
Number of Provider Licenses	The number of ASR port licenses purchased from the ASR vendor.
Grammar Variant	Vendor-specific grammar setting. Valid options: • Nuance Open Speech Recognizer servers version 9.0 and above (OSR 3.1.x) • Nuance 11.x and below version ASR servers (Nuance) • IBM WVS ASR servers (2003 SISR)

Step 4 Click **Add** icon in the tool bar in the upper left corner of the window or the **Add** button that displays at the bottom of this window to apply changes.

Note
After you update MRCP ASR/TTS Providers, Servers, and Groups, the corresponding provider needs to be refreshed for changes to take effect. The Unified CCX Engine does not need to be restarted. However, during a Refresh, Unified CM Telephony triggers using affected groups will fall back to the dialog group that is configured and the MRCP Provider being refreshed will go NOT_CONFIGURED until the reload is complete.

Your changes appear in the MRCP ASR Providers List page. You are now ready to provision MRCP ASR Servers.

Note
If you delete an ASR/TTS provider and all of its associated servers and then create a new ASR/TTS provider, its status might become IN_SERVICE immediately, even before you create any servers for it. In this situation, click **Refresh** for that ASR/TTS provider, or click **Refresh** All. These actions change the status of the ASR/TTS provider to NOT_CONFIGURED.

Provision MRCP ASR Servers

Use the MRCP ASR Server Configuration web page to specify information about the speech server's name, port location, and available language.

Note You must have an MRCP ASR Provider defined before you can provision an MRCP ASR Server.

Procedure

Step 1 From the Unified CCX Administration menu bar, choose **Subsystem** > **MRCP ASR** > **MRCP ASR Servers**.

The MRCP ASR Server Configuration web page opens, displaying a list of previously configured servers, if applicable with the following information:

Column	Description

Computer Name	Hostname or IP address in which the ASR server software is installed. **Note** ASR server deployment over WAN is not supported in Unified CCX. The ASR server should be in the same LAN where Unified CCX is. You need to specify the ASR server hostname or IP address that is local with the Unified CCX node while installing the ASR server software in this field.
Provider	The MRCP ASR Provider to which this server is associated.
Port	The default TCP port number that is used to connect to a MRCP server. • OSR 3.1.x—4900 for MRCPv1 and 5060 for MRCPv2 • 2003 SISR—554 • Nuance—554
Status	Status or state of the subsystem.

Step 2 Click **Add New** icon that is displayed in the tool bar in the upper, left corner of the window or the **Add New** button that is displayed at the bottom of the window to provision a new MRCP ASR Server.

The MRCP ASR Server Configuration web page opens.

Step 3 Use this web page to specify the following fields.

Field	**Description**
Server Name	Hostname or IP address of the server where the MRCP ASR server software is installed.
Provider Name	Select the name of the MRCP ASR Provider to which this server is associated from this drop-down list.
Port Number	The default TCP port number that is used to connect to a MRCP server. Though the default value is shown as 4900. You need to provide any one of the following values in this field based on the TCP provider or grammar variant you have selected while configuring an MRCP ASR provider: • OSR 3.1.x—4900 for MRCPv1 and 5060 for MRCPv2 • 2003 SISR—554 • Nuance—554

Locales	Languages supported by the ASR Provider. Select a language (or multiple languages) from the drop-down list and click **Add Language**; the selected language appears with a check box in the Enabled Languages list. **Note** Use the check box to enable or disable a language.

Step 4 Click **Add** to apply changes.

Your changes appear in the MRCP ASR Server list web page. You are now ready to provision MRCP ASR Groups.

Provision MRCP ASR Dialog Groups

Use the MRCP Groups Configuration web page to specify information about MRCP ASR dialog control groups, which enable Unified CCX applications to use speech recognition.

Note You must have a MRCP ASR Provider defined before you can provision a MRCP ASR Group. Also, you should configure MRCP ASR Servers for the specific MRCP Provider before configuring the MRCP ASR Groups. This allows users to configure languages for the groups based on the languages supported by the configured servers.

Procedure

Step 1 From the Unified CCX Administration menu bar, choose **Subsystem** > **MRCP ASR** > **MRCP ASR Dialog Groups**.

The MRCP ASR Dialog Group Configuration web page opens to display a list of preconfigured entries, if applicable with the following information:

Field	Description
Group ID	Identifier for the group.
Description	Description of this dialog group.
Provider	Name of the MRCP ASR provider.
Channels	Maximum number of sessions.

This web page also displays the Number of Licensed IVR Channels.

Step 2 Click **Add New** icon that displays in the tool bar in the upper, left corner of the window or the **Add New** button that displays at the bottom of the window to provision a MRCP ASR Group.

The MRCP ASR Dialog Group Configuration web page opens.

Step 3 Use this web page to specify the following fields:

Field	Description
Group ID	Associated group ID.
Description	Description of this dialog group. **Tip** Include languages that will be used by this Group to the description. Doing so will provide insight into the languages this Group uses when you set up the dialog group in the Unified CM Telephony trigger configuration. This also ensures that the locales used by the application configured in the Unified CM Telephony trigger match the locales supported by the MRCP ASR dialog group being selected.
Number Of Provider Licenses	Display only.
Number Of Licensed IVR Ports	Display only.
Maximum Number Of sessions	Maximum number of sessions associated with this dialog group. **Note** You can assign any value for Maximum Number Of Channels, but restrictions are placed on the system when a call is made. This restriction is imposed by the number of licensed IVR ports on your system. **Caution** Under heavy load, calls that utilize a channel from an MRCP ASR Dialog Control Group, might have a reduced call completion rate as the MRCP channels used by calls can take some additional time to clean up all the sessions set up with MRCP resources. To address this situation, you can overprovision the value of this field by a factor of 1.2 or by an additional 20 percent. For example, if your application requires 100 MRCP ASR channels, modify the value in this field to be 120 MRCP ASR channels.
Provider Name	Select a MRCP Provider name from the drop-down list that contains a list of all previously defined provider names.
Enabled Languages	Select the languages that you wish to configure for this group from the list displayed. The displayed languages represent the locales configured for all MRCP ASR servers for the specified provider. If there are no MRCP ASR servers configured, no languages are displayed. In this case, you must update the group configuration once MRCP ASR servers have been configured for the specified provider.

Step 4 Click **Add** to apply changes.

Your changes appear in the MRCP ASR Groups list web page.

MRCP TTS Subsystem

The MRCP TTS subsystem converts plain text (UNICODE) into spoken words to provide a user with information, or prompt a user to respond to an action.

For example, a company might use TTS to read back a customer's name, address, and telephone number for verification before the company ships a requested product to the customer's location. Or a customer might dial into a pre-designated phone number, access a voice portal, and listen to the latest weather report or stock quotes. TTS can also convert email text to speech and play it back to the customer over telephone.

To provision the MRCP TTS subsystem, define the following information:

- **MRCP TTS Providers**—Information about the vendor of your TTS system.

Note If you delete an ASR/TTS provider and all of its associated servers and then create a new ASR/TTS provider, its status might become IN_SERVICE immediately, even before you create any servers for it. In this situation, click Refresh for that ASR/TTS provider, or click Refresh All. These actions change the status of the ASR/TTS provider to NOT_CONFIGURED.

- **MRCP TTS Servers**—Information about the TTS server's name, port location, and available languages.
- **MRCP TTS Default Genders**—Information about the default gender setting for the Locales specified during TTS Server provisioning.

Note You will need at least one MRCP TTS Provider for each vendor requiring TTS server installation.

Provision MRCP TTS Providers

Use the MRCP TTS Providers Configuration web page to specify information about the vendor of your TTS server.

Note After you update MRCP ASR/TTS Providers, Servers, and Groups, the corresponding provider needs to be refreshed for changes to take effect. The Unified CCX Engine does not need to be restarted. However, during a Refresh, Unified CM Telephony triggers using affected groups will fall back to the dialog group that is configured and the MRCP Provider being refreshed will go NOT_CONFIGURED until the reload is complete.

Procedure

Step 1 From the Unified CCX Administration menu bar, choose **Subsystems** > **MRCP TTS** > **MRCP TTS Provider**.

The MRCP TTS Provider Configuration web page opens. If providers are already configured, this page lists the provider name and corresponding status.

Step 2 Click **Add New** icon that displays in the tool bar in the upper, left corner of the window or the **Add New** button that displays at the bottom of the window.

Use this web page to specify the MRCP TTS Provider supported by Unified CCX.

The MRCP TTS Provider Configuration web page reopens. The Provider Name drop-down list displays the existing MRCP TTS Providers. Choose the MRCP TTS Provider supported by Unified CCX from this list.

Note
Support for High Availability and remote servers is available only in multiple-server deployments.

Step 3 Click **Add** to apply changes.

Your changes appear in the MRCP TTS Provider Configuration web page. You are now ready to provision MRCP TTS Servers.

Configure Default TTS Provider for Unified CCX System

Optionally, you can configure a default TTS provider. The Unified CCX Prompt Manager uses the default TTS provider for rendering TTS prompts if a TTS provider is not configured in the TTS Prompt. This usually happens in the case of VXML applications. For additional information on supported VXML tags for Unified CCX, see *Cisco Unified Contact Center Express Getting Started with Scripts* and for supported grammars see *Cisco Unified Contact Center Express Editor Step Reference Guide*.

To configure a default TTS provider, follow these steps.

Procedure

Step 1 Choose **System** > **System Parameters**.

Step 2 In the Default TTS Provider drop down list below Media Parameters section, select the provider you wish to be the system default. You must select only a preconfigured TTS provider as the Default TTS Provider.

Note
If you are deploying an VXML applications and the only TTS functionality you need is to play pre-recorded `.wav` files, select the **Cisco LiteSSMLProcessor** option as the Default TTS Provider. This option allows you to run SSML that has .wav file references in them.

Step 3 Click **Update**.

Provision MRCP TTS Servers

Use the MRCP TTS Servers Configuration web page to configure the TTS server's name, port location, and available languages.

You need at least one MRCP TTS Server associated with each configured provider.

Note You must have an MRCP TTS Provider defined before you can provision an MRCP TTS Server.

Procedure

Step 1 From the Unified CCX Administration menu bar, choose **Subsystems** > **MRCP TTS** > **MRCP TTS Server**.

The MRCP TTS Server Configuration web page opens, displaying a list of previously configured servers, if applicable, with the following information:

Column	Description
Computer Name	Hostname or IP address of the server in which the TTS server software is installed. **Note** TTS server deployment over WAN is not supported in Unified CCX. In other words, the TTS server should be in the same LAN where Unified CCX is. Thus, you need to specify the TTS server hostname or IP address that is local with the Unified CCX node while installing the TTS server software in this field.
Port	TCP port number used to connect to an MRCP server. Following are the different TCP Provider names: • MRCP Server • Nuance Vocalizer • Scansoft Realspeak
Provider	The MRCP TTS Provider to which this server is associated.
Status	Status or state of the subsystem.

Step 2 Click **Add MRCP TTS Server** icon that displays in the tool bar in the upper, left corner of the window or the **Add New** button that displays at the bottom of the window to provision a new MRCP ASR Server.

The MRCP TTS Server Configuration web page opens.

Step 3 Specify the following fields:

Field	Description
Server Name	Hostname or IP address of the server the MRCP TTS server software is installed.
Provider Name	Select the name of the MRCP TTS Provider to which this server is associated from this drop-down list.

Port Number	The default TCP port number used to connect to a MRCP TTS server. The port number is automatically displayed based on the provider or grammar variant that you have selected while configuring an MRCP TTS provider. Following are the different TCP Provider names along with their port numbers: • MRCP Server—554 • Nuance Vocalizer—4900 for MRCPv1 and 5060 for MRCPv2 • Scansoft Realspeak—4900
Locales	Languages supported by the TTS Provider. Select a language (or multiple languages) from the drop-down list and click **Add Language**; the selected language appears in the Enabled Language list. **Note** Use the check box to disable/enable a language.

Step 4 Click **Add** to apply changes.

Your changes appear in the MRCP TTS Server Configuration web page. You are now ready to provision MRCP TTS Default Genders.

Note
Whenever a new language is added for an MRCP Server—and if this is the first instance of this language being added for the corresponding MRCP Provider—then the default gender for that locale and for the specified provider is set to Neutral. You should check the MRCP Locales page to review the default genders that are set automatically per locale per provider. Default genders are used when a prompt for a specific locale is used without specifying any gender.

Provision MRCP TTS Default Genders

Use the MRCP TTS Default Genders Configuration web page to configure the default gender settings per Locale per Provider. TTS uses default genders when a prompt for a specific locale is used without specifying the gender.

Procedure

Step 1 From the Unified CCX Administration menu bar, choose **Subsystems** > **MRCP TTS** > **MRCP TTS Default Genders**.

The MRCP TTS Default Gender Configuration web page opens, displaying the default genders currently configured for each locale for every MRCP TTS Provider that is currently configured.

Step 2 Optionally, change the default gender setting for each locale for each provider.

Note

The Locale radio button has the Male, Female, or Neutral options. By default, the “Default Gender” is set to “Neutral” unless configured explicitly.

Step 3 Click **Update** to apply changes.

The system updates the default gender setting for each Locale per Provider.