NetVanta Unified Communications Technical Note

Integrating Cisco CallManager using SIP

1 Integration Overview

This integration note applies to Cisco CallManager Version 5.1 and later.

The UC server integrates with Cisco CallManager using SIP Trunks.

1.1 Minimum Software Versions

- Cisco CallManager 5.1 or higher (Cisco CallManager 6.0 has been interoperability tested)
- NetVanta Unified Communications Server Software Revision: Software version 4.3 and higher

1.2 Supported Features

The following are the features supported with this Integration:

- Call forwarding to personal greeting
  - Busy
  - Ring-no-answer
  - Unconditional
- Direct call – Manage messages (prompt for mailbox password)
- Transfer Capabilities
  - Blind transfers
  - Supervised/Assisted transfers
- Message Waiting Lights
- Caller ID (if supplied by PBX)
- Notification Services
  - Active Message Delivery
  - Pager Notification
  - E-mail Notification
- Outbound notification (database integration with outbound notification)

NOTE: This integration method does not support faxing. See *Integrating the Cisco CallManager Fax* technical note, available online at *http://kb.adtran.com*, for information about integrating fax with Cisco CallManager.

2 Configuring the Cisco CallManager

In order to simplify these steps, **X → Y → Z** are used to describe the menus in the Cisco CallManager Web UI, which means click menu **X**, choice **Y**, sub-choice **Z**.

2.1 Checking the Licensing

1. On the administration UI, go to **System → Licensing → License Unit Report**.
2. Confirm that the license file is uploaded by confirming that there are a number of **Units Authorized**.

Phone License Feature

License Server	Units Authorized	Units Used	Units Remaining
ccm-sdk	50	8	42
Total Units for Feature	$\boxed{\textbf{50}}$	**8**	**42**

CCM Node License Feature

License Server	Units Authorized	Units Used	Units Remaining
ccm-sdk	1	1	0
Total Units for Feature	$\boxed{\textbf{1}}$	**1**	**0**

2.2 Configuring the SIP Trunk Security Profile

In order for the SIP trunk to communicate, you require a properly configured SIP Trunk Security Profile.

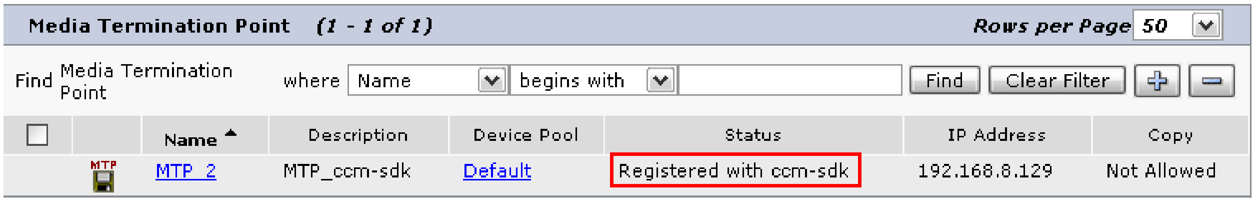
1. Go to **System → Security Profile → SIP Trunk Security Profile.**
2. Click **Find.**
3. Select a SIP Trunk Security Profile.
4. Select **UDP** from the **Outgoing Transport Type** list.
5. Ensure that the **Incoming Port** is 5060.
6. Ensure that these options are selected:
    a. Accept Presence Subscription
    b. Accept Out-of-Dialog REFER
    c. Accept Unsolicited Notification
    d. Accept Replaces Header
7. Click **Save**.

The SIP Trunk Security Profile will be referred to as the "Configured SIP Trunk Security Profile".

2.3 Configuring the Media Termination Point

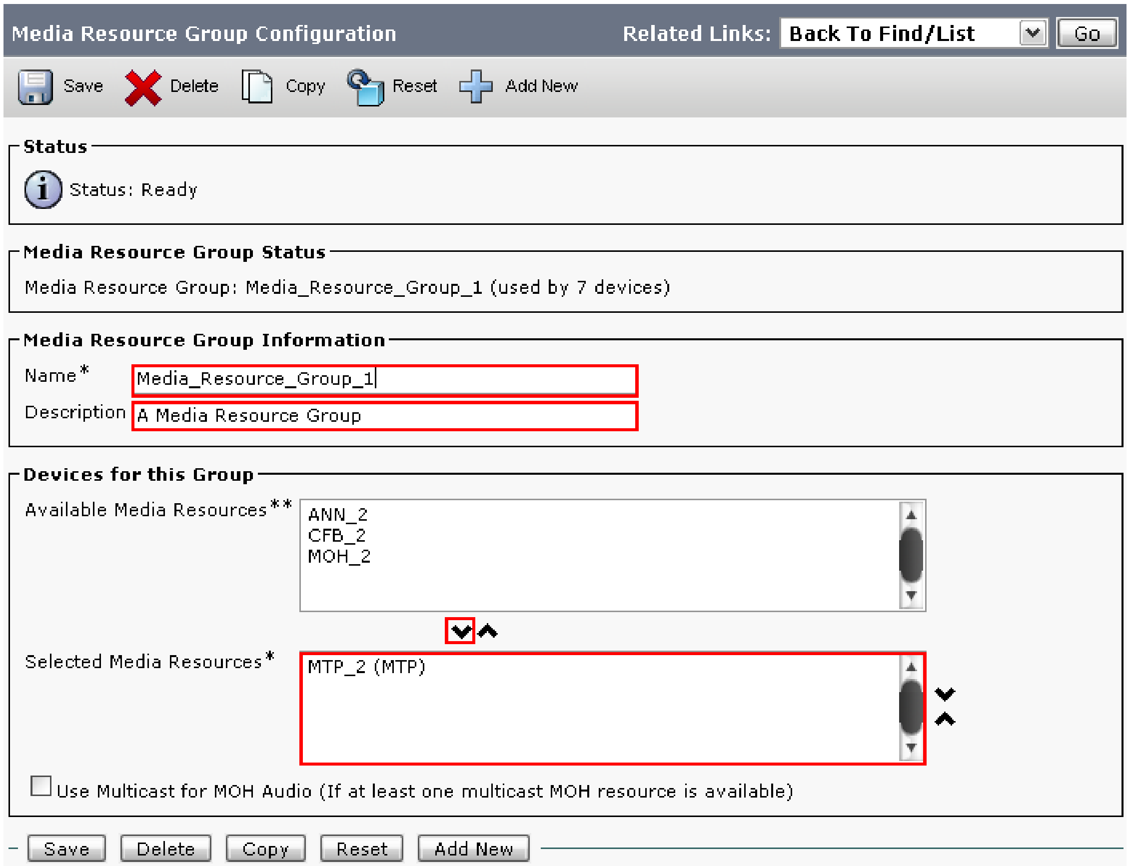
In order to send a Session Description with the INVITE SIP packet, Cisco CallManager requires a Media Termination Point.

1. Go to **Media Resources → Media Termination Point**.
2. Click **Find**.
3. Ensure there is an entry in the list and confirm that the status reads “Registered with (Cisco CM server name)”.

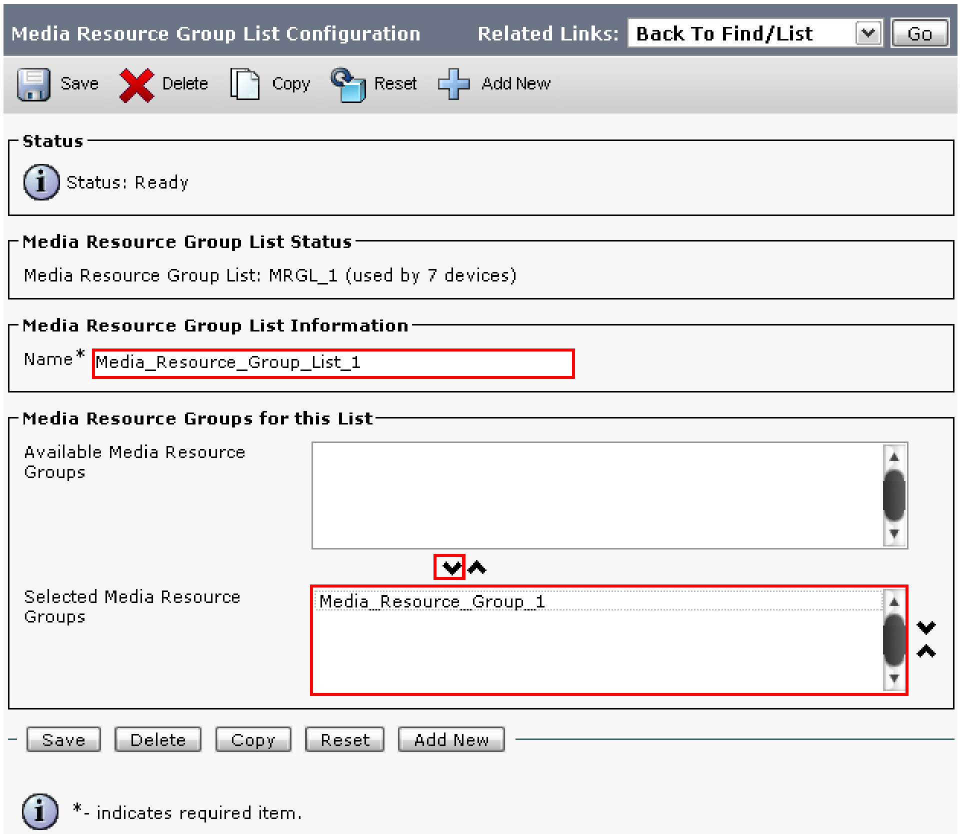


Note: If it does not read as such, the service is either not activated or stopped and it needs to be activated or started. To do this, follow the steps at the end of this section in 2.3.1 - Activating Cisco IP Voice Media Streaming App.

4. Go to **Media Resources → Media Resource Group**.
5. Click **Find**.
6. If there are no entries, or you require a new Media Resource Group, follow “a” through “d” below, otherwise, open an existing group and ensure it has a Media Termination Point in the Selected Media Resources.
   a. Click **Add New.**
   b. Enter a name such as “Media_Resource_Group_1” and a description.
   c. In the Devices for this Group section, select a Media Termination Point such as **MTP_2** in the Available Media Resources list and click the down arrow between the two lists to move it to the Selected Media Resources list.
   d. Click **Save**.

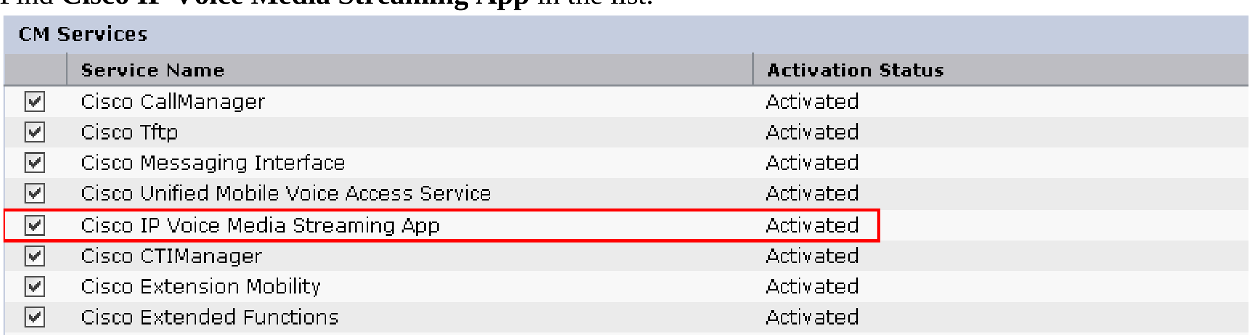


7. Go to **Media Resources → Media Resource Group List.**
8. Click **Find**.
9. If there are no entries, or you require a new Media Resource Group List, follow "a" through "d" below, otherwise open an existing list and ensure it has the previous Media Resource Group in the Selected Media Resource Groups.
    a. Click **Add New**.
    b. Enter a name such as "Media_Resource_Group_List_1".
    c. In the Devices for this Group section, select the previous Media Resource Group in the Available Media Resource Groups list and click the down arrow between the two lists to move the group to the Selected Media Resource Groups list.
    d. Click **Save**.



2.3.1 Activating Cisco IP Voice Media Streaming App

1. Select “Cisco Unified Serviceability” from the list in the upper right corner of the Cisco web configuration interface and click **Go**.
2. Go to **Tools → Service Activation.**
3. Find **Cisco IP Voice Media Streaming App** in the list.



4. Select the check box next to it.
5. Click the **Save** button at the top or bottom of the page.
6. Go to **Tools → Control Center - Feature Services.**
7. Find **Cisco IP Voice Media Streaming App** in the list.

CM Services

	Service Name	Status*	Activation Status	Start Time	Up Time
○	Cisco CallManager	Started	Activated	Thu Feb 21 16:07:42 2008	19 days 17:45:48
○	Cisco Tftp	Started	Activated	Thu Feb 21 16:07:51 2008	19 days 17:45:39
○	Cisco Messaging Interface	Not Running	Activated		
○	Cisco Unified Mobile Voice Access Service	Started	Activated	Mon Feb 25 18:19:14 2008	15 days 15:34:16
○	Cisco IP Voice Media Streaming App	Started	Activated	Mon Feb 25 18:05:55 2008	15 days 15:47:35
○	Cisco CTIManager	Started	Activated	Mon Feb 25 18:19:14 2008	15 days 15:34:16
○	Cisco Extension Mobility	Started	Activated	Mon Feb 25 18:19:14 2008	15 days 15:34:16
○	Cisco Dialed Number Analyzer	Started	Activated	Mon Feb 25 18:19:24	15 days 15:34:06

8. If the status is not **Started** then select the service and press the **Start** button at the top or bottom of the page.
9. Select "Cisco Unified CM Administration" from the list in the upper right corner of the Cisco web configuration interface and click **Go**.

2.4 Creating New SIP Trunks

In order to communicate with the UC server, Cisco CallManager requires a SIP trunk.

1. Go to **Device → Trunk**.
2. Click **Add New.**
3. Select **SIP Trunk** from the **Trunk Type** list.
4. Ensure that **SIP** is selected from the **Device Protocol** list.
5. Click **Next.**
6. Enter a **Device Name** such as "UCServer_SIP_Trunk".
7. Select the appropriate device pool from the **Device Pool** list.
8. Select a Media Resource Group List that contains a Media Termination Point from the **Media Resource Group List**.
9. Select a location from the **Location** list.
10. Ensure **Media Termination Point Required** is selected.
11. In the **Inbound Calls** section, ensure **Redirecting Diversion Header Delivery – Inbound** is selected.
12. In the **Outbound Calls** section
    a. Select **Originator** from the **Calling Party Selection** list.
    b. Ensure **Redirecting Diversion Header Delivery – Outbound** is **checked.**

13. In the **SIP Information** section

    a. Enter the UC server's IP Address or FQDN for the **Destination Address.**

    b. Enter 5080 for the **Destination Port**.

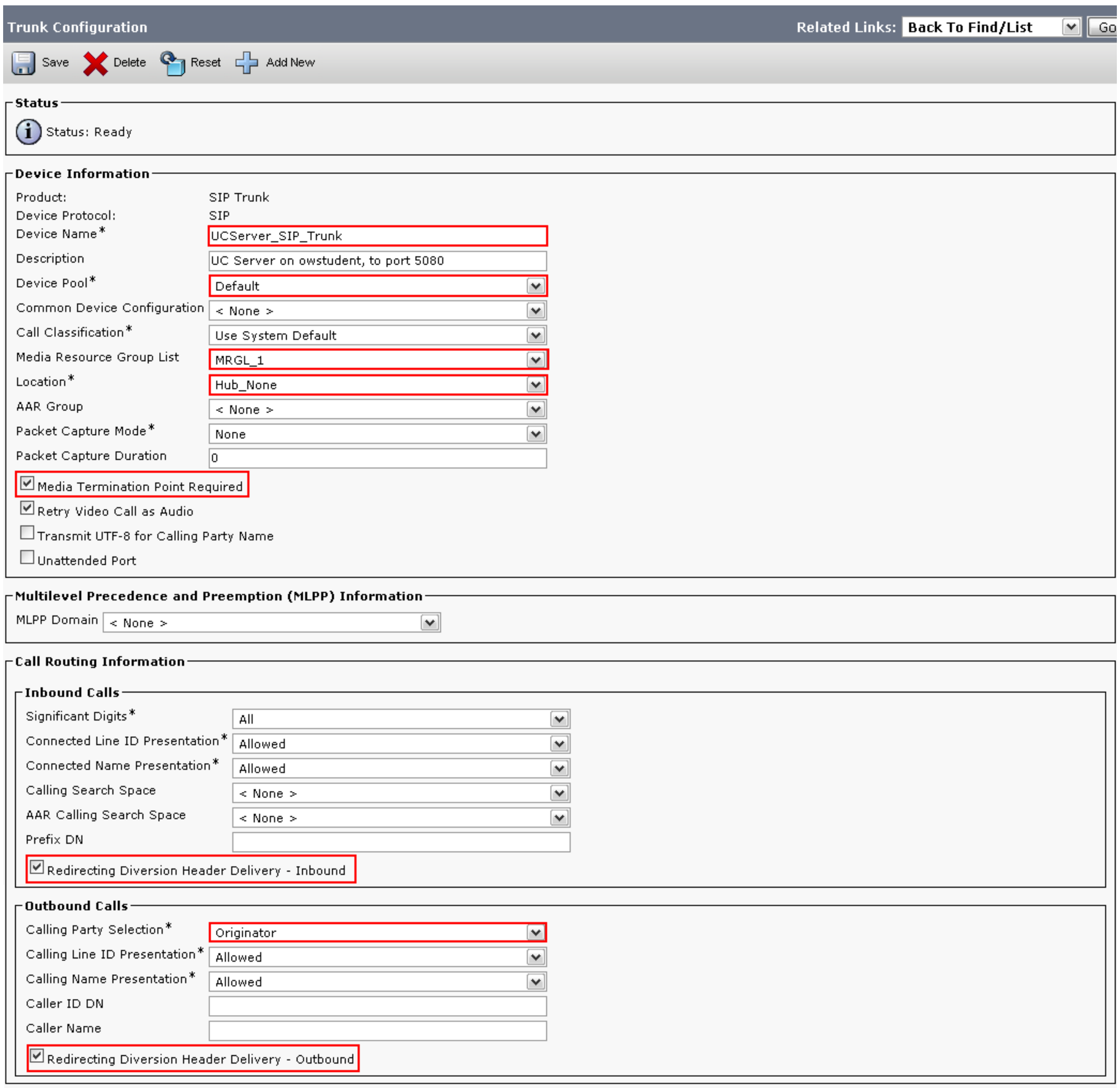
    c. Select **711uLaw** from the **Preferred Originating Codec** list.

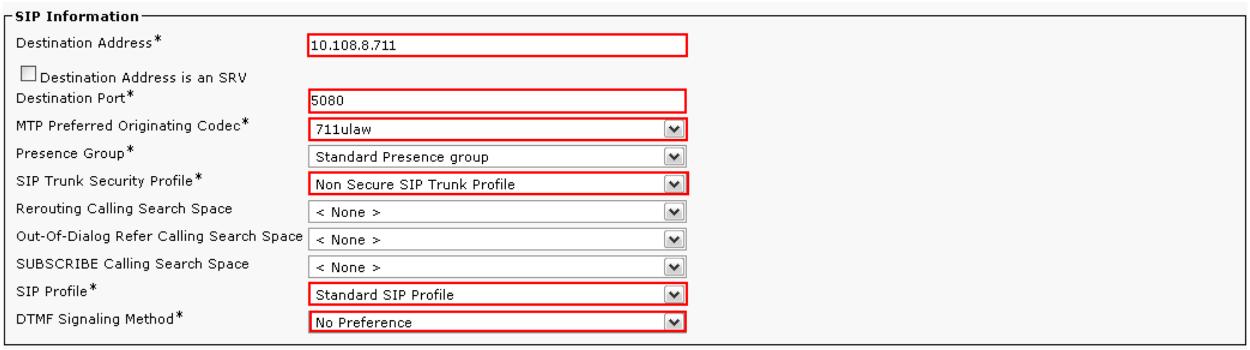
    d. Select the Configured SIP Trunk Security Profile from the **SIP Trunk Security Profile** list.

    e. Select a SIP profile from the **SIP Profile** list.

    f. Select **No Preference** from the **DTMF Signaling Method** list.

14. Click **Save.**



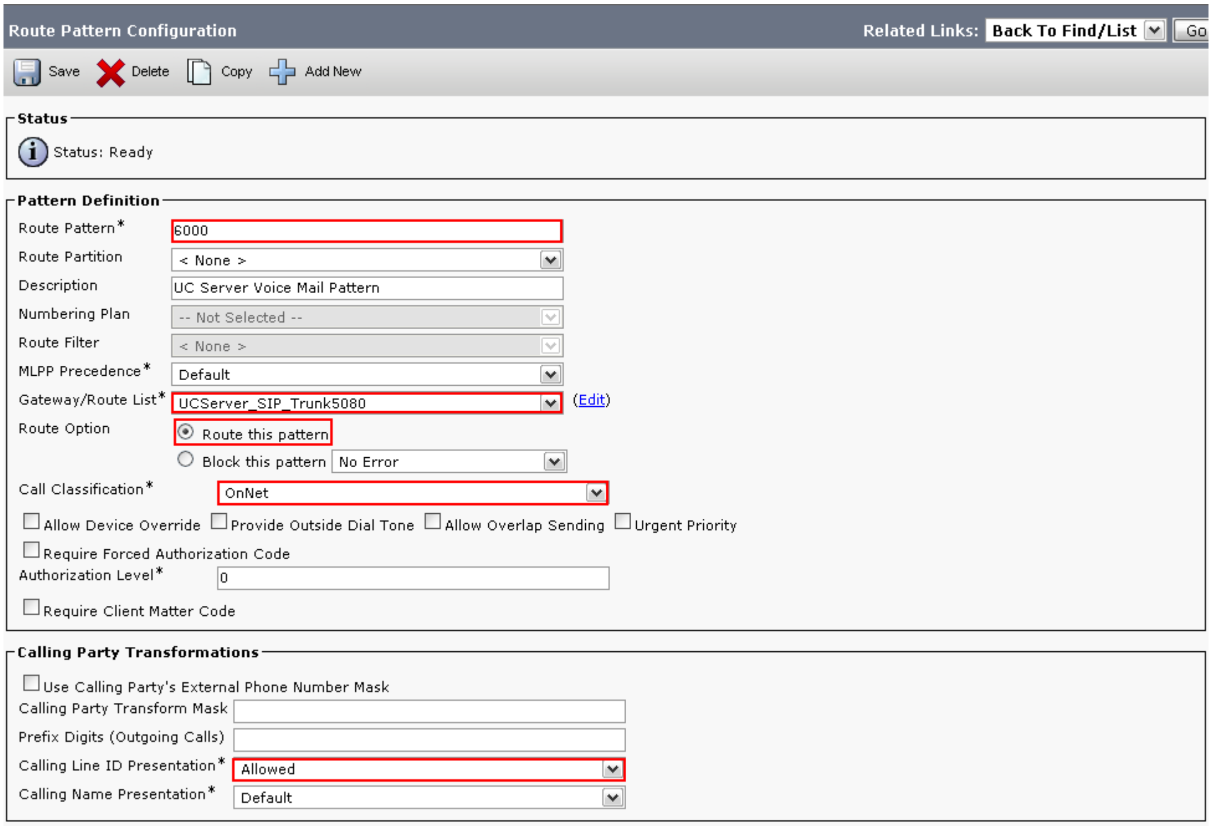


2.5 Creating New Route Patterns

Creating a route pattern tells Cisco CallManager which voicemail calls need to be sent to the UC server.

1. Go to **Call Routing → route/hunt → route pattern**.
2. Click **Add New**.
3. Set the **Route Pattern** to the number used to contact the UC server, such as 6000.
4. Select **UCServer_SIP_Trunk** from the **Gateway or Route List** menu.
5. Select **Route this pattern** for the **Route Option.**
6. Select **OnNet** from the **Call classification** menu.
7. In the **Calling party Transformations** section, Select **Allowed** from the **Calling Line ID Presentation** list.

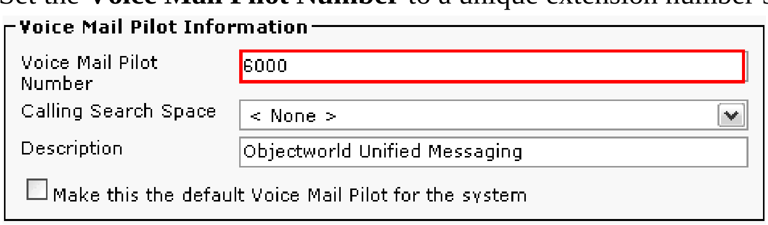
8. Click **Save**.



2.6 Configuring the Voicemail Profile

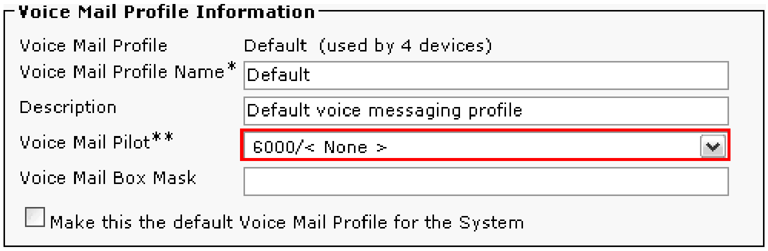
In order to configure a telephone's Message/Voicemail button, a Voicemail profile is required. The number chosen corresponds to the dial plan created earlier to route these calls to voicemail to the UC server.

1. Go to **Voice Mail → Voice Mail Pilot**.
2. Click **Add New.**
3. Set the **Voice Mail Pilot Number** to a unique extension number such as 6000.



4. Click **Save.**
5. Go to **Voice Mail → Voice Mail Profile**.
6. Click **Find** then select an existing Voice Mail Profile such as **Default** or click **Add New** to create a new one.
   a. If it is a new profile, enter a **Voice Mail Profile Name** such as UC_Server_Voicemail.

7. Select **6000/< None >** from the **Voice Mail Pilot** list.



   Note: If a number other than 6000 is used, the number that is used replaces 6000 in the selection.

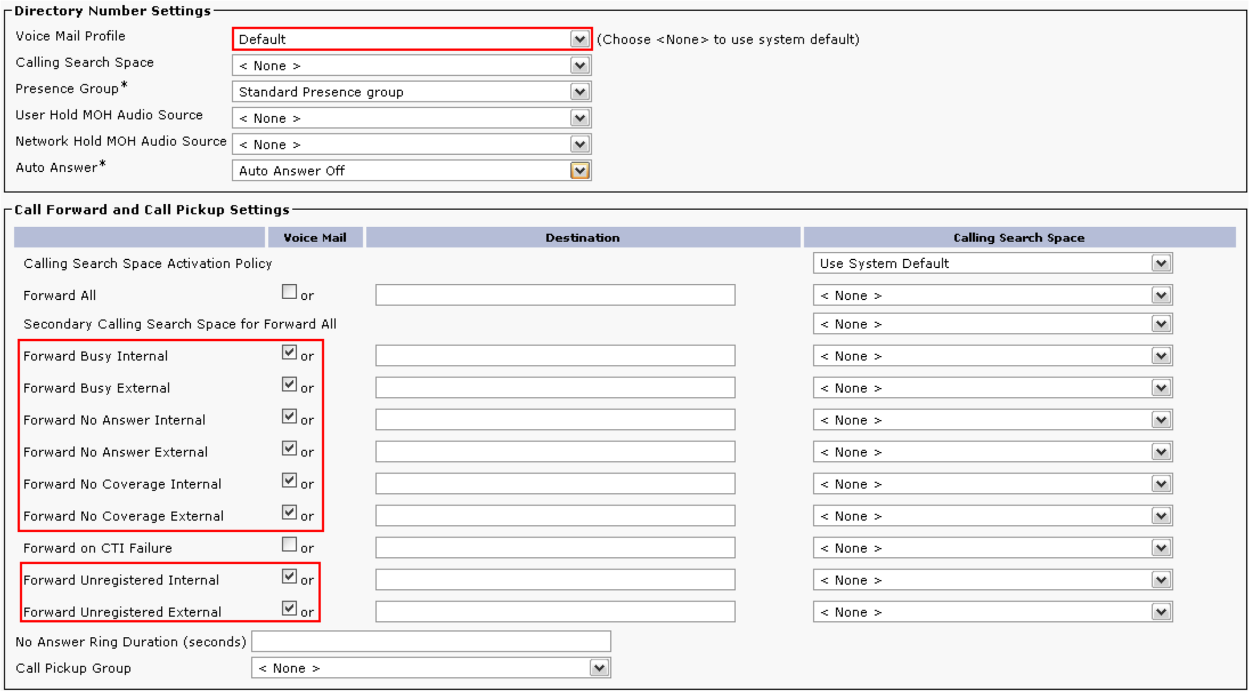
8. Click **Save.**

2.7 Checking the Phone Configuration

Each phone has a line associated with it. For calls to be forwarded to voicemail under specific circumstances, such as when nobody answers, and for the message button on phones to work, these settings must be configured.

1. Go to **Device → Phone** and click the phone that you want to configure.
2. Click the line for which you need to configure voice mail settings.
3. Ensure that the **Voice Mail Profile** is set to the Voice Mail Profile configured in the previous step.
4. Ensure that these options are selected.
   a. Forward Busy Internal
   b. Forward Busy External
   c. Forward No Answer Internal
   d. Forward No Answer External
   e. Forward No Coverage Internal
   f. Forward No Coverage External
   g. Forward Unregistered Internal
   h. Forward Unregistered External

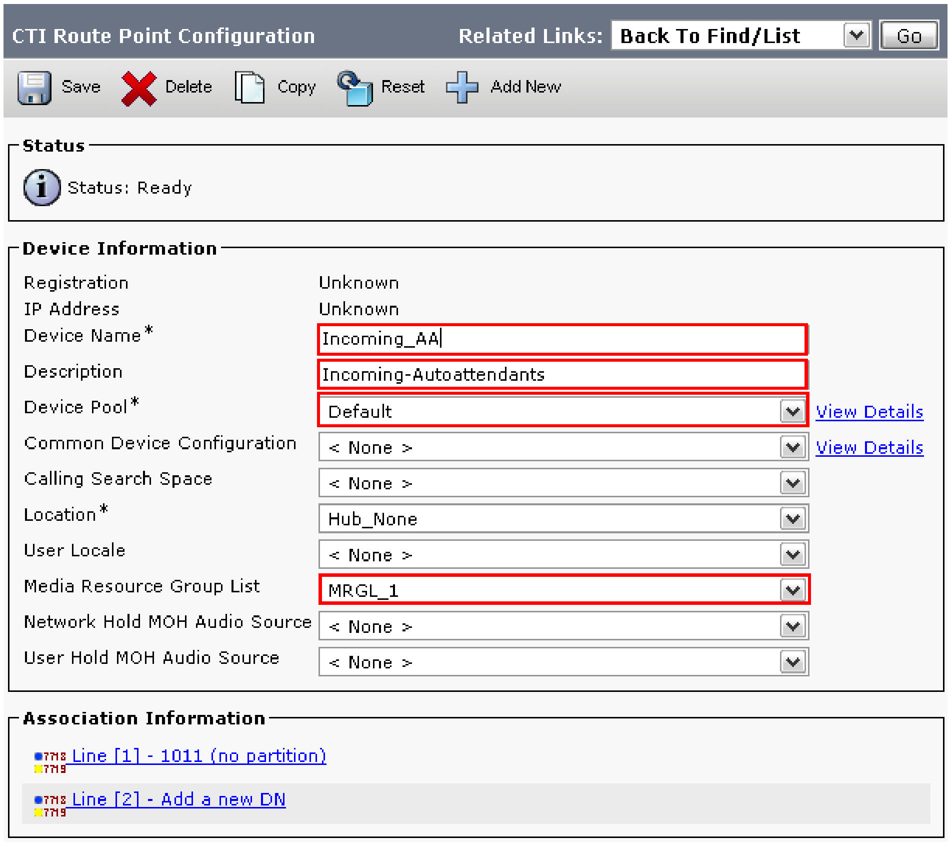
5. Click **Save**.



2.8 Creating the CTI Route Point for the Incoming Auto Attendant

In order for Cisco CallManager to route inbound calls from the PSTN to an Auto Attendant, it requires a CTI Route Point to route the calls to. The CTI Route Point is usually associated with an attendant service to help callers find who they want to talk with, or a self-service function.

1. Go to **Device → CTI Route Point**.
2. Click **Add New**.
3. Enter a name such as “Incoming_AA”.
   Note: The name must be a unique identifier, consisting of 1 to 15 alphanumeric, dot, dash, or underscore characters.
4. Enter a description such as “Incoming Autoattendant”.
5. Select **Default** from the **Device Pool** list.
6. Select **Media_Resource_Group_List_1**from the **Media Resource Group List**.
7. Click **Save.**



8. Click **Line [1] - Add a new DN**.

9. Enter a unique **Directory Number** such as 1011 that is either configured as an attendant service or currently unused on the UC server.

10. Enter a **Description** such as “Auto Attendant”.

11. Enter an **ASCII Alerting Name** such as “Auto Attendant”.

12. Select **UC_Server_Voicemail** from the **Voice Mail Profile** list.

13. Ensure that these options are selected.
    Under **Call Forward and Call Pickup Settings:**

    a. Forward Busy Internal

    b. Forward Busy External

    c. Forward No Answer Internal

    d. Forward No Answer External

    e. Forward No Coverage Internal

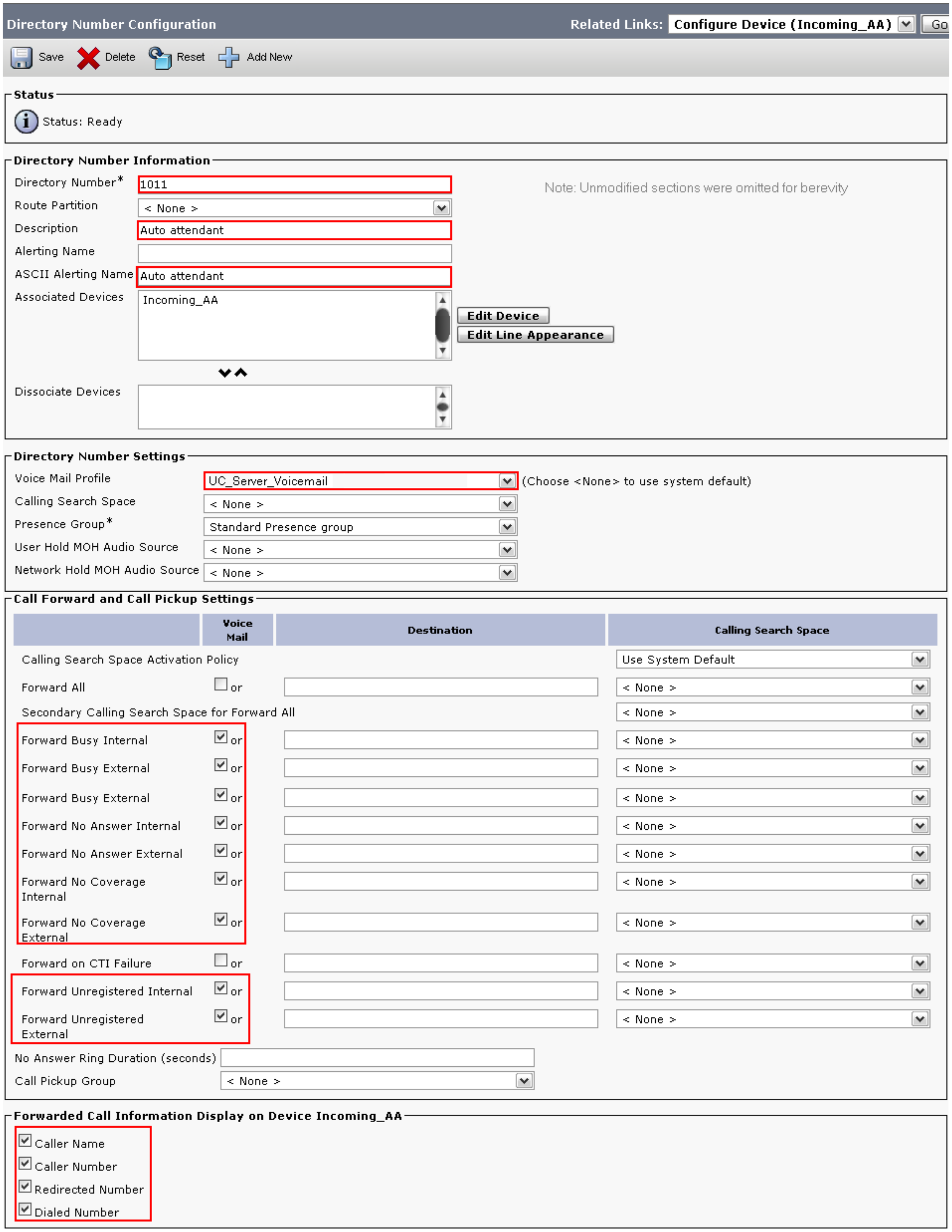
    f. Forward No Coverage External

    g. Forward Unregistered Internal

    h. Forward Unregistered External
    Under **Forwarded Call Information Display on Device Incoming_AA:**

i. Caller Name
j. Caller Number
k. Redirected Number
l. Dialed Number

14. Click **Save**.

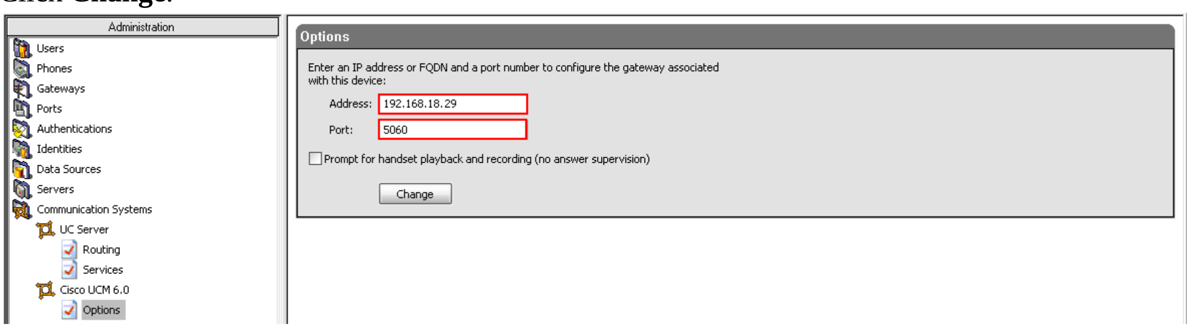

Directory Number Configuration
Related Links: Configure Device (Incoming_AA) Go
Save Delete Reset Add New
Status
Status: Ready
Directory Number Information
Directory Number* 1011
Route Partition < None >
Description Auto attendant
Alerting Name
ASCII Alerting Name Auto attendant
Associated Devices Incoming_AA
Edit Device
Edit Line Appearance
Dissociate Devices
Note: Unmodified sections were omitted for berevity
Directory Number Settings
Voice Mail Profile UC_Server_Voicemail (Choose <None> to use system default)
Calling Search Space < None >
Presence Group* Standard Presence group
User Hold MOH Audio Source < None >
Network Hold MOH Audio Source < None >
Call Forward and Call Pickup Settings
Voice Mail
Destination
Calling Search Space
Calling Search Space Activation Policy Use System Default
Forward All or < None >
Secondary Calling Search Space for Forward All < None >
Forward Busy Internal or < None >
Forward Busy External or < None >
Forward Busy External or < None >
Forward No Answer Internal or < None >
Forward No Answer External or < None >
Forward No Coverage Internal or < None >
Forward No Coverage External or < None >
Forward on CTI Failure or < None >
Forward Unregistered Internal or < None >
Forward Unregistered External or < None >
No Answer Ring Duration (seconds)
Call Pickup Group < None >
Forwarded Call Information Display on Device Incoming_AA
Caller Name
Caller Number
Redirected Number
Dialed Number

3 Configuring the UC Server

3.1 Creating the Communications System

In order for the UC server to work with Cisco CallManager, it requires a configured communications system. The type of system that is used is called Generic PBX (via Gateway).

1. In the Administration navigation pane, select **Communications Systems**.
2. Right-click in the content pane and select **New…**
3. Click **Next**, select **Generic PBX (via Gateway)** from the list, and then click **Next**.
4. Enter a name for the communications system such as "Cisco CM 6.0" and an answering group number that is the same as the Voice Mail Pilot on the Cisco CallManager, such as 6000. Click **Next.**
5. Select the network interface and click **Next**.
6. Ensure that the "Automatically Configure Windows Firewall for the UC server" option is selected and click **Next**.
7. Click **Submit**, **Next**, and **Finish**.
8. In the Administration navigation pane, under **Communications Systems**, under **Cisco CM 6.0**, select **Options**.
9. Enter the IP address or FQDN of the Cisco CallManager and the port.
10. Click **Change**.



3.2 Configuring the Port for the Communications System

The method and network port used to communicate with the Cisco CallManager is determined by the settings in the port created in the process of creating the communications system. A few of the default setting must be changed to work with Cisco CallManager.

1. In the Administration navigation pane, select **Ports**.
2. Right-click the Generic PBX (via Gateway) entry and select **Open**.
3. Select **UDP** from the **Protocol** list.
   NOTE: If some features need to be disabled, such as outbound dialing, ensure that those features are not selected. It is also possible to change the name of the port if it is required.

4. Click **OK**.

3.3 Creating the Attendant Identity

The CTI Route point on the Cisco CallManager is pointing to an attendant identity on the UC server to answer calls from the PSTN. If the attendant identity does not already exist, it needs to be created.

1. In the Administration navigation pane, select **Identities**.
2. Right-click in the content pane and select **New Identity…** and Click **Next**.
3. Select "Cisco CM 6.0" from the communication system list.
4. Select "Admin" from the user profile list.
5. Select the **Attendant Service** class and click **Next**.
6. Enter a **Name** such as "Cisco CM 6.0 Attendant"
7. Enter the address configured earlier in the CTI Route point (1011 if you used the suggested address).
8. Select a service for the attendant to run when it is called, click **Next** and then **Finish**.
   Note: If you have not created a custom service, "Default Trunk Service" is the best option.

3.4 Creating the Users

In order for each Cisco phone to have its own mailbox and answering behavior, each phone must have a user and Identity associated with it.

For each user:

1. In the Administration navigation pane, select **Users**.
2. Right-click in the content pane and select **New…**
3. Click **Next**. Select **Local User** and click **Next**.
4. Enter the user's **First**, **Last,** and **Display** names.
5. Enter the user's Directory Number as the Identity #.
6. Select "Cisco CM 6.0" from the list next to the Identity #. Click **Next**.
7. Enter the user's Password and PIN, and then click **Next**.
8. Click **Next**. Select Personal Assistant or Personal Business Assistant and the operator choice depending on your needs.
9. Click **Next**, **Submit**, and **Finish**.