OpenText RightFax Installation Guide Supplement

Integrating with Cisco Voice and Unified Communications Products

(Includes Cisco Unified Communications Manager 8.5.10000-23)

January 5, 2011

This supplementary guide to OpenText RightFax describes deployment models and procedures for using OpenText RightFax in a Fax-over-IP (FoIP) deployment with Cisco Voice and Unified Communications products - Cisco Unified Communications Manager (CUCM) version 8.x and Cisco Gateways.

Contents

Abstract

This supplement to the OpenText RightFax product documentation describes deployment models and procedures required to integrate OpenText RightFax and Cisco Voice and Unified Communications products. This documentation has been updated to address integration between OpenText RightFax version 9.4 Feature Pack 1 Service Release 2 and Cisco Unified Communications Manager version 8.5

Overview

As companies migrate to Cisco IP-based telephony solutions, fax communication over IP networks requires consideration. OpenText RightFax connects to telephony environments using Cisco Voice and Unified Communications Products through Plain Old Telephone Service (POTS) technology and using Fax-over-IP (FoIP) technologies. In a FoIP solution, OpenText RightFax can connect to Cisco Unified Communications Manager, Cisco IOS Voice Gateways, and Cisco Universal Gateways over IP networks. This integration to send and receive fax documents utilizes either Session Initiation Protocol (SIP) or H.323 and T.38 real-time Fax-over-IP.

Common document delivery solutions using OpenText RightFax and Cisco Voice and Unified Communications products consist of the following components:

- OpenText RightFax version 9.4 FP1 SR2 or later, containing either Dialogic® Brooktrout® SR140 software-only FoIP, or TR1034-series IP-enabled fax boards.
- Cisco Unified Communications Manager (CUCM)
- Cisco IOS Voice Gateways

About OpenText RightFax

OpenText RightFax utilizes all three International Telecom Union (ITU) fax transmission protocols:

- **T.30** – Send faxes over the public switched telephone network (PSTN), also known as the Plain Old Telephone System (POTS).
- **T.37** – Send faxes using store-and-forward over the Internet. Uses email protocols like MIME or SMTP to translate faxes into emails.
- **T.38** – Real-time faxing over the internet, delivered like a fax call. Encapsulates the T.30 protocol into a T.38 data stream.

Cisco Requirements for OpenText RightFax Interoperability

OpenText RightFax supports Cisco IOS Gateways, Cisco Universal Gateways, and Cisco Unified Communications Manager as follows:

- Cisco Unified Communications Manager (CUCM)
    - For H.323: Release 4.2.3 or later (within the 4.2.x product line)
    - For SIP: Release 5.0.4(a) or later (within the 5.0.x product line)
    - For SIP and H.323: OpenText RightFax v9.4 supports v7

- Cisco IOS Gateway Series (those capable of supporting T.38)
    - SIP, H.323 and MGCP protocols
    - Cisco IOS version 12.3T and later versions

OpenText RightFax Installation and Deployment

OpenText RightFax software may be installed on any supported system, and may be deployed in a variety of configurations. For more information, consult the OpenText RightFax product documentation.

Each OpenText RightFax main server or Remote DocTransport Server instance may contain a maximum of 120 channels, in any combination of physical fax boards and boardless channels. The main server and all Remote DocTransports support a combined maximum of 1024 channels.

OpenText RightFax channels are enabled by purchasing Document Delivery Channels (DDCs). Additionally, you must obtain physical fax boards or Dialogic SR140 licenses containing the desired number of channels for use in conjunction with the fax server's DDCs.

Dialogic® Brooktrout® SR140 FoIP Software

The Dialogic SR140 host-based FoIP solution may be used with OpenText RightFax 9.3 Feature Pack 1 and later versions. All media processing and call control functions are performed using host system CPU and memory, without the use of fax hardware. SR140 works with both SIP and H.323 protocols.

Dialogic® Brooktrout® Fax Boards

Each physical fax board may be operated in either TDM mode or IP mode, but not both. A single fax server or Remote DocTransport server may contain a maximum of four boards operating in different modes.

When operating in IP mode, the fax board may send and receive faxes to and from multiple T.38-enabled Cisco routers. The board firmware will be licensed for the ordered number of concurrent fax transmissions. Dialogic Brooktrout TR1034-series IP-enabled fax boards work with both SIP and H.323 protocols.

Configuring OpenText RightFax

This guide assumes the reader has requisite knowledge and resources available to install and configure the necessary OpenText RightFax application and telephony configurations required for production operation, including configuration of Dialogic Brooktrout fax boards and SR140 Fax-over-IP.

Information on configuring OpenText RightFax and Dialogic Brooktrout products, consult the OpenText RightFax product documentation, and Dialogic Brooktrout documentation. If you are having difficulties, please contact your appropriate OpenText Technical Support resource for further assistance.

Guidelines for OpenText RightFax

1. T.38 Fax-over-IP (FoIP) capability is supported on OpenText RightFax version 9.3 and higher.
2. Information on configuring Dialogic Brooktrout fax boards and SR140 for communication with telephony equipment, please consult the Dialogic *Windows End User Guide*, available online at http://www.dialogic.com/webhelp/Brooktrout/SDK63/WindowsEndUserGuide.pdf.
3. Cisco IOS Voice Gateways require T.38 protocol support.

Guidelines for Dialogic Brooktrout SR140 FoIP Software

1. SR140 support for G.711 and voice features requires OpenText RightFax version 9.4 FP1 SR2.

Guidelines for Dialogic Brooktrout TR1034 IP-Enabled Fax Boards

1. Dialogic Brooktrout TR1034 board models ending in -1N are T.38 compatible (e.g. TR1034+P24-T1-1N). Models ending in -0N may be upgraded to support T.38.
2. T.38 Fax-over-IP uses the Ethernet network interface of the host server for call setup (SIP), and the Ethernet network interface of the fax board for T.38 fax transmission.
3. The TR1034 Ethernet interface requires static IP address settings.
4. The TR1034 Ethernet interface and Cisco Gateway must be on the same network subnet.
5. OpenText RightFax voice features (e.g. Human Answered Fax, Docs-On-Demand) are not supported with the TR1034 configured for T.38 FOIP.

Interoperability Notes

Levels of T.38 fax relay support in Cisco Unified Communications Manager Software Release versions and OpenText RightFax versions are listed in Table 1.

Table 1: T.38 Fax Relay Support in Cisco Unified Communications Manager

T.38 Protocol Support	CUCM Software Release	OpenText RightFax Version
H.323 Only	4.1(1), 4.2(3), 5.0(1), 6.0(1), and higher	9.4 FP1 SR2 and later **
H.323 & MGCP Only	4.2(3), 6.0(1), and higher	9.4 FP1 SR2 and later **
H.323 & SIP Only	5.0(1), 6.0(1), and higher	9.4 FP1 SR2 and later **
H.323, SIP & MGCP	6.0(1) and higher	9.4 FP1 SR2 and later **

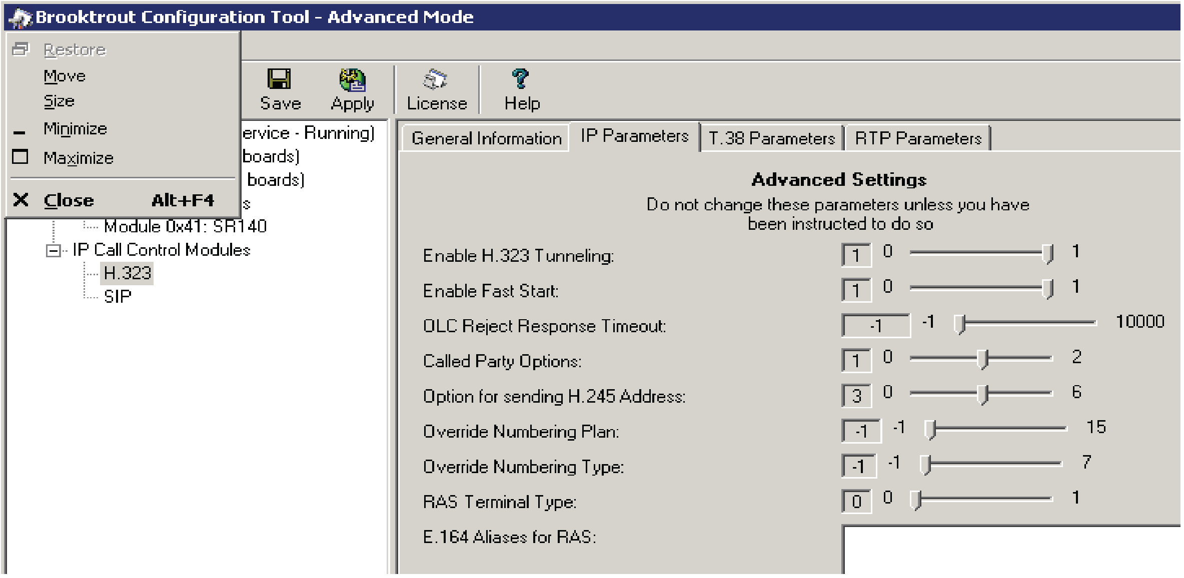
*** RightFax versions prior to have known interoperability issues with H.323 in CUCM environments – When CUCM sends a second reinvite, Dialogic SR140 software does not respond correctly to the second invite request.*

Integration with versions of Cisco Unified Communications Manager that do not support *H.323 fast start* and *H.245 tunneling*, require changes in the Brooktrout Configuration Tool.

Changes required in Brooktrout Configuration Tool (also see Figure 5 below):

- **Enable Fast Start (**`h323_Faststart) = 0`
- **Enable H.323 Tunneling (**`h323_h245Tunneling) = 0`
- **Option for sending H.245 Address (**`h323_H245Stage) = 3`

Figure 5: H.323 Fast Start and H245 Tunneling Parameters in the Brooktrout Configuration Tool



Environments with OpenText RightFax version 9.3 and Cisco Unified Communications Manager versions 6.1 or 7.0 may experience problems with SIP interoperability. Use one of the following options to avoid issues:

- **Recommended:** Use OpenText RightFax version 9.4 Feature Pack 1 Service Release 3, and Cisco Unified Communications Manager version 8.5.

Deployment Models

Most integrations of OpenText RightFax in a Cisco Voice and Unified Communications infrastructure fall under one of the following categories:

- TDM Connection
- Cisco Voice Gateway FoIP Integration
- Cisco Unified Communications Manager FoIP Integration

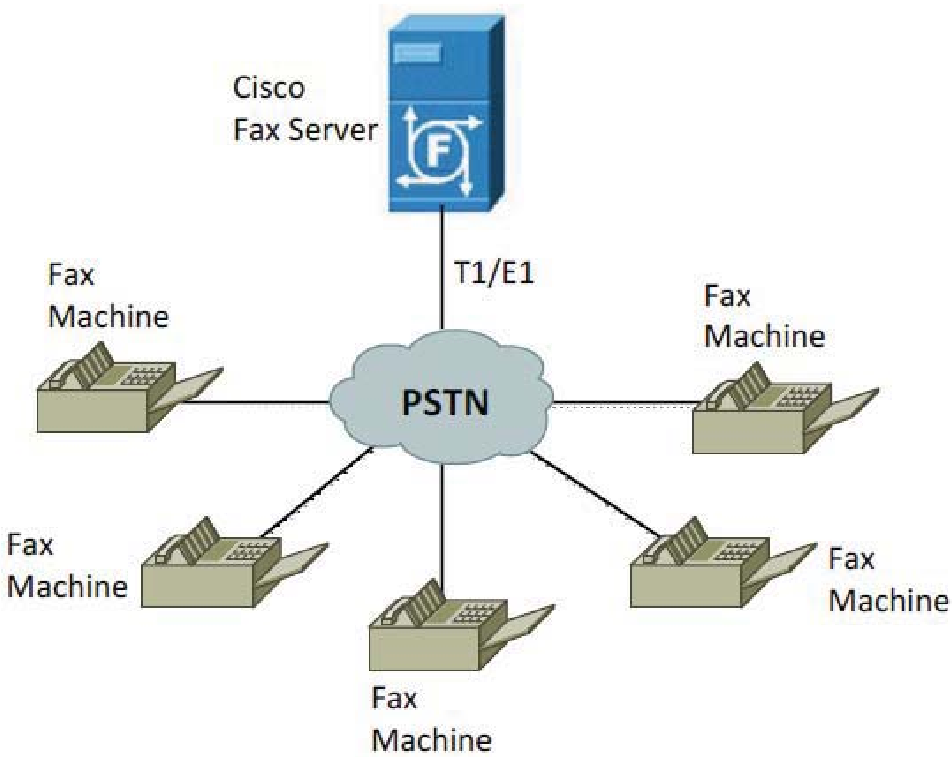
TDM deployments were used before FoIP became a practical alternative. In this model, OpenText RightFax is connected to Cisco Communications equipment by direct T1/E1 circuits. The majority of the OpenText RightFax server deployments now use IP-based connections. FoIP integrations enable fax communication over the IP Telephony infrastructure.

TDM Connection

Customers with existing investment in Brooktrout TR1034-series IP-enabled fax boards may choose to implement a TDM deployment, and migrate to a FoIP deployment in the future. TDM connections required dedicated circuits to the PSTN, either a full T1/E1 or dedicated fax channels on a T1/E1 circuit.

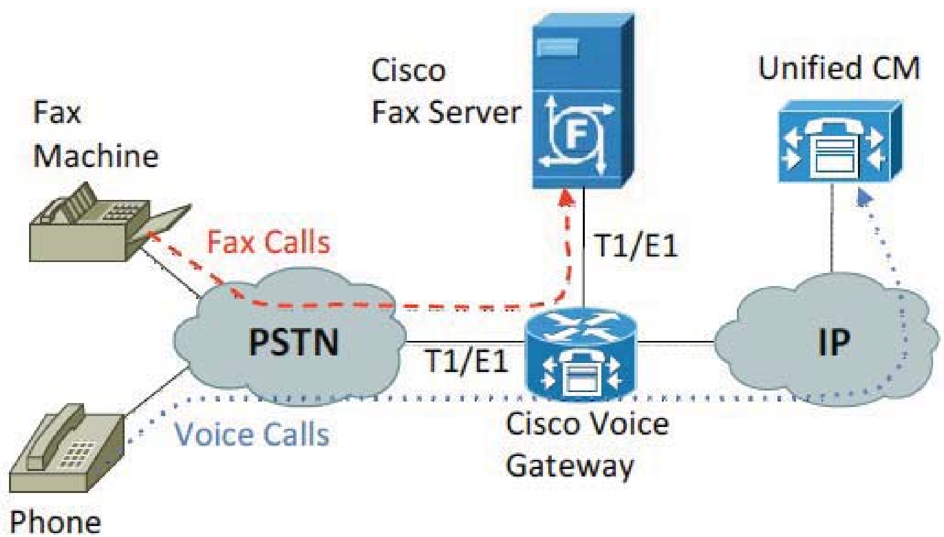
In Figure 1 below, the OpenText RightFax server is connected directly to T1/E1 TDM circuits using Dialogic Brooktrout fax boards installed in the server or Remote DocTransport servers.

Figure 1: OpenText RightFax Connected Directly to PSTN by T1/E1 Circuit


Cisco
Fax Server
T1/E1
PSTN
Fax
Machine
Fax
Machine
Fax
Machine
Fax
Machine
Fax
Machine
Fax
Machine

In Figure 2 below, calls are routed between the RightFax and the PSTN through telephony ports on a Cisco voice gateway. Fax calls are cross-connected between two ports on the gateway. This is commonly referred to as a "hairpin call".

Figure 2: Hairpin Calling between OpenText RightFax, Cisco IOS Voice Gateway, and the PSTN



In this scenario, voice and fax calls use the same physical PSTN T1 connection terminated on the Cisco IOS voice gateway. Another T1 circuit on a separate gateway voice port connects directly to the OpenText RightFax. The Cisco voice gateway distinguishes between voice and fax calls inbound from the PSTN by evaluating the DNIS number and routes the voice and fax calls appropriately.

In Figure 2 above, voice calls received on the PSTN T1 circuit are converted to IP and routed to the Cisco Unified Communications Manager. Fax calls are cross-connected to the T1 voice port connected to OpenText RightFax.

When using a hairpin scenario, make sure that the connection is "DSP-less". The DSP will drop out of the call path and OpenText RightFax connects directly to the PSTN through the Cisco voice gateway. Otherwise, the DSP continues to process and make slight changes to the TDM stream.

To ensure the DSP drops out of the hairpin call, follow these guidelines:

- Enable `local-bypass` under the `voice-card` submenu of the Cisco IOS voice gateway.

- If the T1/E1 voice ports reside in separate module slots on the voice gateway make sure the gateway has a TDM backplane, and use the `network-clock-participate` command to ensure both are part of the backplane clocking scheme.

- DSPs involved in the hairpin call must be of the same type.

Hairpin calling is set up using an inbound and outbound POTS dial peer on the Cisco voice gateway. For more information on administering dial peers please see the following link on www.cisco.com:

> http://www.cisco.com/en/US/docs/ios/voice/dialpeer/configuration/guide/vd_dp_feat_cfg_ps6350_TSD_Products_Configuration_Guide_Chapter.html

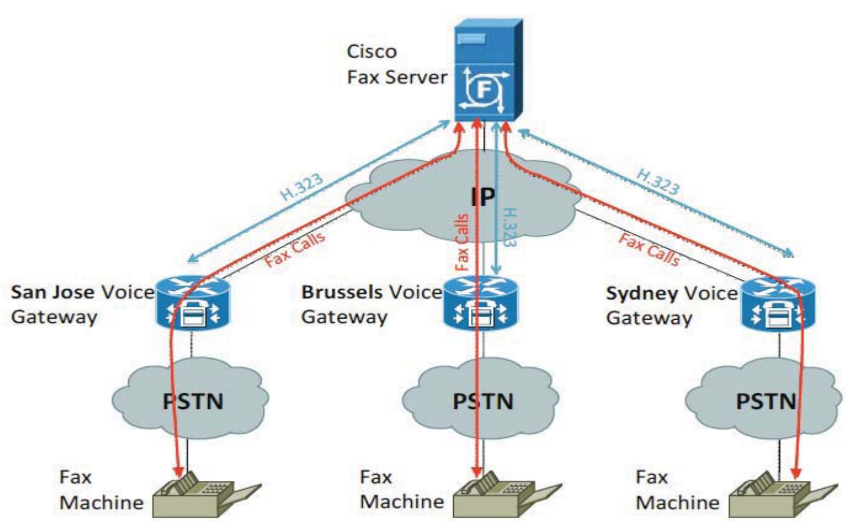
Cisco Voice Gateways

OpenText RightFax servers with IP-enabled fax boards or Dialogic SR140 FoIP software communicate with Cisco voice gateways using the IP protocol. The Cisco voice gateway must support ITU-T standard T.38 fax relay. Cisco IOS voice gateways such as the 2800 and 3800 series are most commonly used. ***Note: Dialogic Brooktrout TR1034 board models ending in -1N are T.38 compatible (e.g. TR1034+P24-T1-1N). Models ending in -0N may be upgraded for T.38 support.***

Call setup between OpenText RightFax servers and Cisco voice gateways occurs using either H.323 or Session Initiation Protocol (SIP). H.323 is older and widely supported; however, SIP is rapidly gaining adoption.

In the simplest voice gateway integration, OpenText RightFax communicates with a single voice gateway. Most deployment models integrate OpenText RightFax with multiple voice gateways to route calls to a gateway local to the fax destination or to achieve a level of fault tolerance. Figure 3 below depicts a multiple voice gateway deployment using H.323. SIP is deployed in the same way.

Figure 3: Voice Gateway Deployment Model for the Open Text RightFax



In the above scenario, configure Dialing Rules in OpenText RightFax to route outbound fax calls through multiple Cisco voice gateways. For more information, see *Configuring Fax over IP Failover* in the RightFax Administrator's Guide included with your OpenText RightFax product documentation.

Configure *voice dial-peers* on your Cisco IOS voice gateways to route inbound fax calls to the appropriate OpenText RightFax. A sample H.323 dial-peer configuration for Cisco IOS voice gateways is shown in Example 1 below.

Example 1: Sample H.323 Dial-Peer Configuration for Communicating with OpenText RightFax

```
!
dial-peer voice 6 voip
 incoming called-number .
 destination-pattern 6000
 codec g711ulaw
 session target ipv4:<IP ADDRESS OF RIGHTFAX>
 fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback none
!
dial-peer voice 7 pots
 destination-pattern 7000
 port 0/0/0
!
```

Calls in a Cisco IOS voice gateway require two call legs. Example 1 above contains two configurations, a VoIP dial peer for the RightFax, and a POTS dial peer for the PSTN connection. H.323 and SIP settings on the Cisco IOS voice gateway are configured on the VoIP dial peer.

To change the configuration from H.323 to SIP, add the `session protocol sipv2` command to the voip dial peer.

The voip dial peer in Example 1 is used for inbound and outbound fax calls. The `destination pattern 6000` command routes calls inbound from the PSTN to OpenText RightFax at IP address 192.168.10.2, shown in the `session target ipv4` parameter. The command `incoming called-number` ensures outbound calls from RightFax to the PSTN match this dial peer and inherit its properties.

Two commands are required for interoperability with OpenText RightFax:

- `codec g711ulaw` – Explicitly specifies G.711 codec. By default, Cisco IOS voice gateways use the G.729 codec. OpenText RightFax supports only G.711, with a-law or u-law. **Note:** You may also configure a voice class codec that includes G.711.
- `t38 ls-redundancy 0 hs-redundancy 0 fallback none` – Explicitly specifies use of T.38 fax relay. Cisco gateways support a number of fax transport protocols; however, OpenText RightFax supports T.38 only. **Note:** This option may also be configured globally under the voice service voip section of the IOS voice gateway configuration.

Cisco Unified Communications Manager

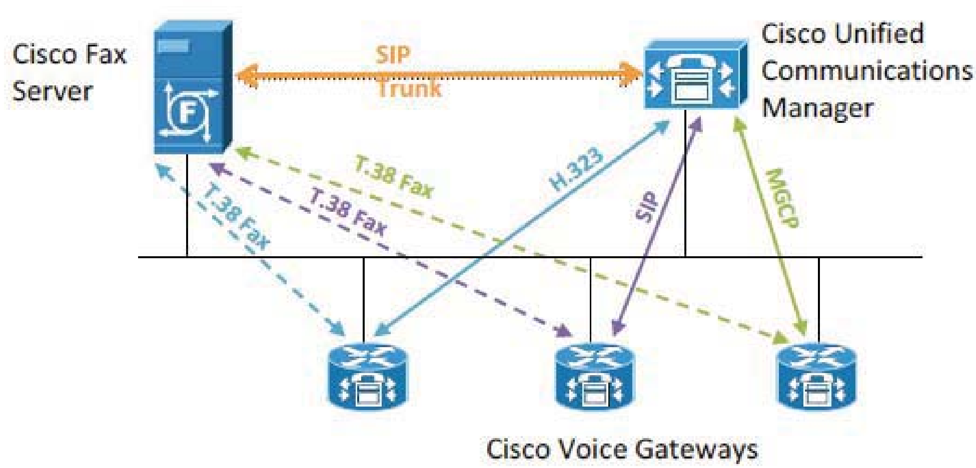
Integrating OpenText RightFax with Cisco Unified Communications Manager (CUCM) provides greater flexibility, redundancy, and easier administration. CUCM supports both H.323 and SIP protocols required by OpenText RightFax.

Key benefits of implementing Cisco Unified Communications Manager:

- CUCM manages call routing for the telephony network.
    - All outbound calls are routed to CUCM, which then determines the most appropriate route for the call. It is not necessary to create Dialing Rules on OpenText RightFax for each Cisco IOS Voice Gateway, and leverages the VoIP dial plan already in place.
    - Inbound calls from the PSTN are routed by CUCM to the RightFax.
- CUCM provides OpenText RightFax access to MGCP-controlled voice gateways by translating SIP and H.323 calls to MGCP as needed.

In H.323 integrations, OpenText RightFax is added to CUCM as an H.323 Gateway. In SIP scenarios, Cisco Unified Communications Manager is configured for a SIP trunk connection to OpenText RightFax. Once H.323 or SIP connection is established between Cisco Unified Communications Manager and OpenText RightFax, then OpenText RightFax has access to all H.323, SIP, and MGCP voice gateways connected to Cisco Unified Communications Manager. Figure 4 shows OpenText RightFax integration with the Cisco Unified Communications Manager.

Figure 4: Cisco Unified Communications Manager Deployment with OpenText RightFax



Configuring Cisco Voice Gateways in CUCM Integration Scenarios

In integrations using Cisco Unified Communications Manager, the Cisco IOS Voice Gateways must be configured to point to the IP address of the CUCM rather than OpenText RightFax. This destination is configured by modifying the `session target ipv4` parameter of the dial-peer configuration, as shown in Example 2 below.

Example 2: Sample Cisco Voice Gateway H.323 Dial-Peer Configuration for Communicating with OpenText RightFax in Cisco Unified Communications Manager Integrations

```
!
dial-peer voice 6 voip
 incoming called-number .
 destination-pattern 6000
 codec g711ulaw
 session target ipv4:<IP ADDRESS OF CUCM SERVER>
 fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback none
!
dial-peer voice 7 pots
 destination-pattern 7000
 port 0/0/0
!
```

Appendix A: Practical Scenarios– OpenText RightFax & Cisco Unified Communications Manager

This appendix describes integration scenarios including overview and detailed configuration information. Each scenario has been deployed and tested to verify functionality

The format of each scenario uses the following outline:

1. Network Diagram
2. Equipment Description and Network Identification Info
3. Dialing Plan Example
4. OpenText RightFax Configuration Notes
5. Cisco Voice Gateway Configuration
6. Cisco Unified Communications Manager Configuration

All scenarios use the following product versions

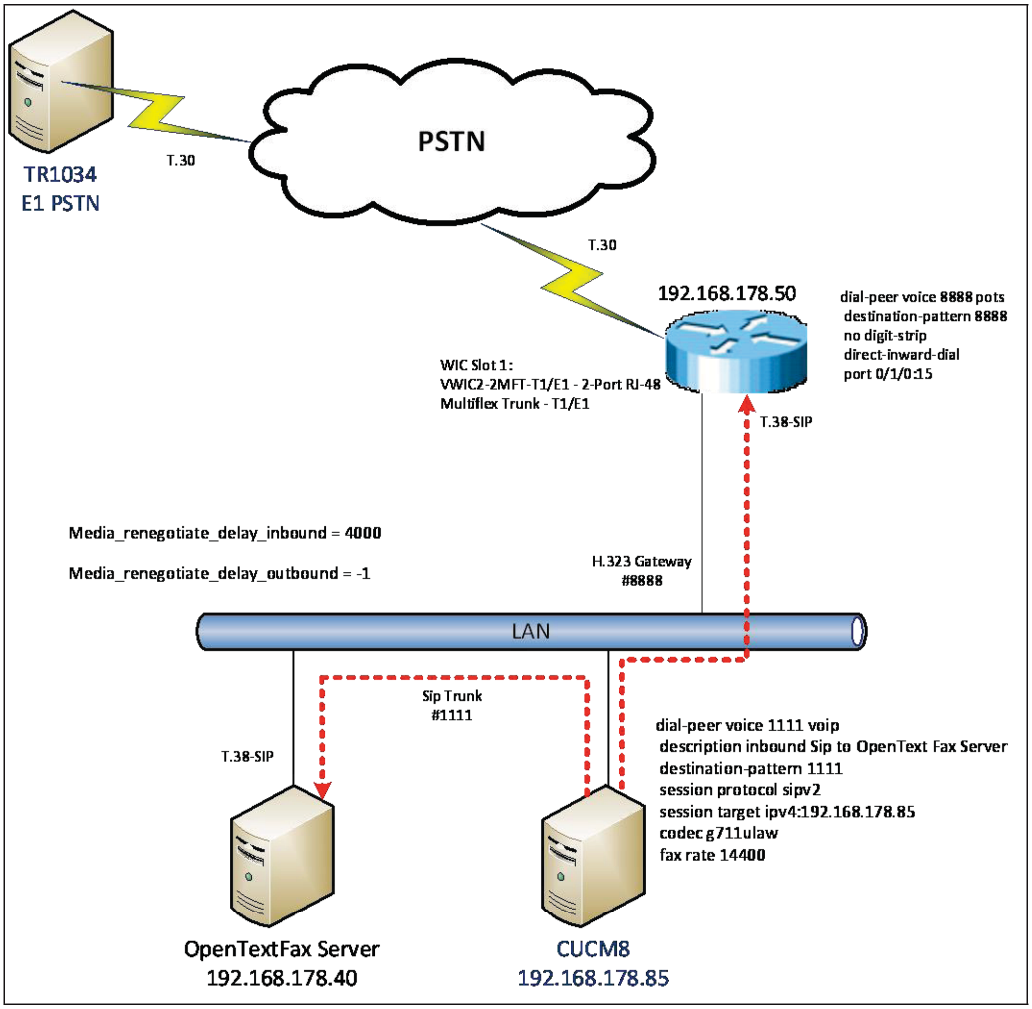
- OpenText RightFax version 9.4 Feature Pack 1 Service Release 2
- Cisco Unified Communications Manager 8.5.1000-23
- Cisco 2800 Integrated Service Router

Outline of Scenarios

1. Scenario 1: SIP-to-SIP Configuration
    a. RightFax <-SIP-> CUCM 8.5 <-SIP-> Gateway
2. Scenario 2: H.323-to-H.323 Configuration
    a. RightFax <-H.323-> CUCM 8.5 <-H.323-> Gateway
3. Scenario 3: SIP-to-MGCP Configuration
    a. RightFax <-SIP-> CUCM 8.5 <-MGCP-> Gateway
4. Scenario 4: H.323-to-MGCP Configuration
    a. RightFax <-H.323-> CUCM 8.5 <-MGCP-> Gateway

Scenario 1: SIP-to-SIP Configuration

Network System Configuration – Sip / Sip Configuration



Network Addresses

Device #	Device Make, Model, and Description	Device IP Address
1	OpenText RightFax	192.168.178.40
2	CUCM 8.5.10000-23	192.168.178.85
3	Cisco 2800 Integrated Service Router	192.168.178.50

Dial Plan Overview

To call OpenText RightFax (SR140) from a POTS phone, dial 1111. The call flow and protocol path behaves as follows:

- POTS (dial 1111) —E1—>
- Cisco Gateway (dial 1111@192.168.178.85) —SIP—>
- CUCM85.10000-23 dial 1111@192.168.178.40)—SIP—>
- OpenText RightFax.

To call the POTS lines of the Gateway, dial 8888@192.168.178.83. The call flow and protocol path behaves as follows:

- OpenText RightFax(8888@192.168.178.85) —SIP—>
- CUCM85.10000-23 dial 8888@192.168.178.50)—SIP—>
- Cisco Gateway (dial 8888)—E1—>
- POTS

OpenText RightFax SR140 Setup Notes

In this scenario, Dialogic SR140 is required non-default values. For RightFax version 9.4 FP1 SR2 (Dialogic SDK 6.3.0 and later), the following parameters must be set under T.38 Parameters:

- Media Renegotiate Delay Inbound, msec = 4000
    - Callctrl.cfg value = Media_renegotiate_delay_inbound
- Media Renegotiate Delay Inbound, msec = -1
    - Callctrl.cfg value = Media_renegotiate_delay_outbound

Dialogic® Brooktrout® TR1034 Fax PSTN Setup Notes

For the sample test configuration, the TR1034 was configured using the default values, consult the Dialogic® Brooktrout® Fax Products Installation and Configuration Guide for details.

Cisco 2800 Gateway Setup Notes

For the sample test configuration, the Cisco 2800 Gateway was configured the Cisco IOS command-line interface. The specific items configured include:

- Enable T.38 support
- Configure line card interface
- Configure IP Protocol
- Configure Dial-Peers – POTS
- Configure Dial-Peers – VoIP

Enable T.38 support

The following lines allow SIP calls and T.38 fax calls

```
voice service voip
 fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none
 SIP
```

Configure line card interface

```
controller E1 0/0/0
 clock source internal
 pri-group timeslots 1-8,16
```

Configure Dial-Peers – POTS

The following allows the phone “8888” to be dialed out though the POTS lines:

```
dial-peer voice 8888 pots
 destination-pattern 8888
 no digit-strip
 direct-inward-dial
 port 0/0/0:15

interface Serial0/0/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5
 isdn protocol-emulate network
 isdn incoming-voice voice
 no cdp enable
```

Configure Dial Peers - VoIP

The following allows the number “1111” to be dialed out through SIP to CUCM:

```
dial-peer voice 1111 voip
 description inbound Fax traffic from Sip to OpenText RightFax
 destination-pattern 1111
 session protocol sipv2
 session target ipv4:192.168.178.85
 codec g711ulaw
 fax rate 14400
```

Note: The `session target ipv4` *parameter contains the IP address for the CUCM.*

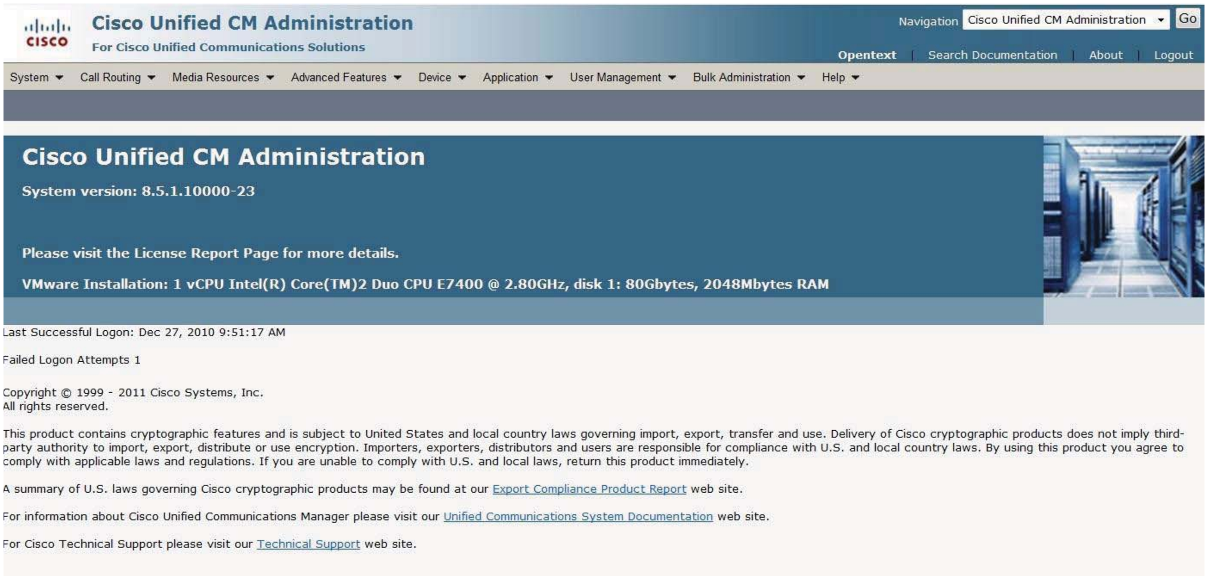
CUCM 8.5 Setup Notes – SIP / SIP Configuration

The following areas of CUCM 8.0(x) are modified in this scenario:

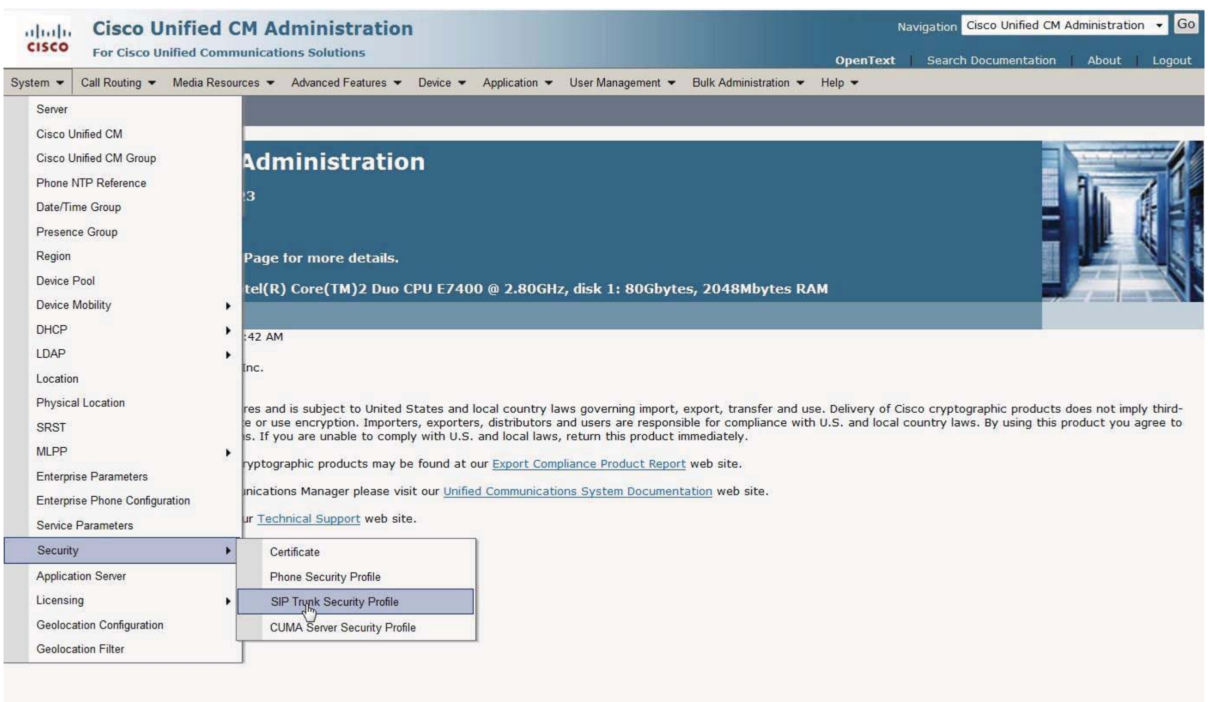
- Configure SIP Trunk Security Profile
- Configure Sip Trunk from CUCM to OpenText RightFax
- Configure Sip Trunk from CUCM to Gateway
- Configure Call Routing
- IOS overview

Configure SIP Trunk Security Profile

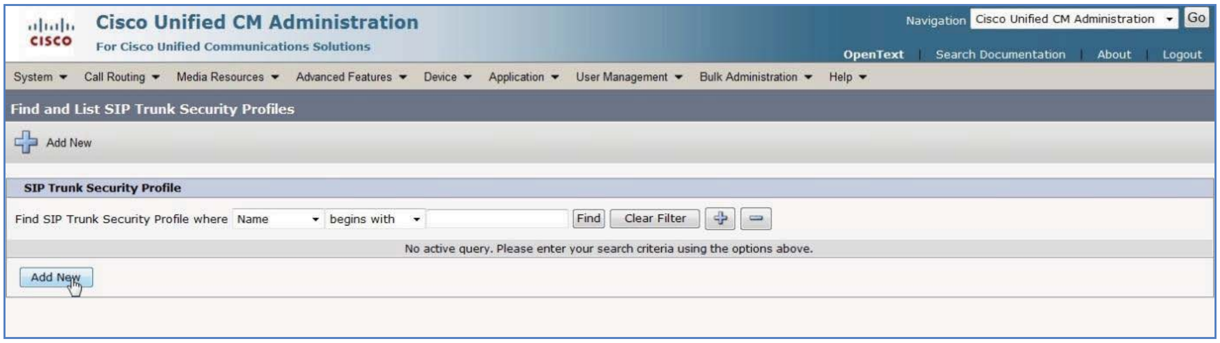
1. Using a web browser, log into the **Cisco Unified CM Administration** screen.



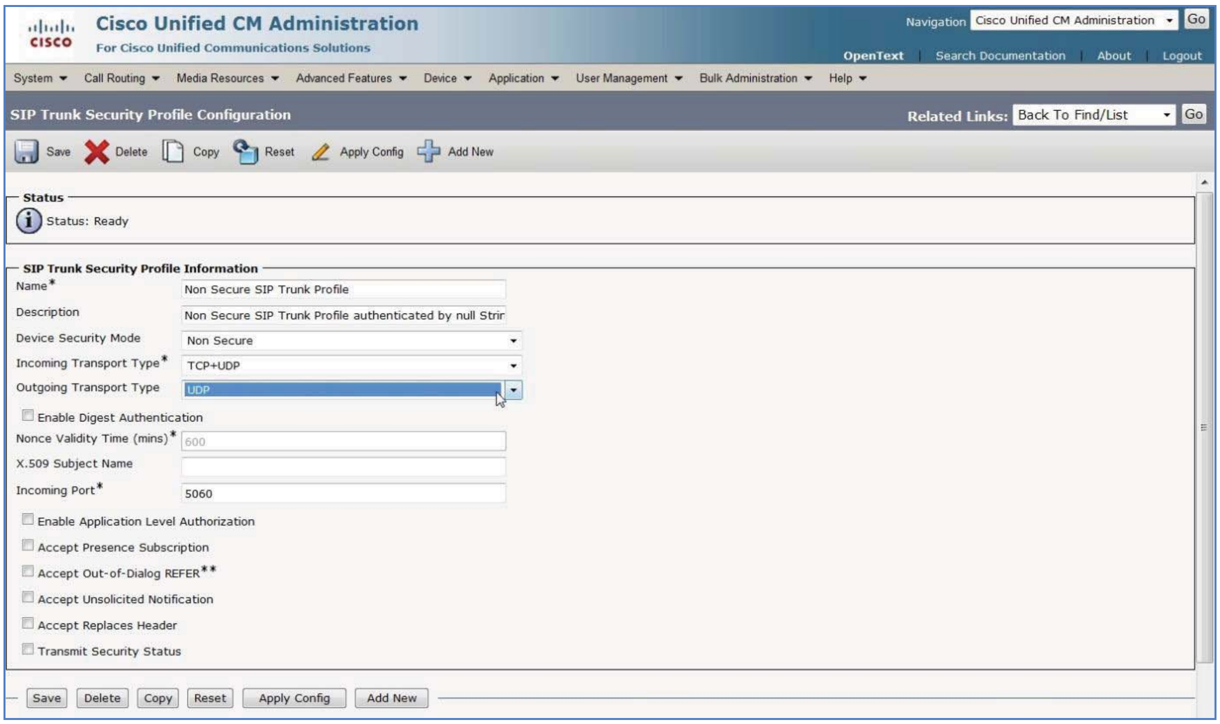
2. From the menu select **System | Security Profile | SIP Trunk Security Profile**.



3. The following screen appears:



4. Click **Find** to edit an existing Sip Trunk Profile or click **Add New** to add a new Sip Trunk Profile. *Note: By default the **Outgoing Transport Type** is set to TCP. OpenText RightFax requires UDP.*

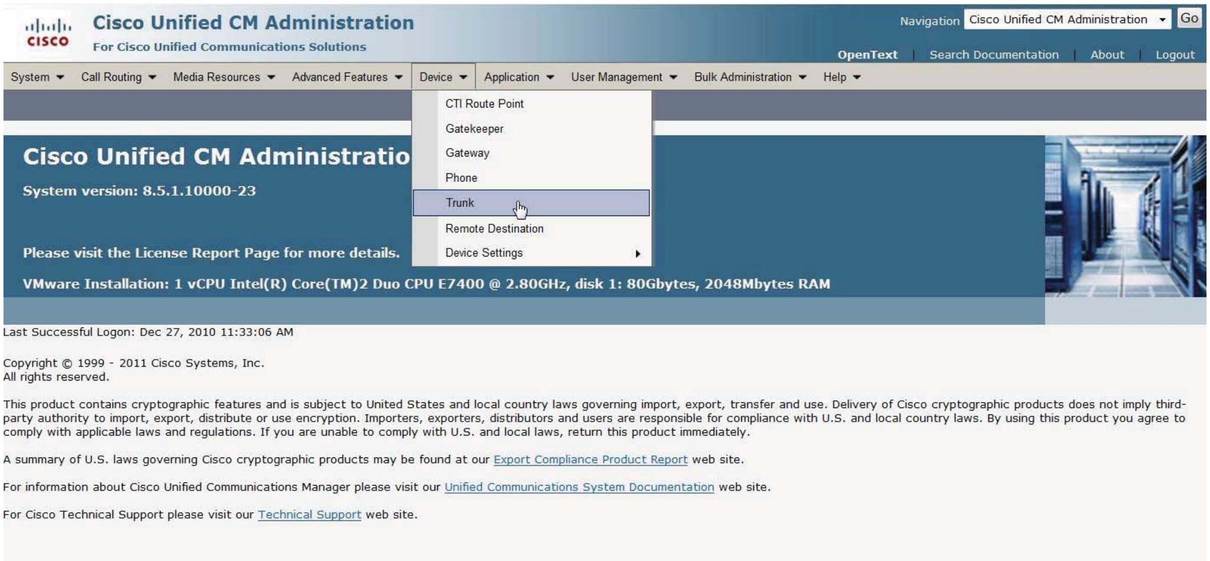


5. Change **Outgoing Transport Type** to UDP.
6. Press **Save**.

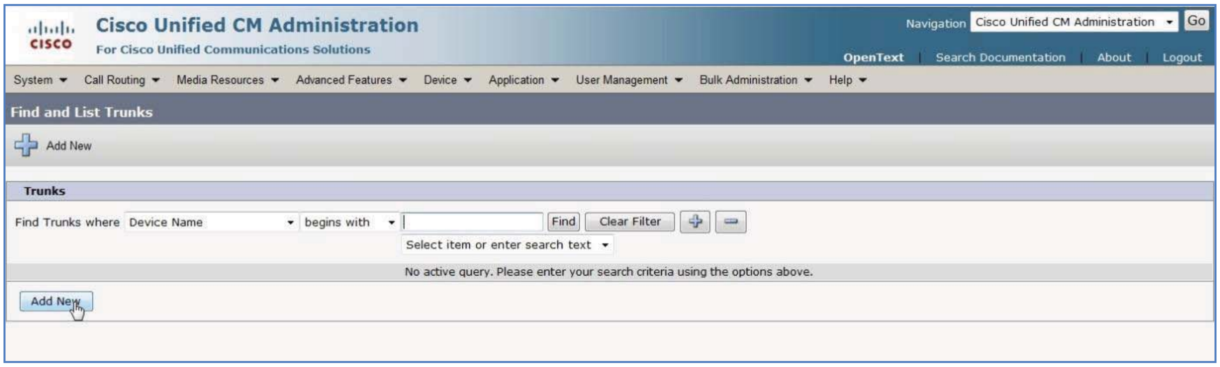
Configure SIP Trunk from CUCM to OpenText RightFax

1. Using a web browser, log into the Cisco Unified CM Administration screen.

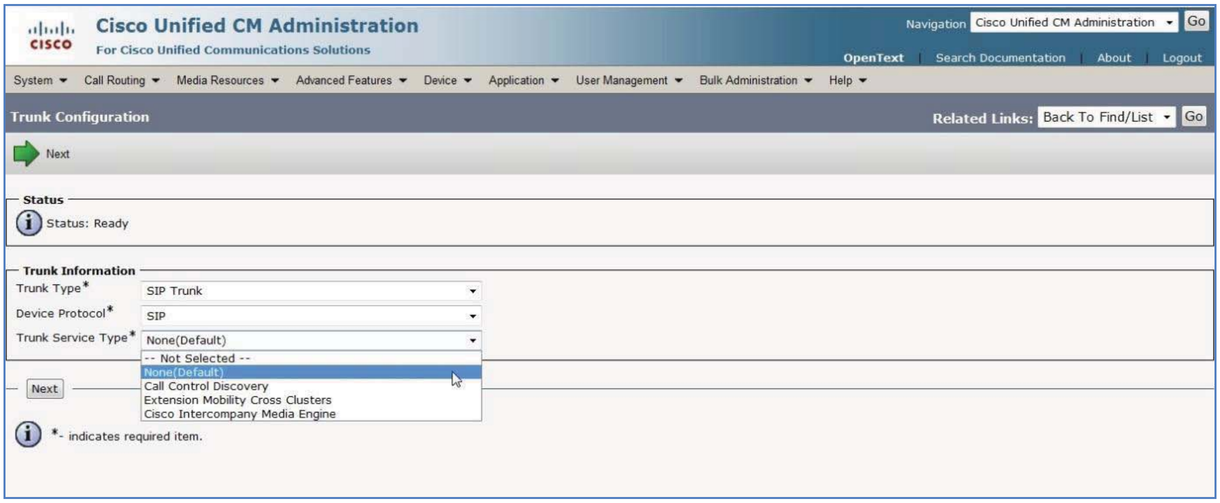
2. From the menu select **Device | Trunk**.



3. The following screen appears:

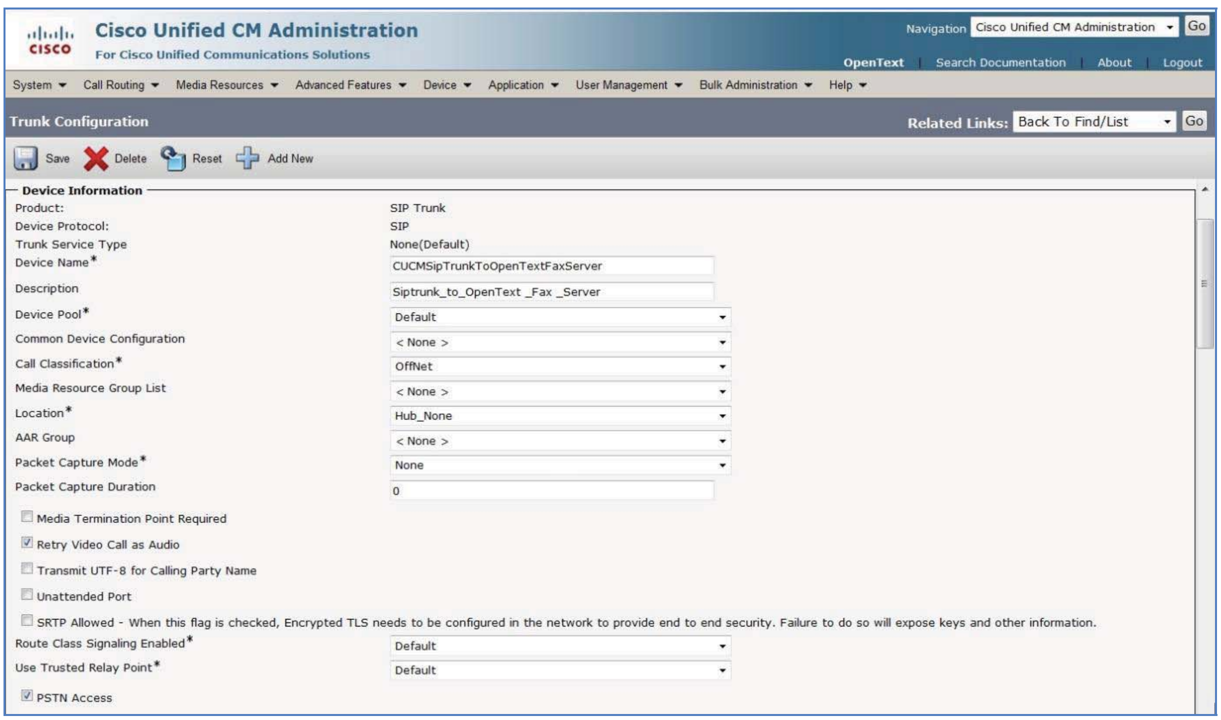


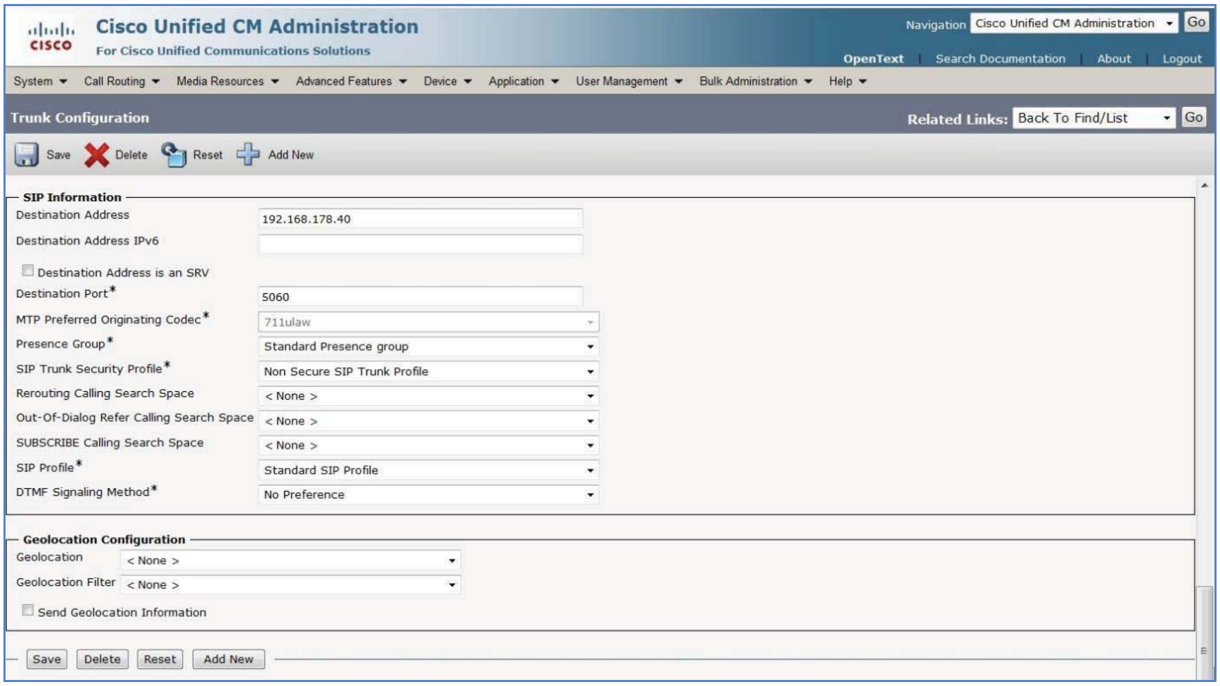
4. Press **Add New** to add a new SIP Trunk.



5. Select the following options and click **Next**:
    a. **Trunk Type** = SIP Trunk
    b. **Device Protocol** = SIP
    c. **Trunk Service Type** = None (Default)

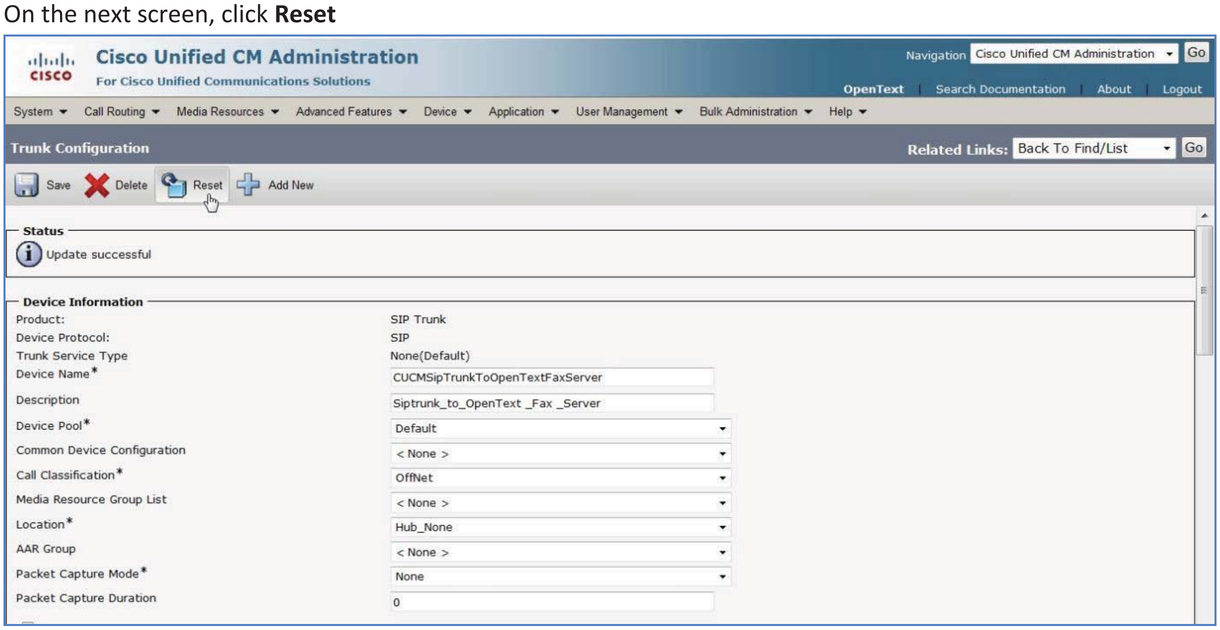
6. The following screen appears:



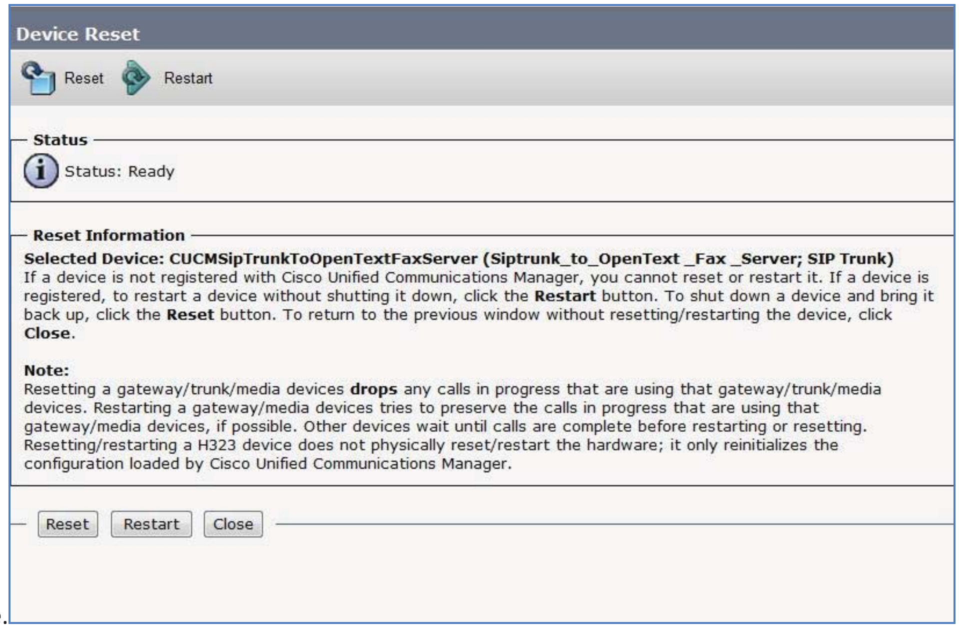


7. Set the following options:
    a. **Device Name**: CUCMSipTrunkToOpenTextFaxServer
    b. **Device Description**: Siptrunk_to_OpenText _Fax _Server
    c. **Device Pool**: Default
    d. **Call Classification**: OffNet
    e. **Destination Address**: 192.168.178.40 (address of OpenText RightFax)
    f. **SIP Trunk Security Profile**: Non Secure SIP Trunk Profile
    g. **SIP Profile**: Standard SIP Profile

8. Click **Save**.
9. On the next screen, click **Reset**



10. Press **Restart** then press

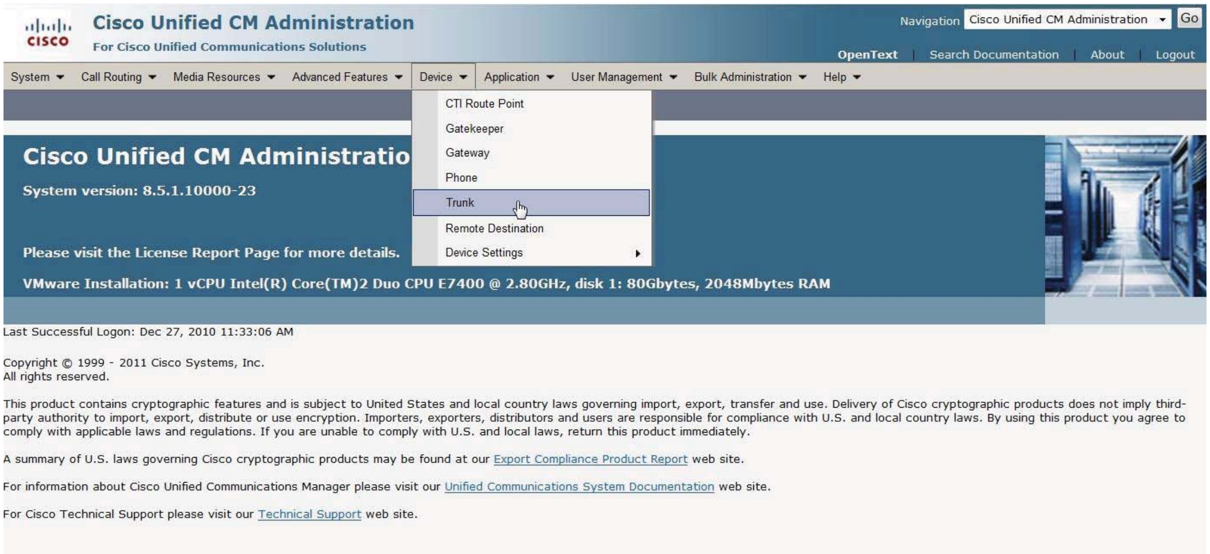


Close.

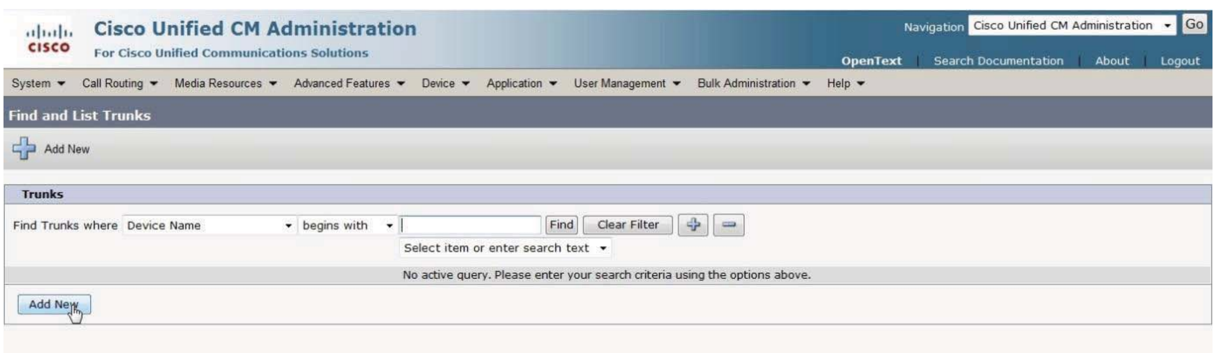
Configure Sip Trunk from CUCM to Gateway

1. Using a web browser, log into the Cisco Unified CM Administration screen.

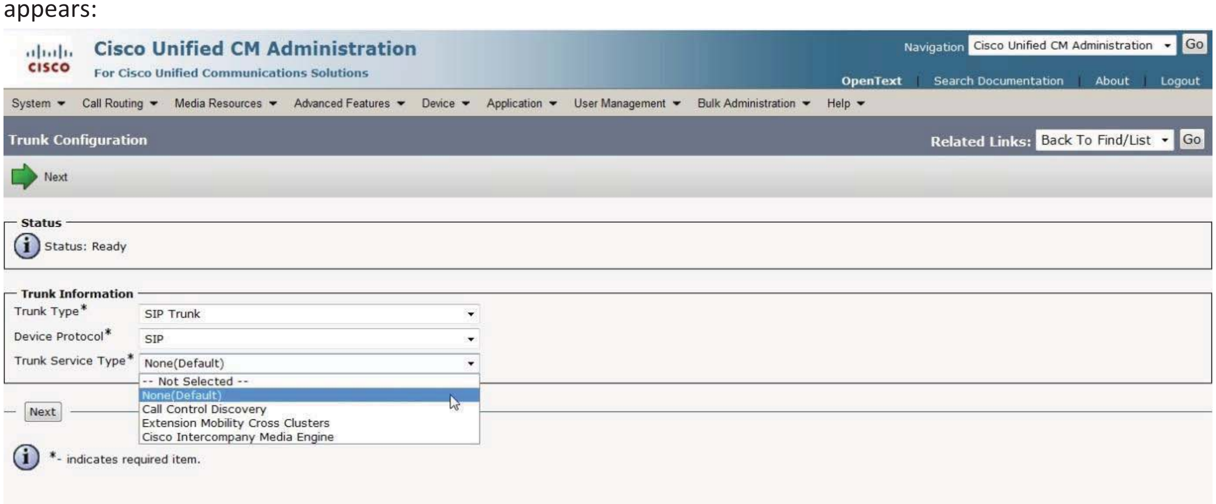
2. From the menu select **Device | Trunk**.



3. Press **Add New**



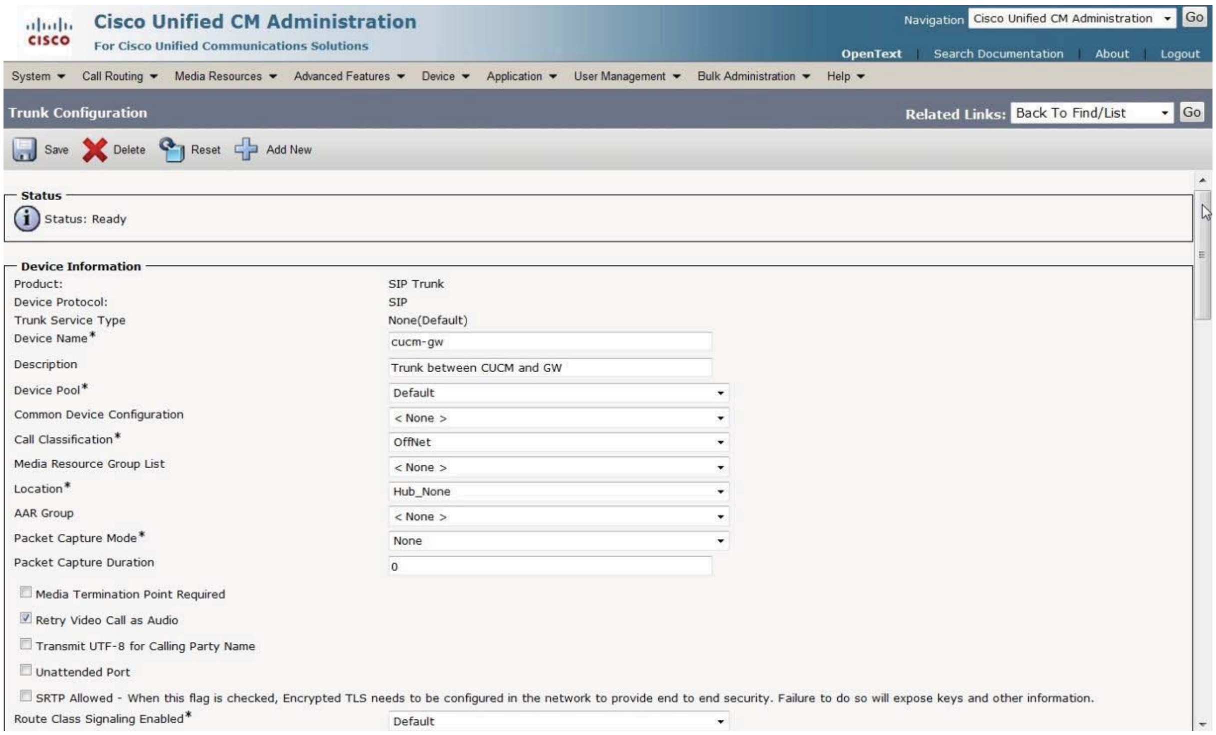
4. The following screen appears:

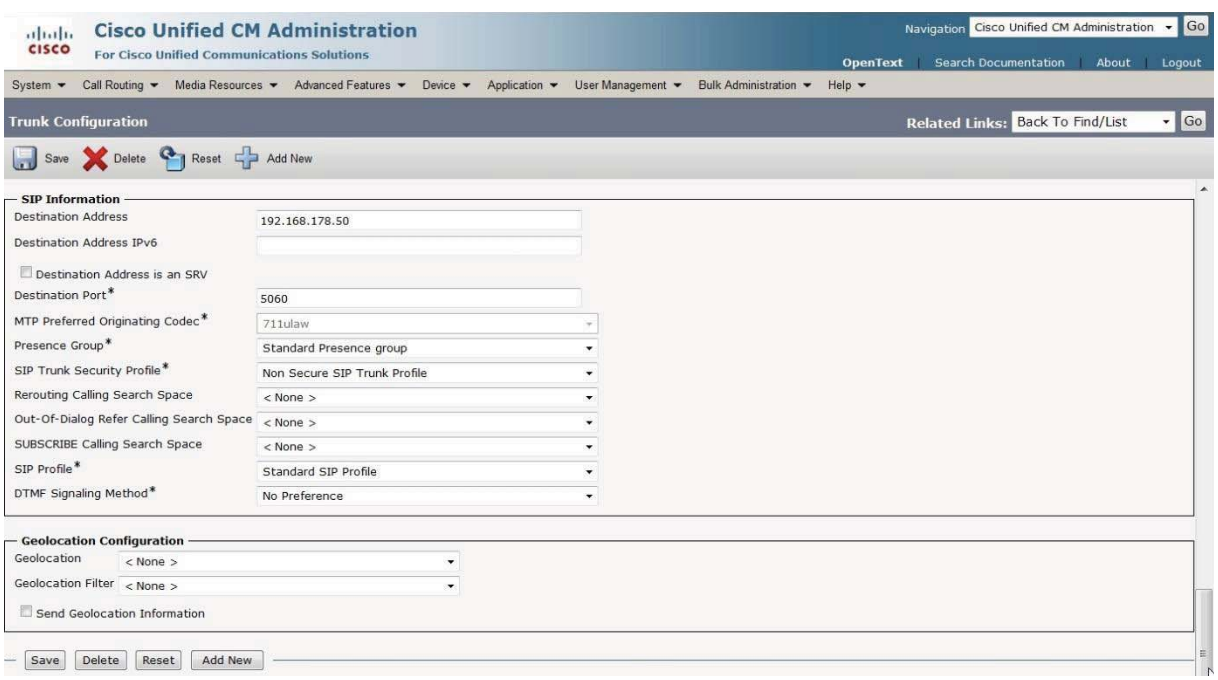


5. Select the following options:
    a. **Trunk Type** = SIP Trunk
    b. **Device Protocol** = SIP
    c. **Trunk Service Type** = None (Default)

6. Click **Next**.

7. The following screen appears:



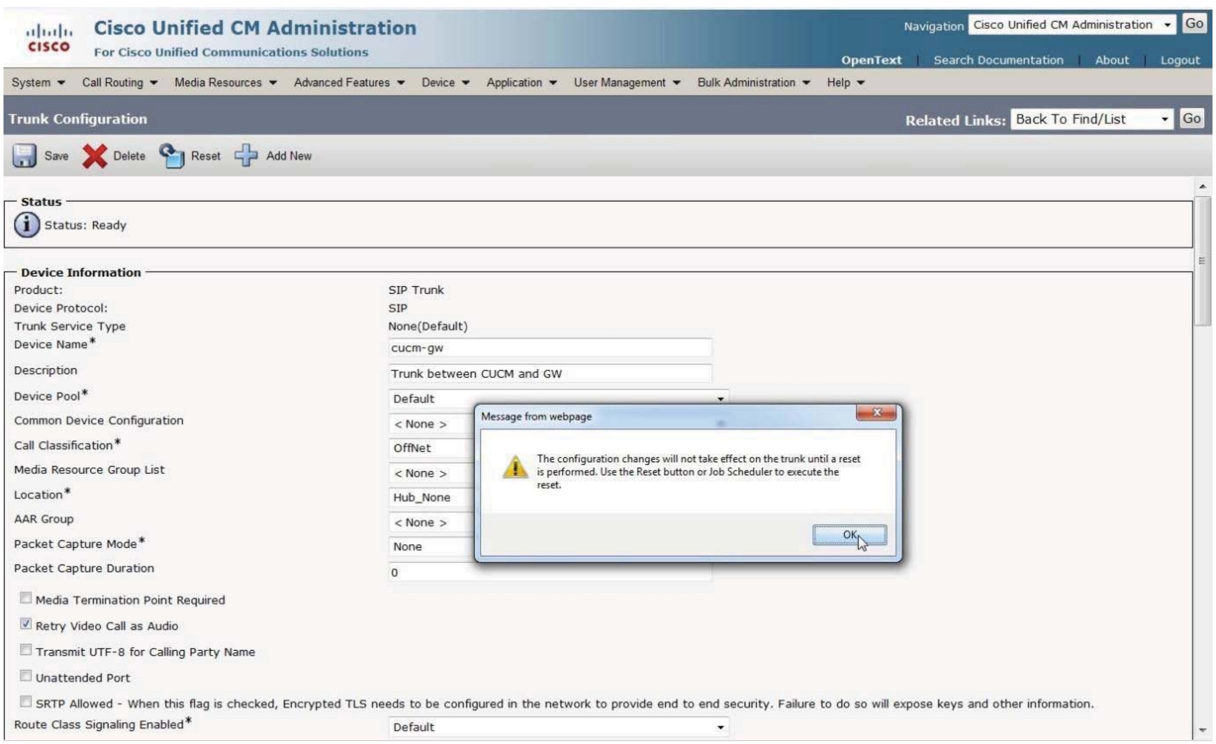


8. Set the following options:

   a. **Device Name**: cucm-gw
   b. **Device Description**: Trunk between CUCM and GW
   c. **Device Pool**: Default
   d. **Call Classification**: OffNet
   e. **Destination Address**: 192.168.178.50
   f. **SIP Trunk Security Profile**: Non Secure SIP Trunk Profile
   g. **SIP Profile**: Standard SIP Profile

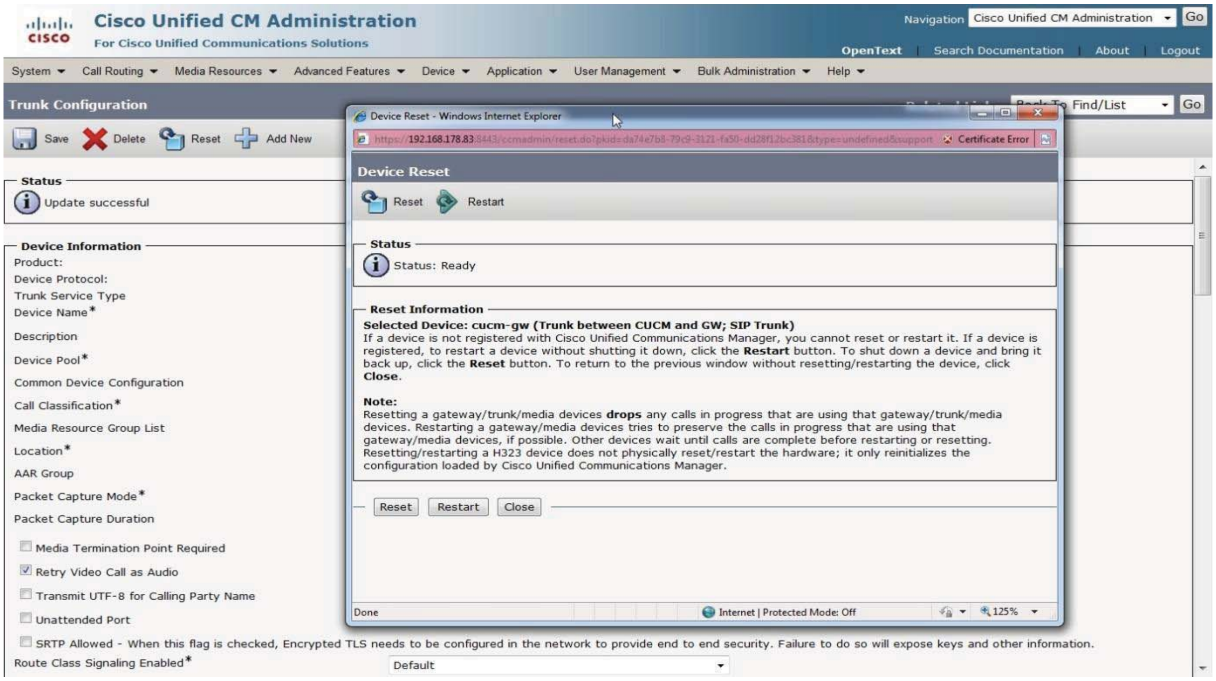
   *Note: **Destination Address** is the IP address of the Gateway.*

9. Press **Save**



10. Press **OK**.

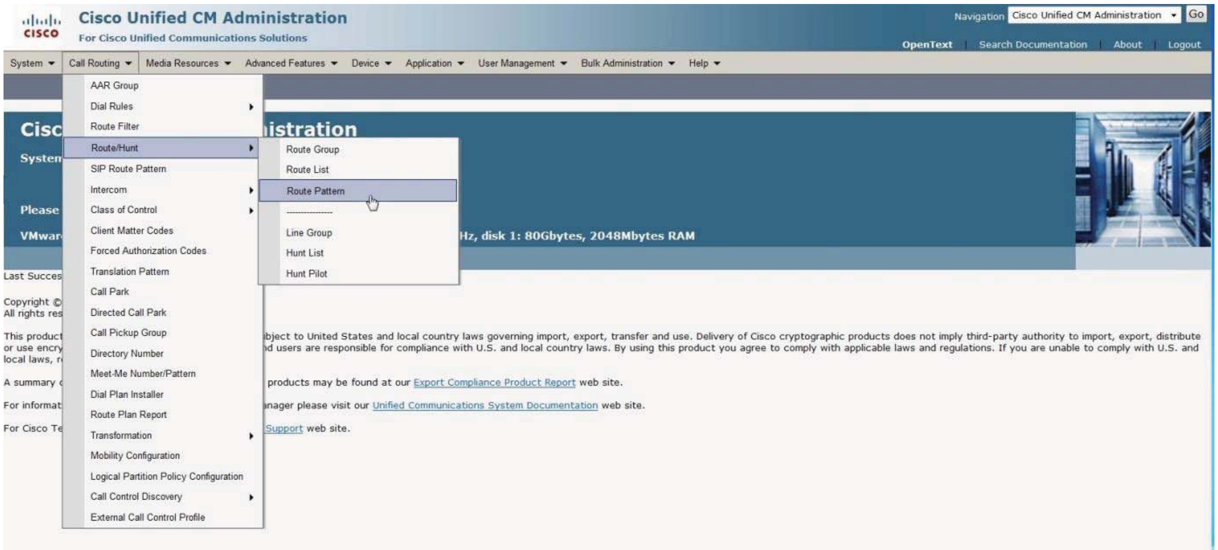
11. Press **Reset.**
12. Press **Restart** and **Close**.



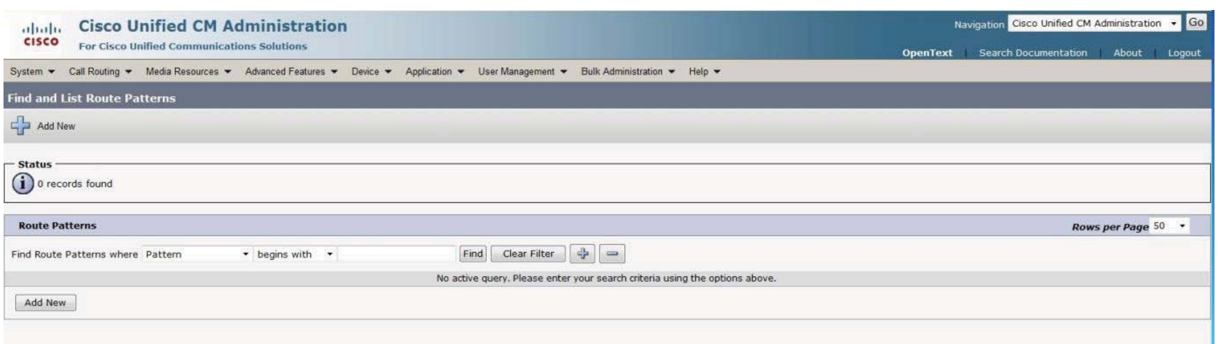
Configure Call Routing (From OpenText RightFax to PSTN)

1. Using a web browser, log into the Cisco Unified CM Administration screen.

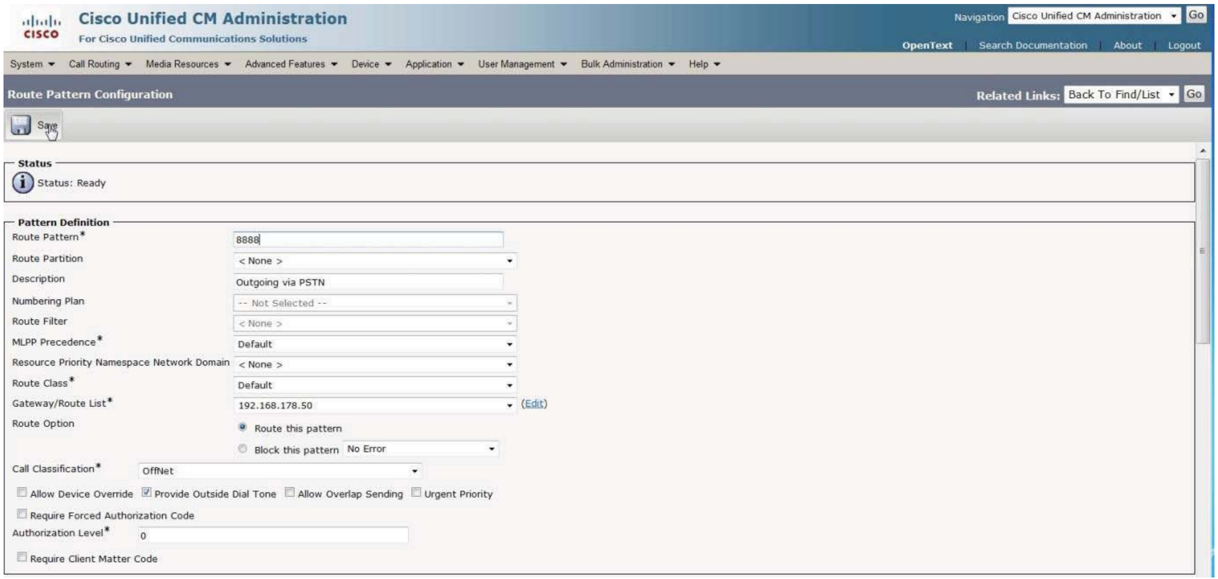
2. From the menu, select **Call Routing | Route / Hunt | Route Pattern**



3. Click on **Add New** to add a new Route Pattern



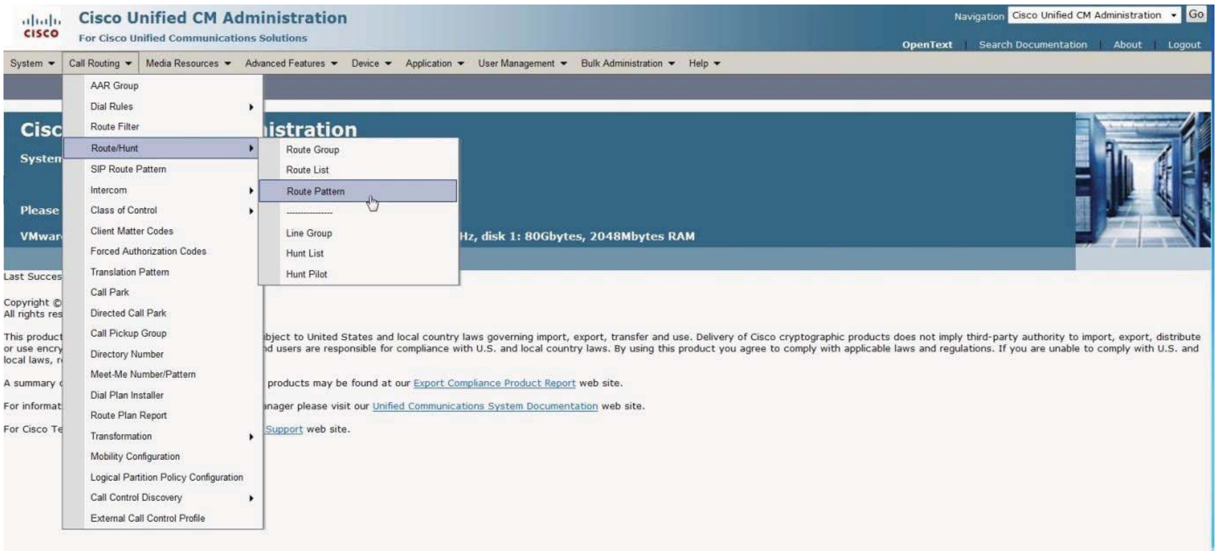
4. Route pattern “8888” is the format to send the fax via the E1 (PSTN)



5. In the **Gateway/Route List**, enter the IP address (192.168.178.50) of the Voice Gateway sending out Fax calls.

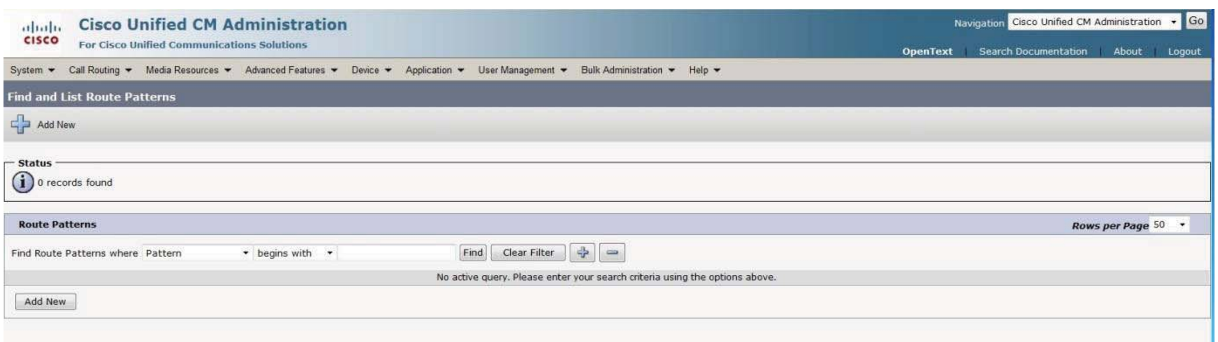
Configure Call Routing (From PSTN to OpenText RightFax)

1. From the Cisco Unified CM Administration screen, select **CallRouting | Route Hunt | Route Pattern**.

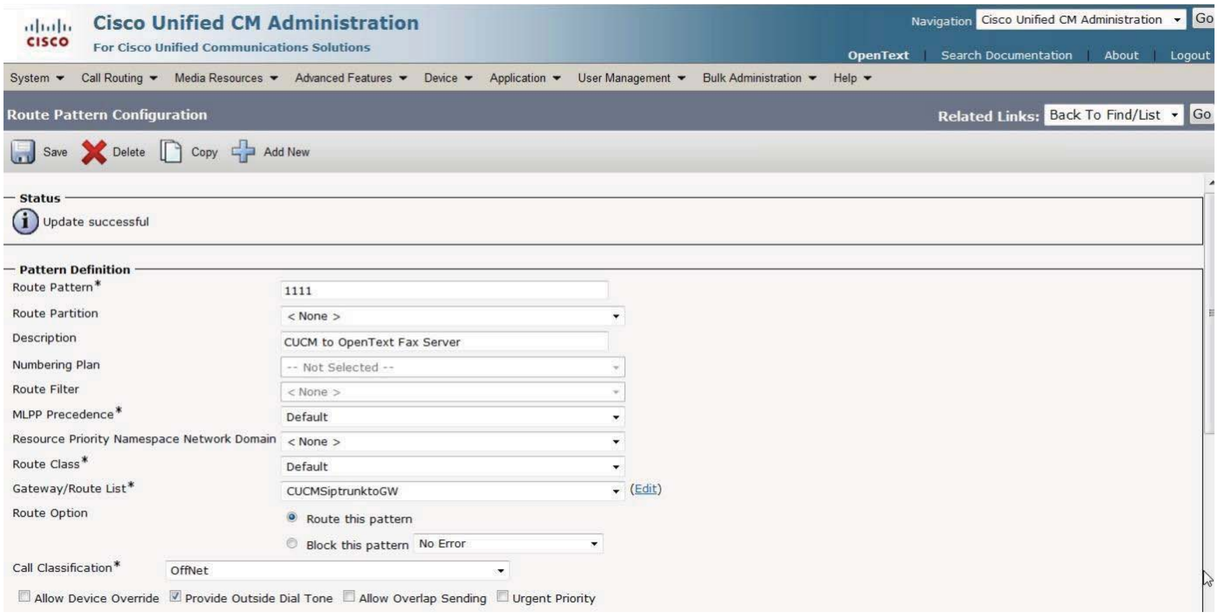


2. Click **Add New**

.



3. The following screen appears:



Set options as follows:

a. • **Route Pattern**: 1111
b. • **Description**: CUCM to OpenText RightFax
c. • **Gateway/Route List**: CUCMSiptrunktoGW
d. • **Call Classification**: OffNet

"1111" in the **Route Pattern** field will send a fax from PSTN to OpenText RightFax thru CUCM.

4. Click **Save**.

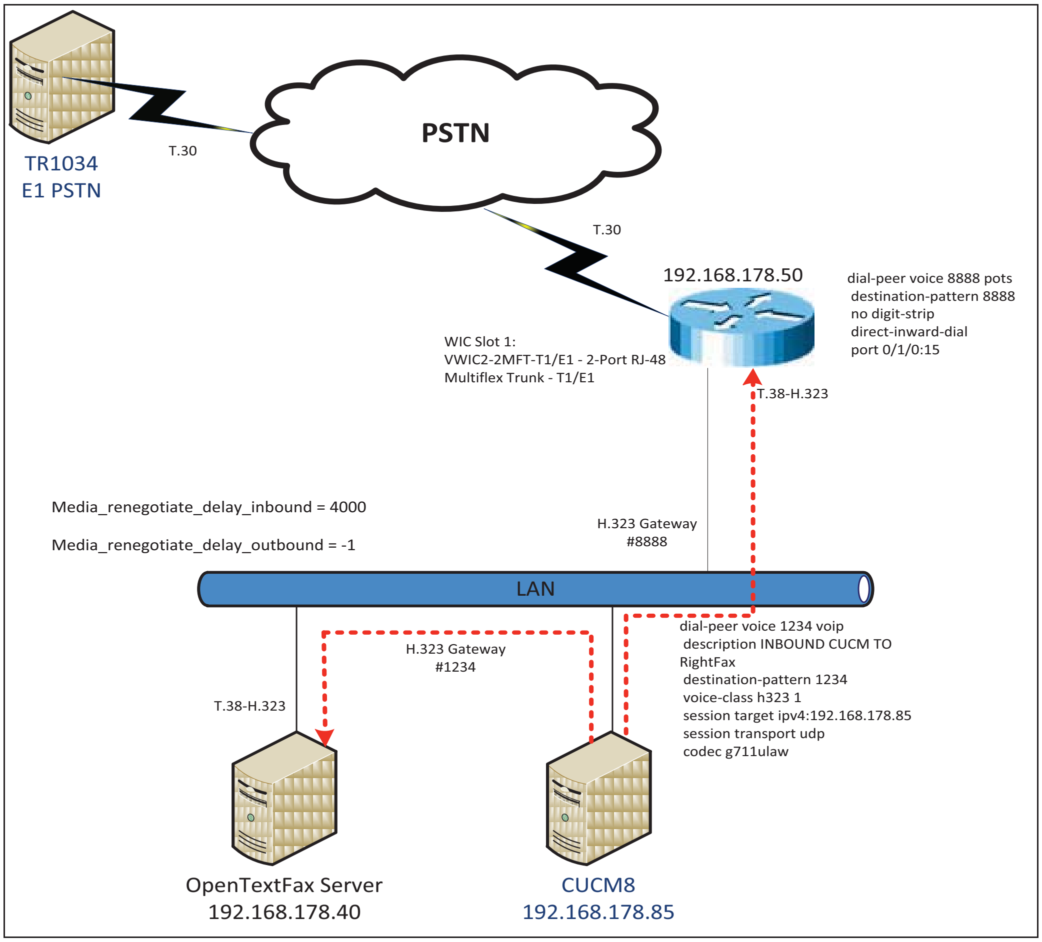
IOS overview

```
ip domain name fritz.box
ip name-server 192.168.178.1
ip auth-proxy max-nodata-conns 3
ip admission max-nodata-conns 3
!
isdn switch-type primary-net5
!
voice-card 0
 dspfarm
!
!
!
voice service voip
 fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none
 sip
!

voice class codec 1
 codec preference 1 g711alaw
!
!
controller E1 0/0/0
 clock source internal
 pri-group timeslots 1-8,16
!
!
interface GigabitEthernet0/0
 ip ddns update dijkje
 ip address 192.168.178.50 255.255.255.0
 duplex half
 speed auto
 no keepalive
 no mop enabled
!
interface Serial0/0/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5
 isdn protocol-emulate network
 isdn incoming-voice voice
 no cdp enable
!
no ip forward-protocol nd
!

!
voice-port 0/0/0:15
```

```
!
voice-port 0/1/0
 compand-type a-law
 cptone NL
 description fxo00
 bearer-cap Speech
!
voice-port 0/1/1
 compand-type a-law
 cptone NL
 description FX01
 bearer-cap Speech
!
!
!
!
!
!
dial-peer voice 1111 voip
 description inbound Fax traffic from Sip to OpenText  RightFax
 destination-pattern 1111
 session protocol sipv2
 session target ipv4:192.168.178.85
 codec g711ulaw
 fax rate 14400
!
dial-peer voice 8888 pots
 destination-pattern 8888
 no digit-strip
 direct-inward-dial
 port 0/0/0:15
gateway
 timer receive-rtp 1200
!
sip-ua
!
!
!
!
scheduler allocate 20000 1000
!
end
```

Scenario 2: H.323-to-H.323 Configuration

Network System Configuration – MGCP / H.323 Configuration



Network Addresses

Device #	Device Make, Model, and Description	Device IP Address
1	OpenText RightFax	192.168.178.40
2	CUCM 8.5.10000-23)	192.168.178.85
3	Cisco 2800 Integrated Service Router	192.168.178.50

Dialing Plan Overview

To call the SR140 from a POTS phone, dial 1234

- POTS (dial 1234—E1—>
- Gateway (dial 1234@192.168.178.85)—H.323—>
- CUCM8.5.10000-23 (dial 1234@192.168.178.40)—H.323—>
- OpenText RightFax.

To call the POTS lines of the Gateway, dial 8888@192.168.178.83

- OpenText RightFax (8888@192.168.178.85)—H.323—>
- CUCM8.5.10000-23 (dial 8888@192.168.178.50)—H.323—>
- Gateway(dial 88088)—E1—>
- POTS

OpenText RightFax SR140 Setup Notes

In this scenario, Dialogic SR140 is required non-default values. For RightFax version 9.4 FP1 SR2 (Dialogic SDK 6.3.0 and later), the following parameters must be set under T.38 Parameters:

- Media Renegotiate Delay Inbound, msec = 4000
  - Callctrl.cfg value = Media_renegotiate_delay_inbound
- Media Renegotiate Delay Inbound, msec = -1
  - Callctrl.cfg value = Media_renegotiate_delay_outbound

Dialogic® Brooktrout® TR1034 Fax PSTN Setup Notes

For the sample test configuration, the TR1034 was configured using the default values, consult the Dialogic® Brooktrout® Fax Products Installation and Configuration Guide for details.

Cisco 2800 Gateway Setup Notes

For the sample test configuration, the Cisco 2800 Gateway was configured the Cisco IOS command-line interface. The specific items configured include:

- Enable T.38 support
- Configure line card interface
- Configure IP Protocol
- Configure Dial-Peers – POTS
- Configure Dial-Peers – VoIP

Enable T.38 support

The following lines allow H.323 calls and T.38 fax calls:

```
voice service voip
 fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none
 h323
 session transport udp
 h245 tunnel disable
```

Note: OpenText RightFax supports FoIP via UDP protocol only; therefore, `session transport` *must contain "udp".*

Configure line card interface

```
controller E1 0/0/0
 clock source internal
 pri-group timeslots 1-8,16
```

Configure Dial-Peers – POTS

The following will allow the phone "8888" to be dialed out though the POTS lines

```
dial-peer voice 8888 pots
 destination-pattern 8888
 no digit-strip
 direct-inward-dial
 port 0/0/0:15

interface Serial0/0/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5
 isdn protocol-emulate network
 isdn incoming-voice voice
 no cdp enable
```

Configure Dial Peers - VoIP

The following allows the phone number "1234" to be dialed out through H.323 to CUCM:

```
dial-peer voice 1234 voip
 description inbound h323 to OpenText RightFax
 destination-pattern 1234
 voice-class h323 1
 session target ipv4:192.168.178.85
 session transport udp
 codec g711alaw
```

Note: The `session target ipv4` *contains the IP address for CUCM.*

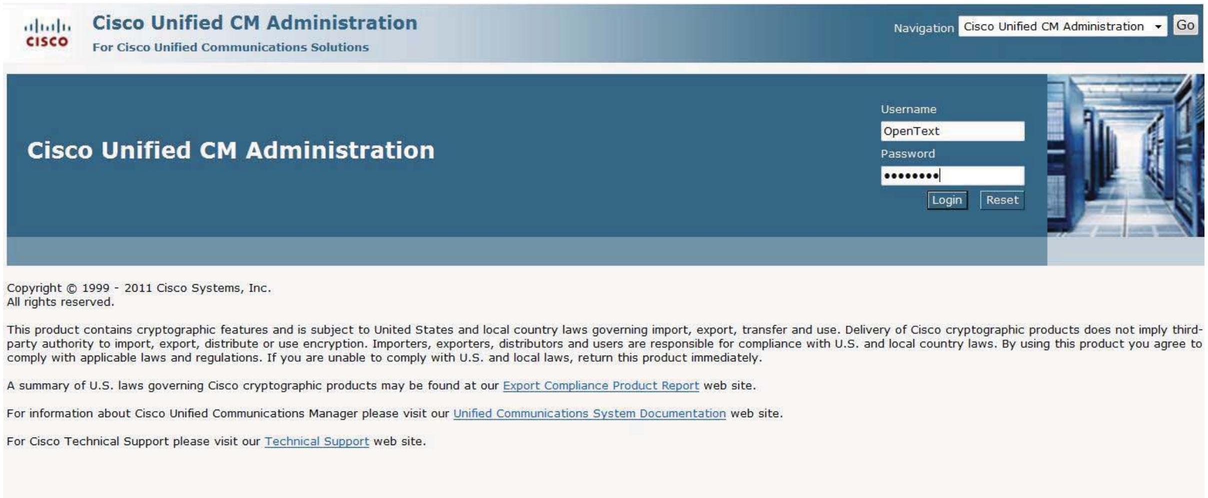
CUCM 8.5 Setup Notes – H.323 / H.323 Configuration

The following areas of CUCM 8.5.10000-23 are modified in this scenario:

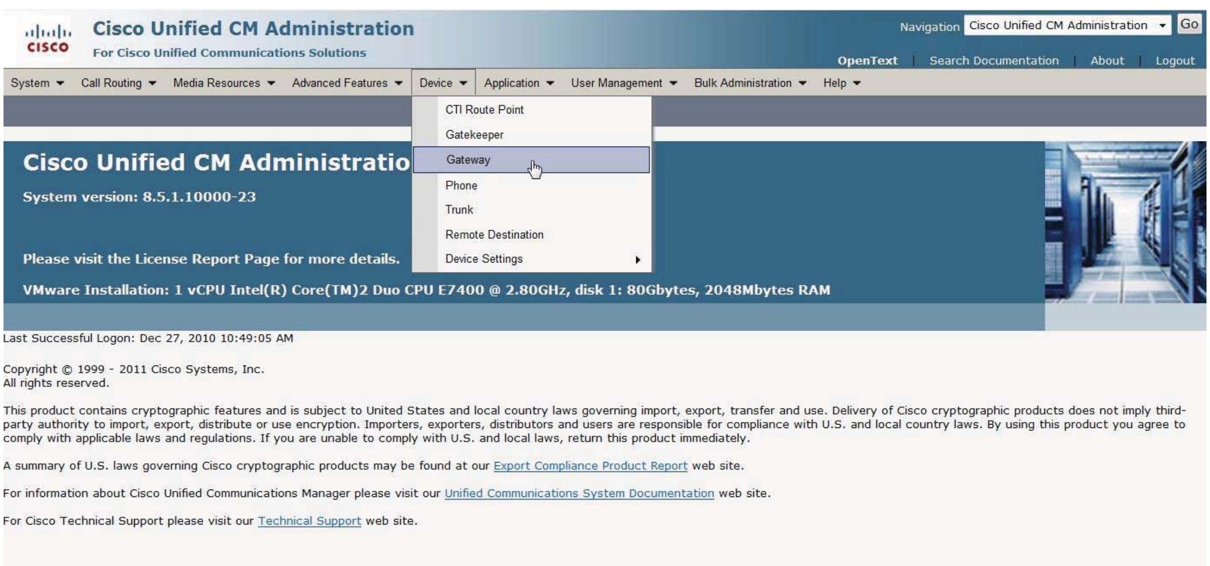
- Configure OpenText RightFax Gateway
- Configure Gateway
- Configure Call Routing

Configure H.323 Gateway to OpenText RightFax

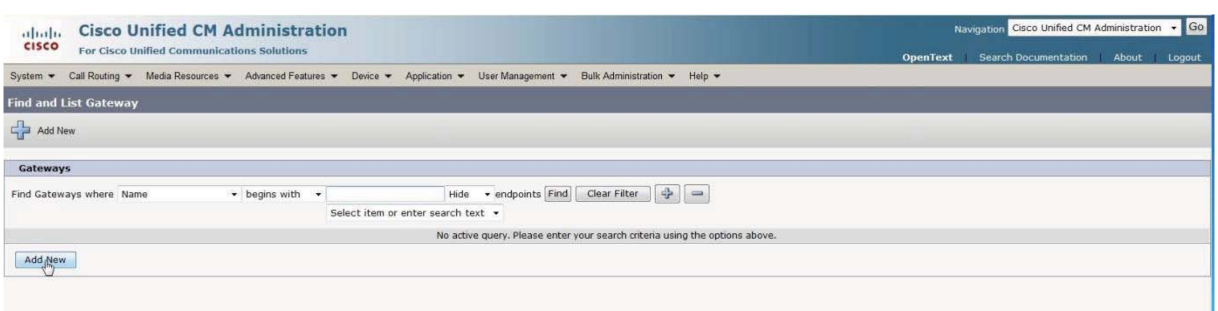
1. Using a web browser, log into the Cisco Unified CM Administration screen.



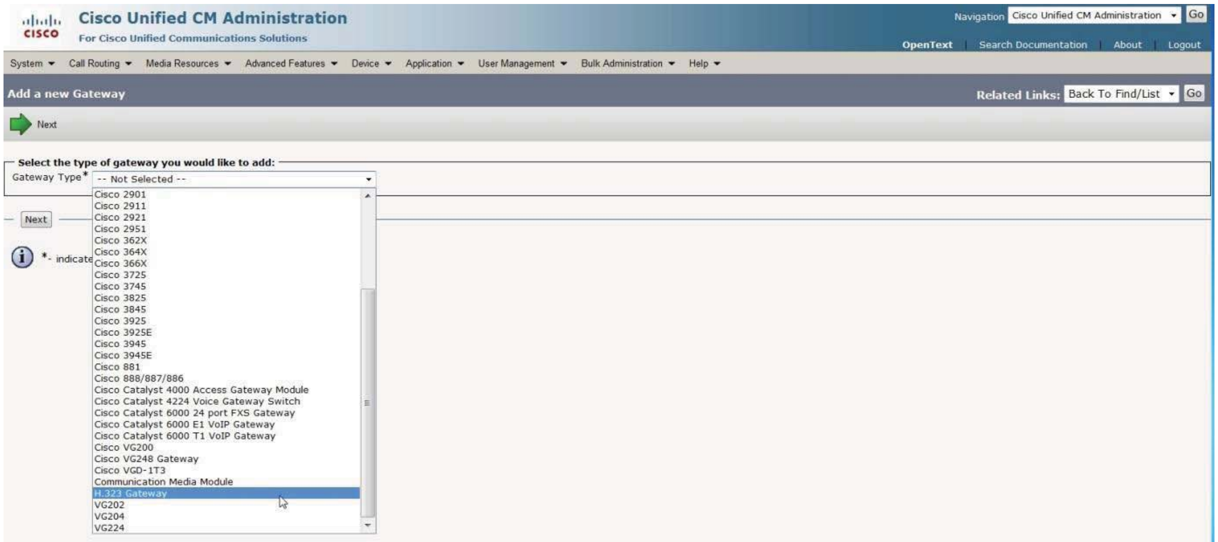
2. From the menu select **Device | Gateway.**



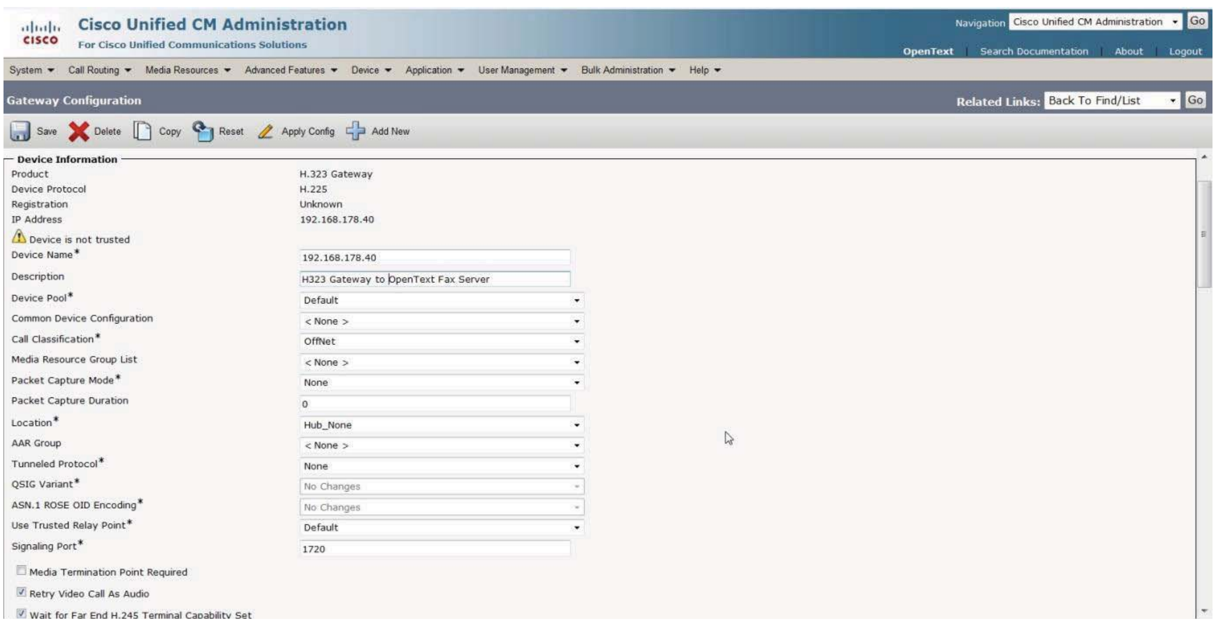
3. Press **Add New** to add a new H.323 Gateway



4. Select **H.323 Gateway** and press Next.



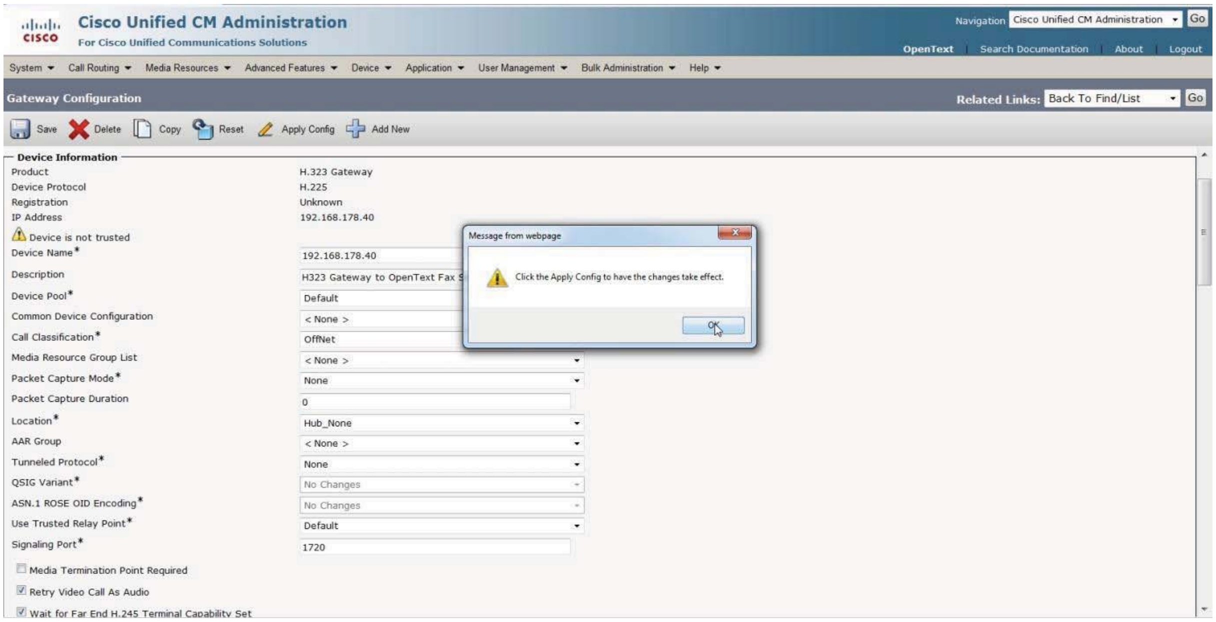
5. The following screen appears:



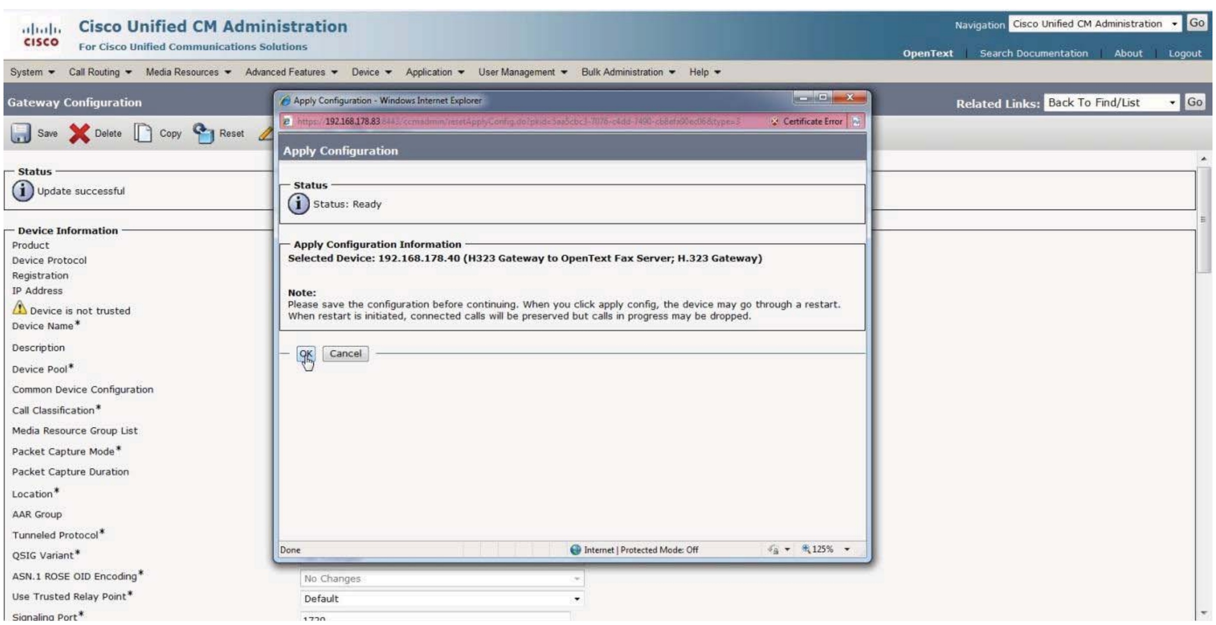
6. Set the following options:

   a. **Device Name**: 192.168.178.40 (address of OpenText RightFax)
   b. **Device Description**: H323 Gateway to OpenText RightFax
   c. **Device Pool**: Default
   d. **Call Classification**: OffNet

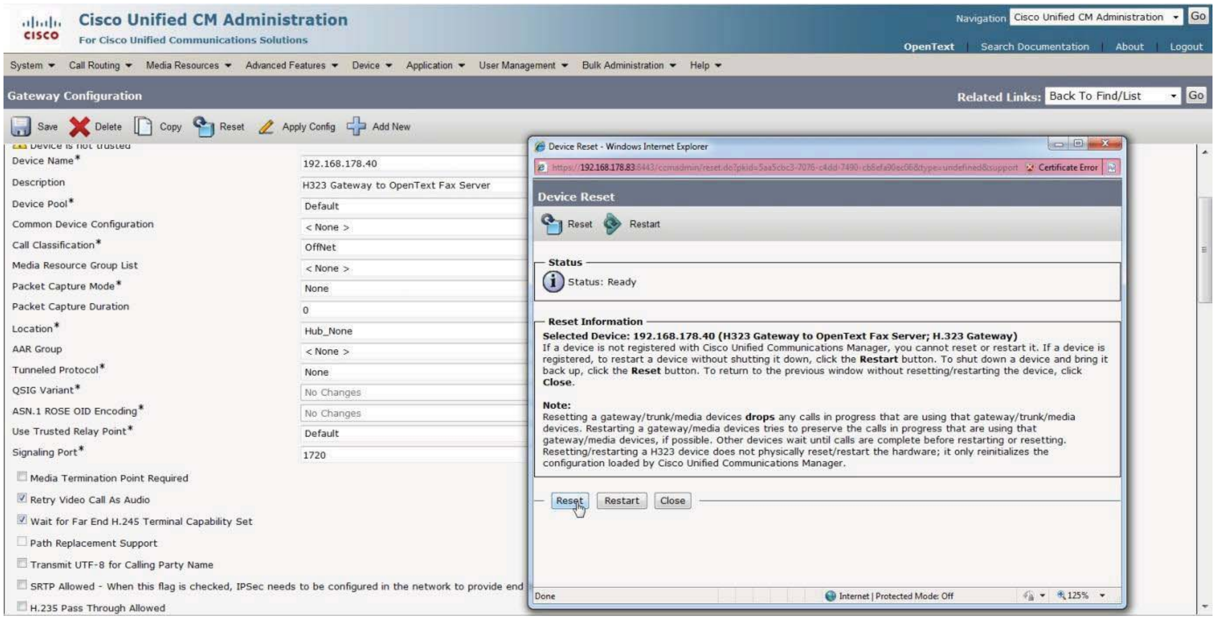
7. Press **Save**.



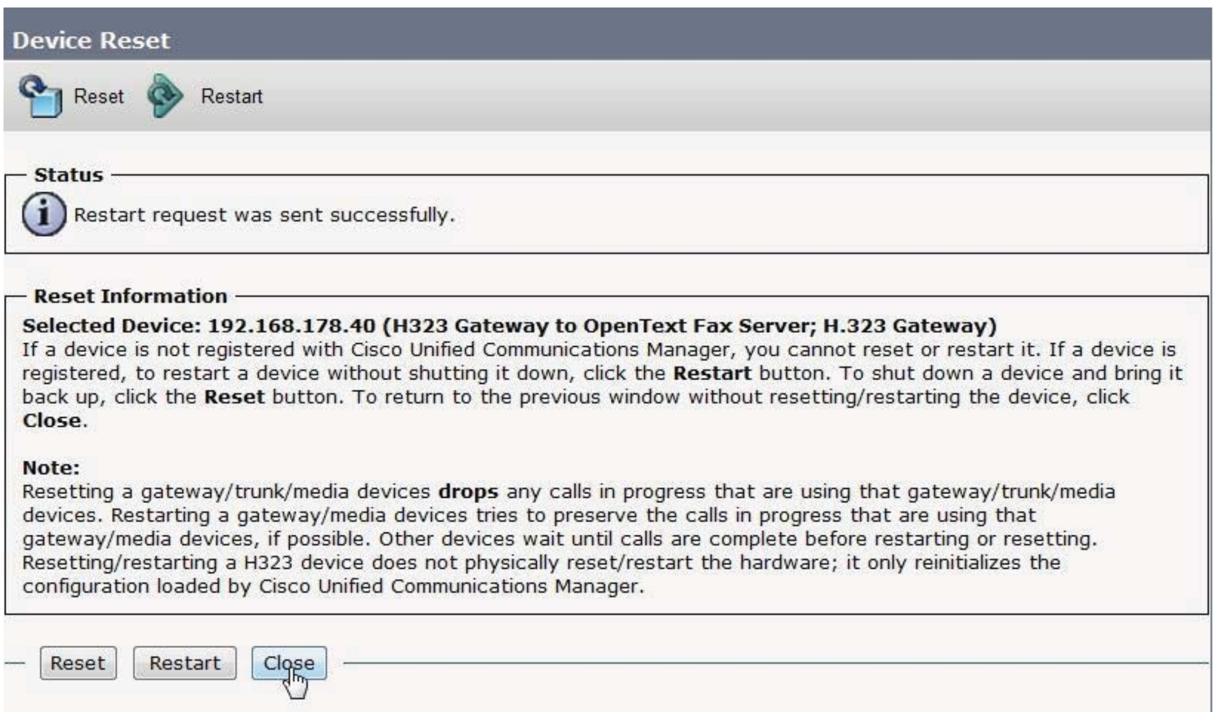
8. Click **OK** then **Apply Config**.



9. Click **OK** then click **Reset**.



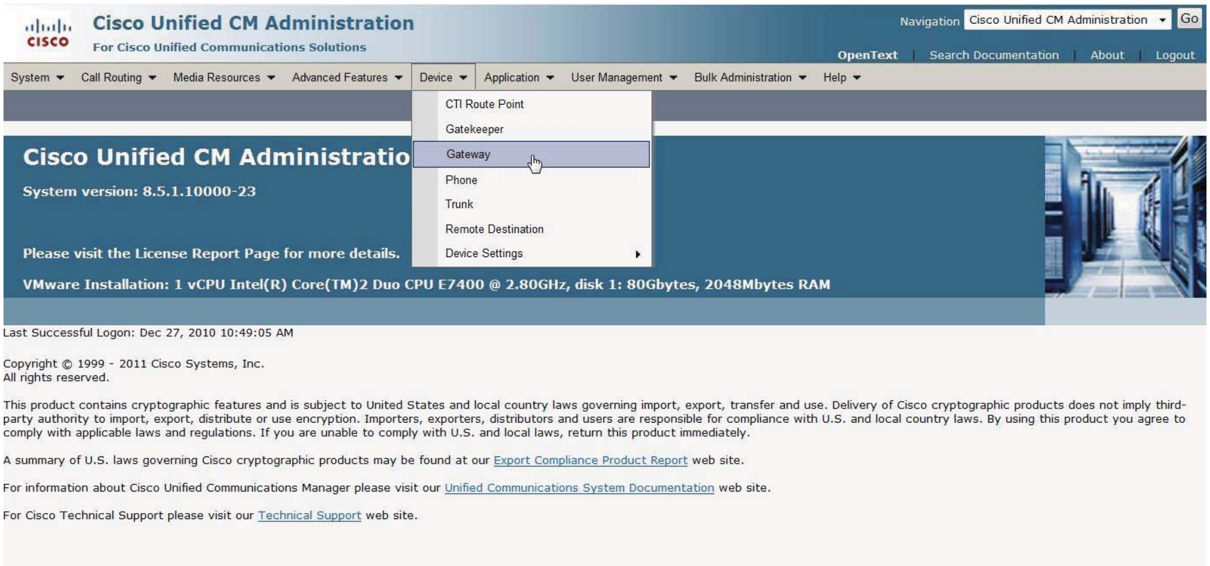
10. Click **Restart** and **Close.**



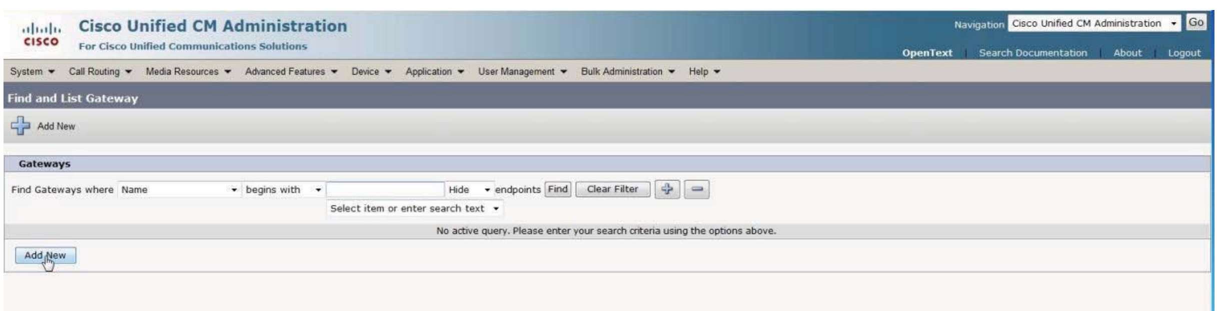
Configure H.323 Gateway to the Cisco Voice Gateway

1. Using a web browser, log into the Cisco Unified CM Administration screen.

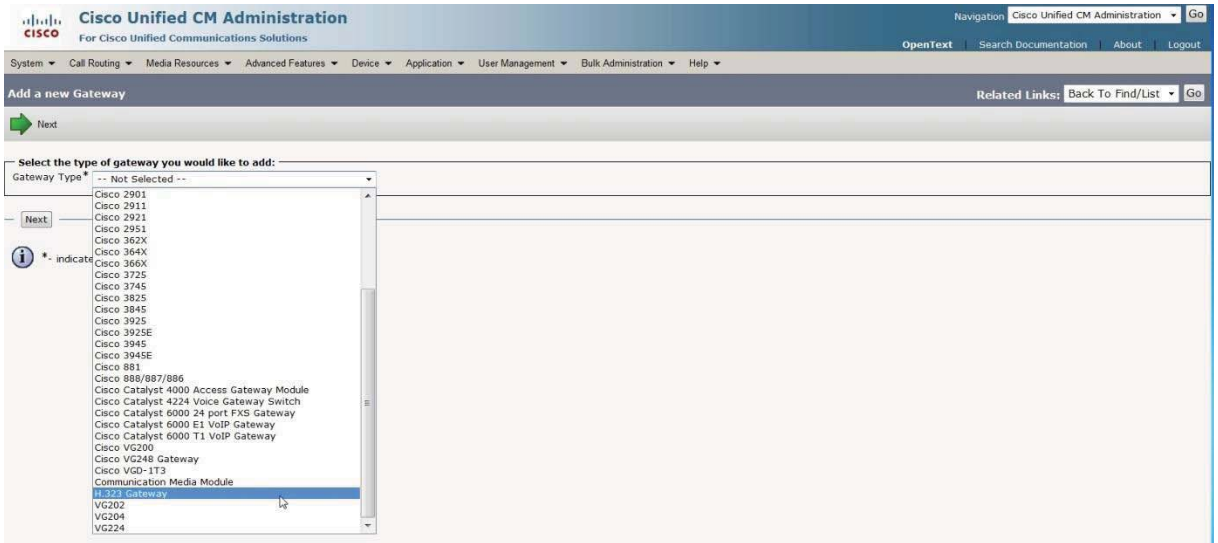
2. From the menu select **Device | Gateway.**



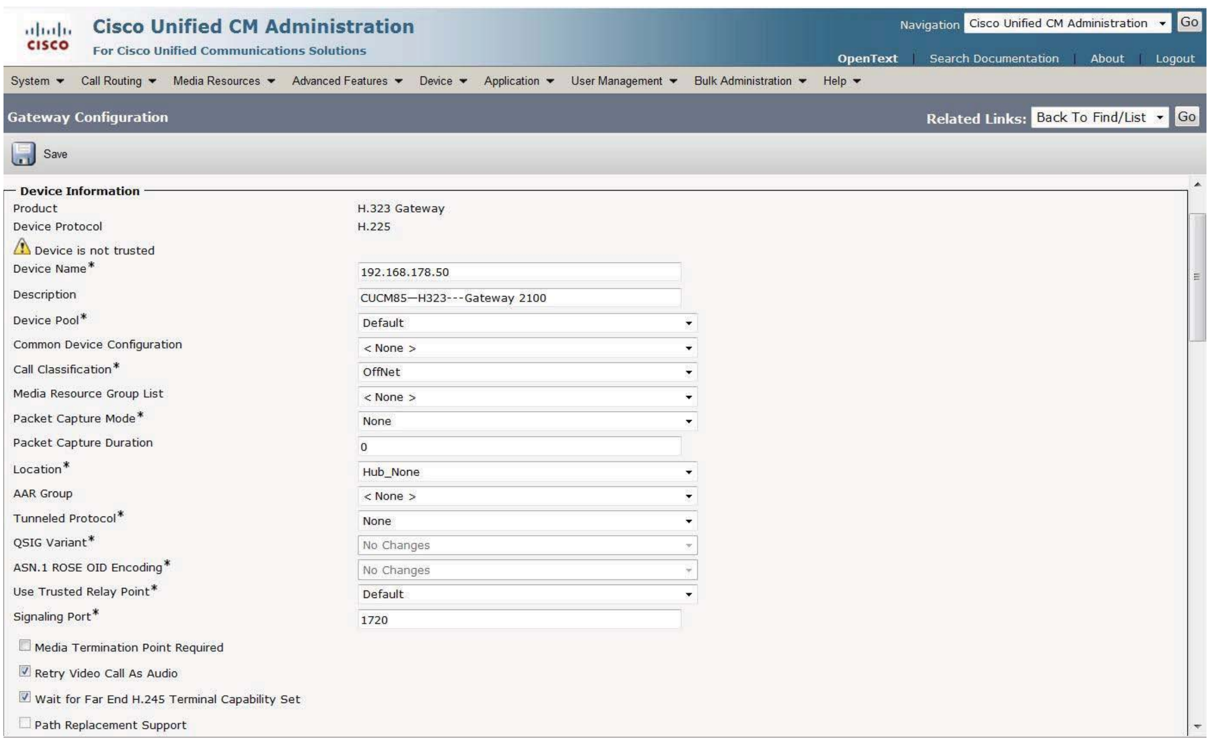
3. Press **Add New** to add a new H.323 gateway



4. Select H.323 Gateway for the **Gateway Type** and press **Next**.



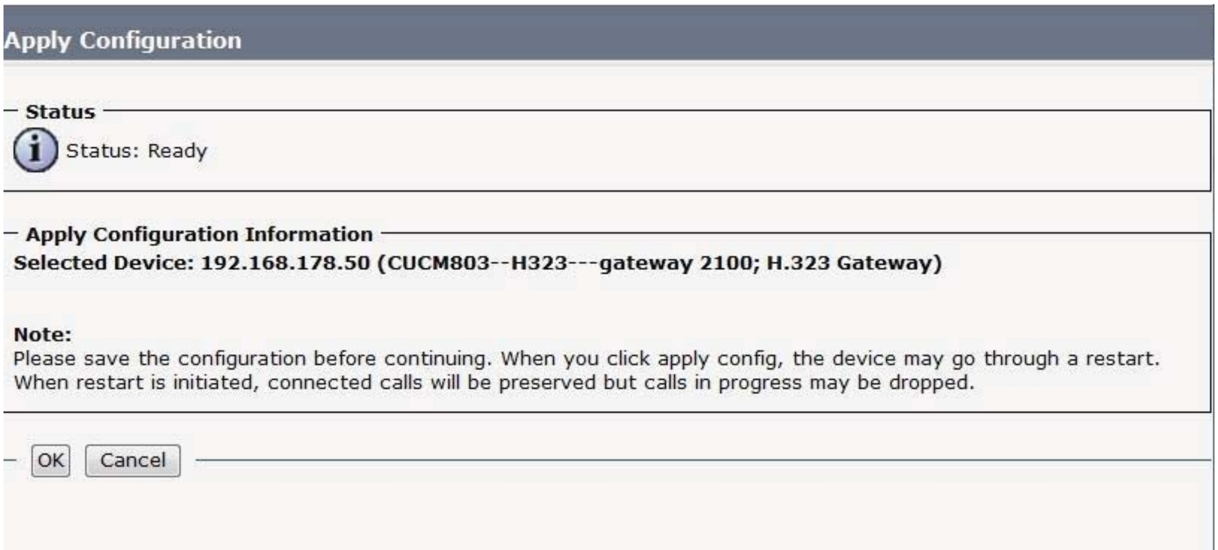
5. The following screen appears:



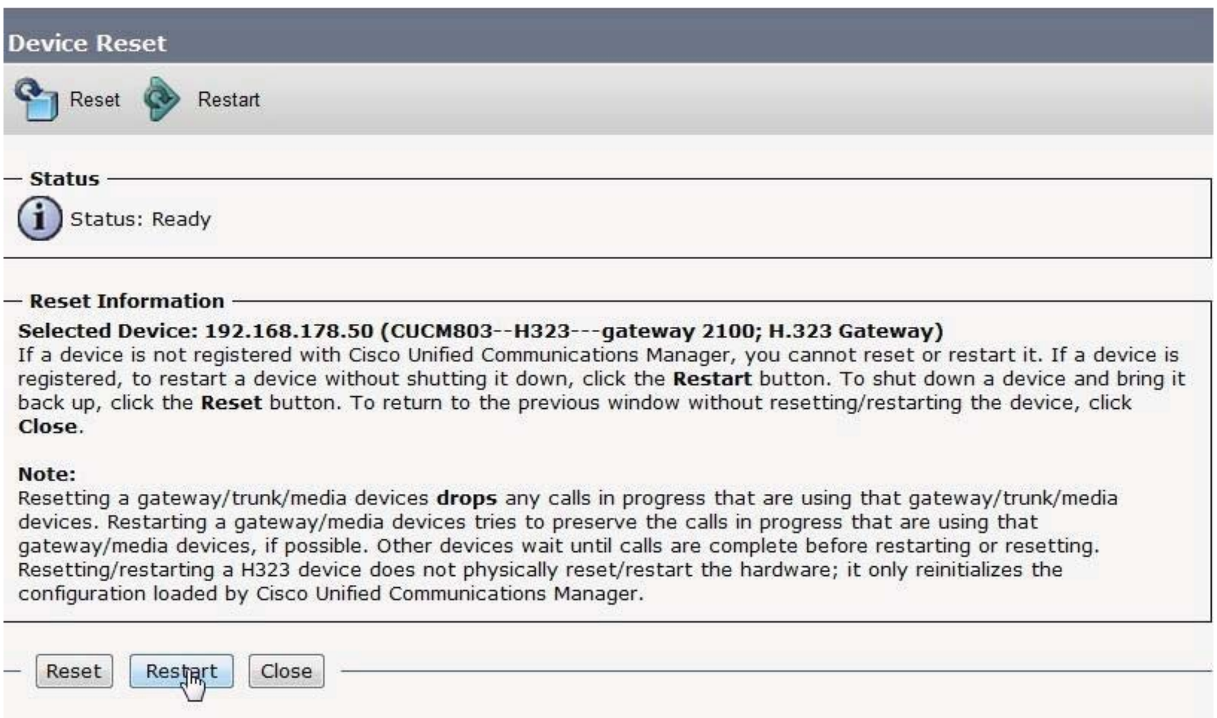
6. Set the following options:

   a. **Device Name:** 192.168.178.50 (address of the Cisco Voice Gateway)
   b. **Device Description**: CUCM85—H323---Gateway 2100
   c. **Device Pool**: Default
   d. **Call Classification**: OffNet

7. Press **Save** and click on **Apply Config**.



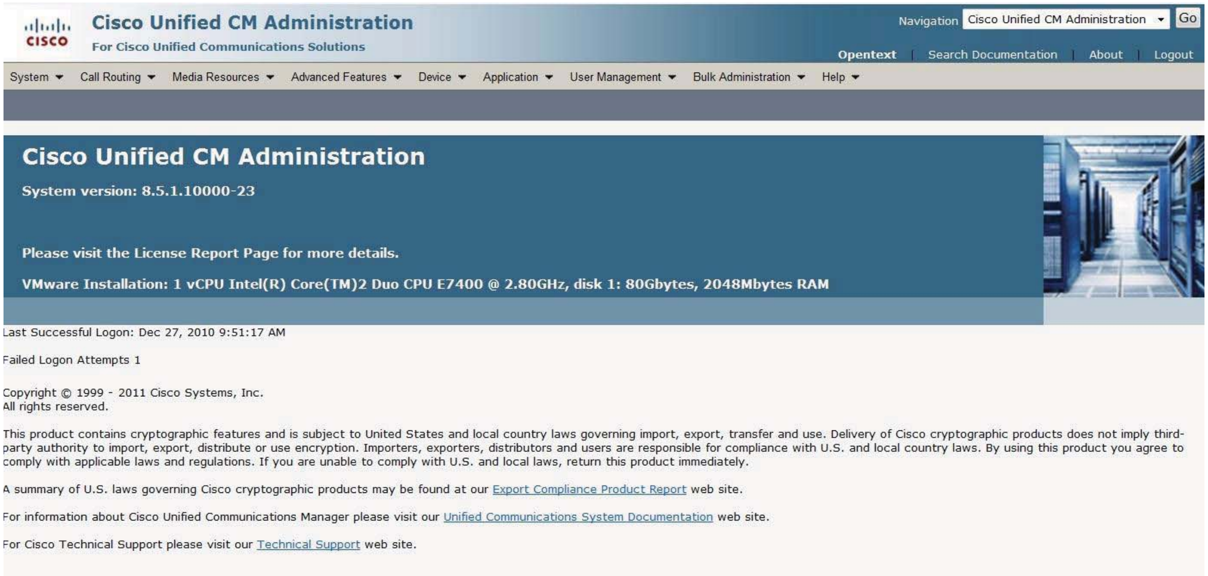
8. Click **OK** to close the window and select **Reset**.



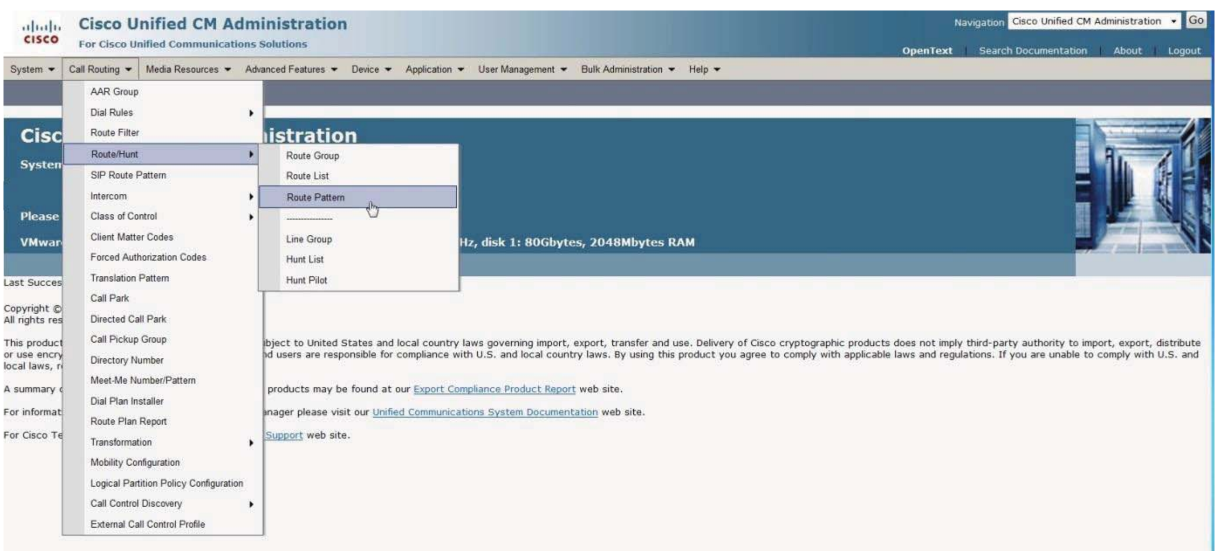
9. Click **Restart** and click **Close** to close the window.

Configure Call Routing (From OpenText RightFax to PSTN)

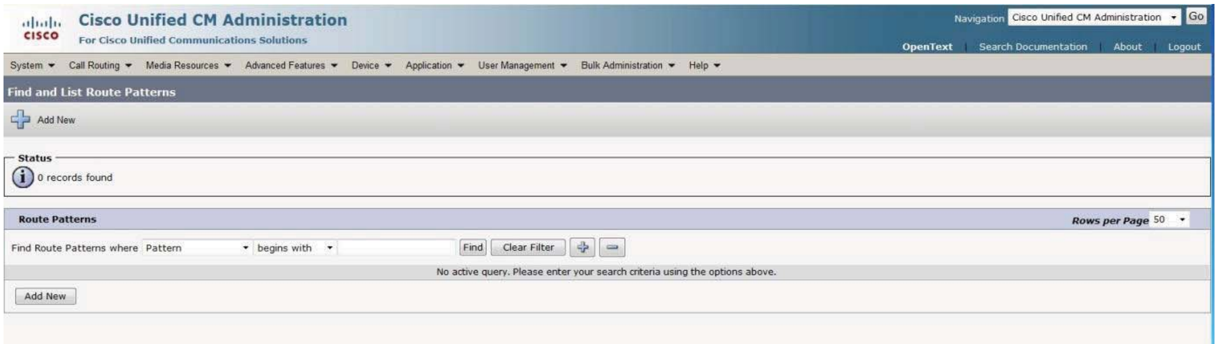
1. Using a web browser, log into the Cisco Unified CM Administration screen.\



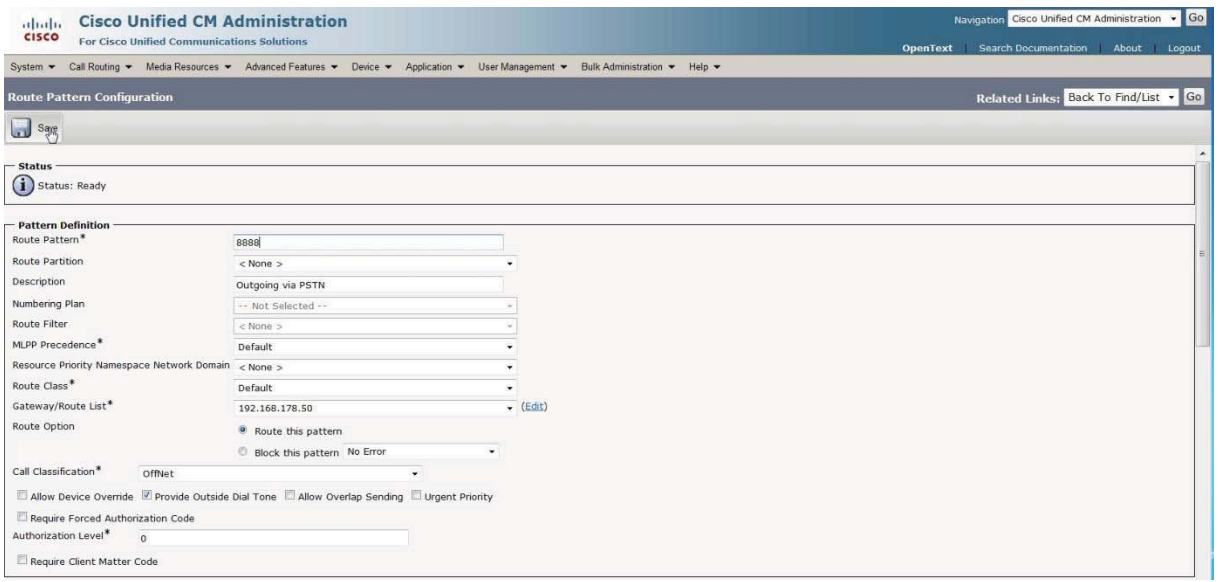
2. From the menu select Call Routing | Route / Hunt | Route Pattern.



3. Click on **Add New** to add a new **Route**



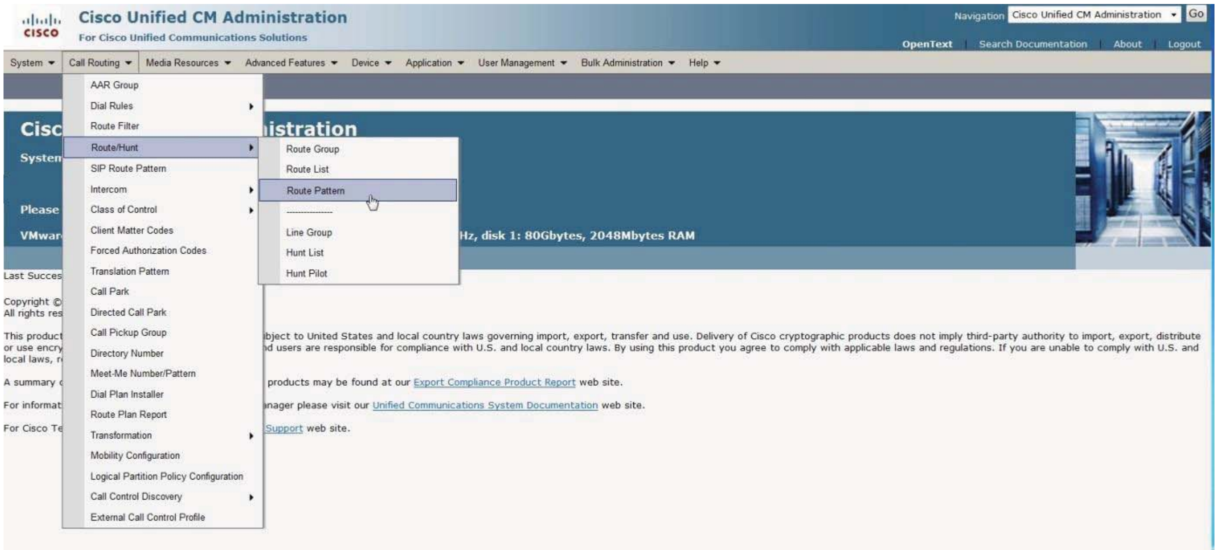
4. Route pattern "8888" is the format to send the fax via the T1/E1 (PSTN)



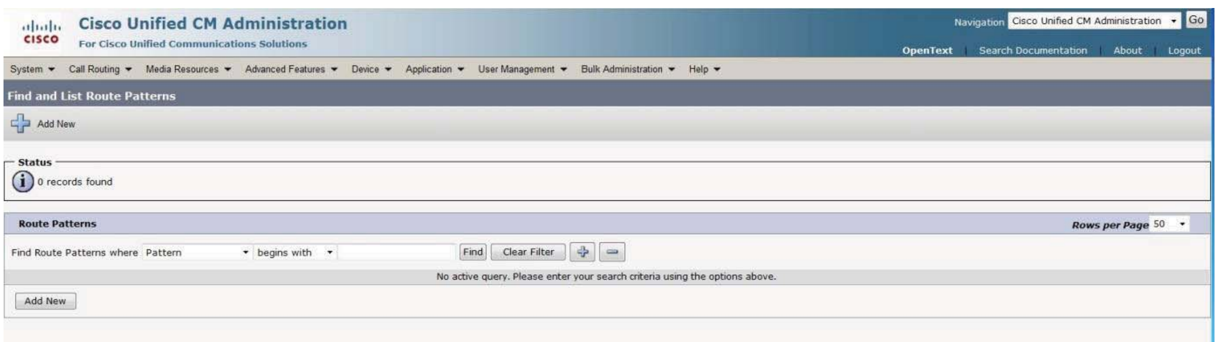
5. In the **Gateway/Route List,** enter the IP address (192.168.178.50) of the Voice Gateway that sends out the Fax call.

Configure Call Routing (From PSTN to OpenText RightFax)

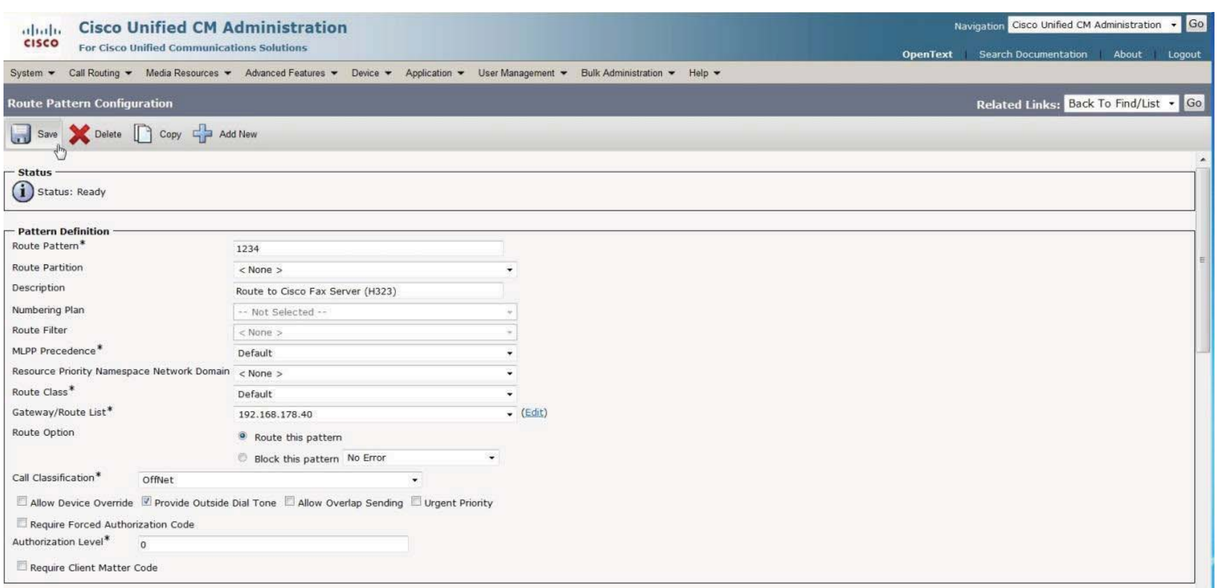
1. Using a web browser, log into the Cisco Unified CM Administration screen.



2. Select **Call Routing | Route Hunt | Route Pattern**.



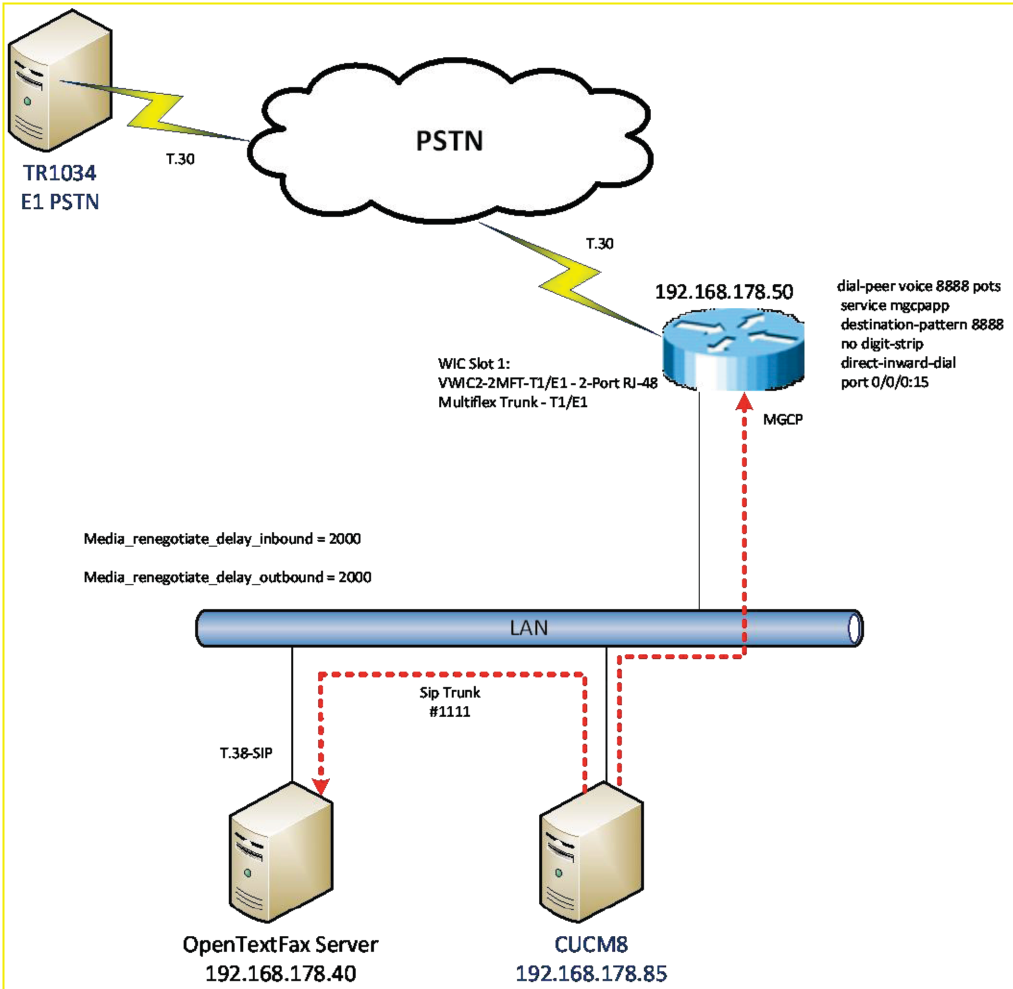
3. Click on **Add New** to add a new **Route Pattern.**



4. “1234” in **Route Pattern** is used to send faxes from PSTN to OpenText RightFax thru CUCM.
5. In the **Gateway/Route List,** enter the IP address (192.168.178.40) of OpenText RightFax.

Scenario 3: SIP-to-MGCP Configuration

Network System Configuration – MGCP / SIP Configuration



Network Addresses

Device #	Device Make, Model, and Description	Device IP Address
1	OpenText RightFax	192.168.178.40
2	CUCM 8.5.10000-23	192.168.178.85
3	Cisco 2800 Integrated Service Router	192.168.178.50

Dialing Plan Overview

To call the OpenText RightFax (SR140) from a POTS phone, dial 1234. The call flow and protocol path behaves as follows:

- POTS (dial 1234—E1—>
- Gateway(dial 1234@192.168.178.85)—H.323—>
- CUCM8.5.10000-23(dial 1234@192.168.178.40)—H.323—>
- OpenText RightFax.

To call the POTS lines of the Gateway, dial 8888@192.168.178.83. The call flow and protocol path behaves as follows:

- OpenText RightFax(8888@192.168.178.85)—H.323—>
- CUCM8.5.10000-23(dial 8888@192.168.178.50)—H.323—>
- Gateway(dial 8888)—E1—>
- POTS

OpenText RightFax SR140 Setup Notes

In this scenario, Dialogic SR140 is required non-default values. For RightFax version 9.4 FP1 SR2 (Dialogic SDK 6.3.0 and later), the following parameters must be set under T.38 Parameters:

- Media Renegotiate Delay Inbound, msec = 2000
  - Callctrl.cfg value = Media_renegotiate_delay_inbound
- Media Renegotiate Delay Inbound, msec = 2000
  - Callctrl.cfg value = Media_renegotiate_delay_outbound

Dialogic® Brooktrout® TR1034 Fax PSTN Setup Notes

For the sample test configuration, the TR1034 was configured using the default values, consult the Dialogic® Brooktrout® Fax Products Installation and Configuration Guide for details.

Cisco 2800 Gateway Setup Notes

For the sample test configuration, the Cisco 2800 Gateway was configured using the Cisco IOS command-line interface. The specific items configured include:

- Enable T.38 support
- Configure line card interface
- Configure MGCP
- Configure Dial-Peers – POTS

Enable T.38 support

The following lines allow SIP calls and T.38 fax calls

```
voice service voip
 fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none
 SIP
```

Configure line card interface

```
controller E1 0/0/0
 clock source internal
 pri-group timeslots 1-8,16 service mgcp
```

Configure MGCP

When enabling MGCP, first configure the following basic router information:

- Hostname
- IP addressing
- Routing information

The next steps to configure MGCP are

- Enable MGCP
- Specify how to reach the call agent
- Specify that the call agent is a Cisco Communications Manager.

Enter the following commands in **Global Configuration Mode** to allow MGCP calls:

```
ccm-manager mgcp
```

!Note: The following command enables music on hold so off-net callers receive streaming music as multicast, rather than unicast:

```
 ccm-manager music-on-hold
ccm-manager config server 192.168.178.85
!
mgcp
mgcp call-agent 192.168.178.85 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp default-package fxr-package
!
mgcp profile default
```

Notes:

- 192.168.178.85 is the IP address of the CUCM.
- Verify that
  - **`mgcp fax t38 inhibit`** does not exist, as it disables T.38

Configure Dial-Peers – POTS

Next, you must bind MGCP to the voice ports:

- Configure a dial peer for each voice port
- Binding MGCP to it using the application `MGCPAPP` command. *Note: This command is case sensitive in some IOS releases. If you are unsure, use all capital letters.*

The following allows the phone “8888* to be dialed out through the POTS lines:

```
dial-peer voice 8888 pots
 service mgcpapp
 destination-pattern 8888
 no digit-strip
 direct-inward-dial
 port 0/0/0:15

interface Serial0/0/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5
 isdn protocol-emulate network
 isdn incoming-voice voice
 no cdp enable
```

CUCM 8.5 Setup Notes – MGCP / SIP Configuration

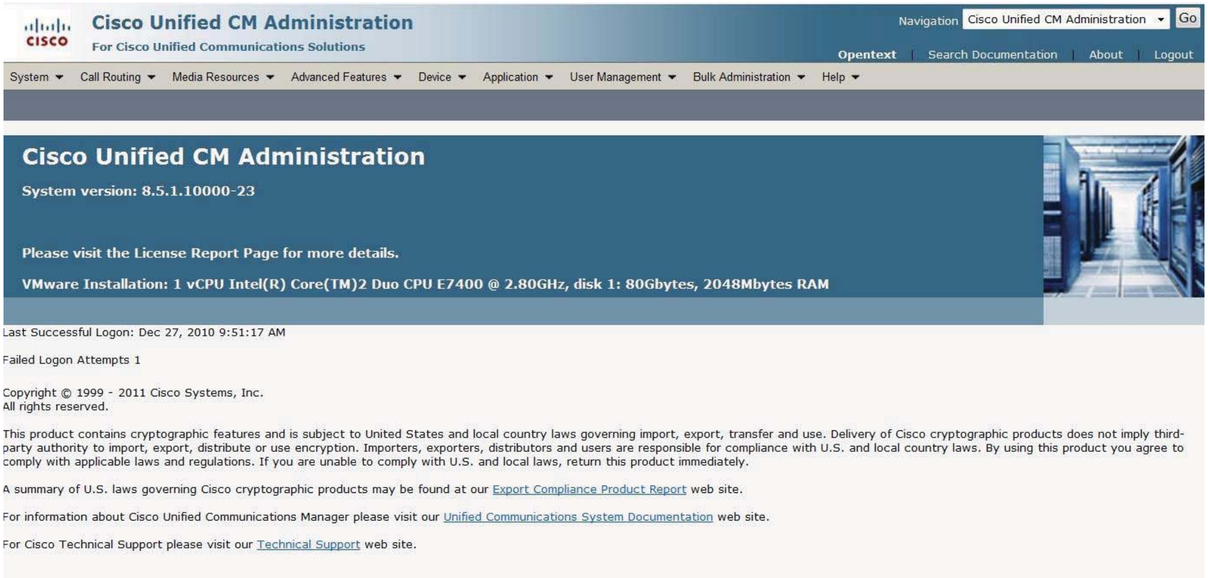
Configuration of CUCM 8.5 consists of the following steps:

- Configure SIP Trunk Security Profile
- Configure Sip Trunk from CUCM to OpenText RightFax
- Configure MGCP Gateway

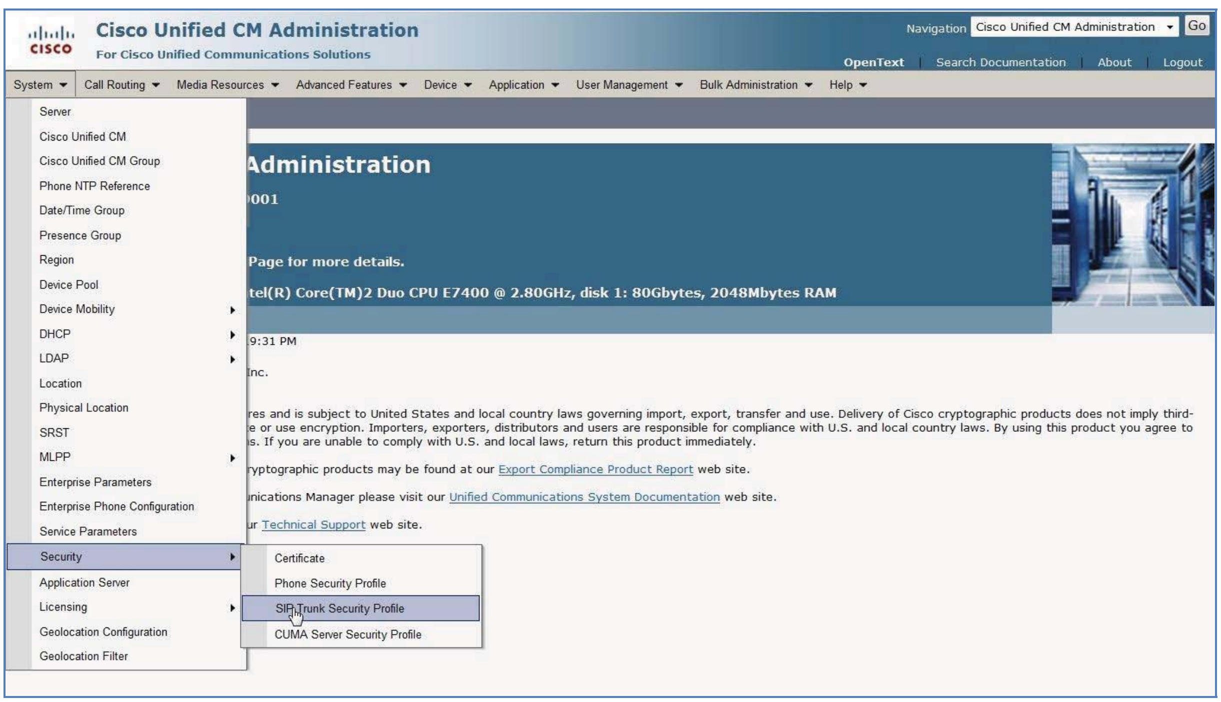
The following items are included at the end of the section:
- IOS overview
- Troubleshooting guidelines

Configure SIP Trunk Security Profile

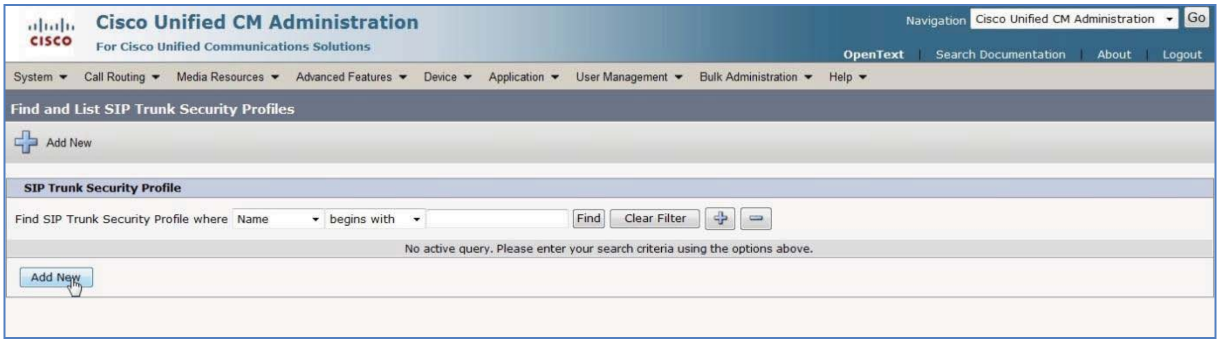
1. Using a web browser, log into the Cisco Unified CM Administration screen.



2. From the menu select **System | Security Profile | SIP Trunk Security Profile**.

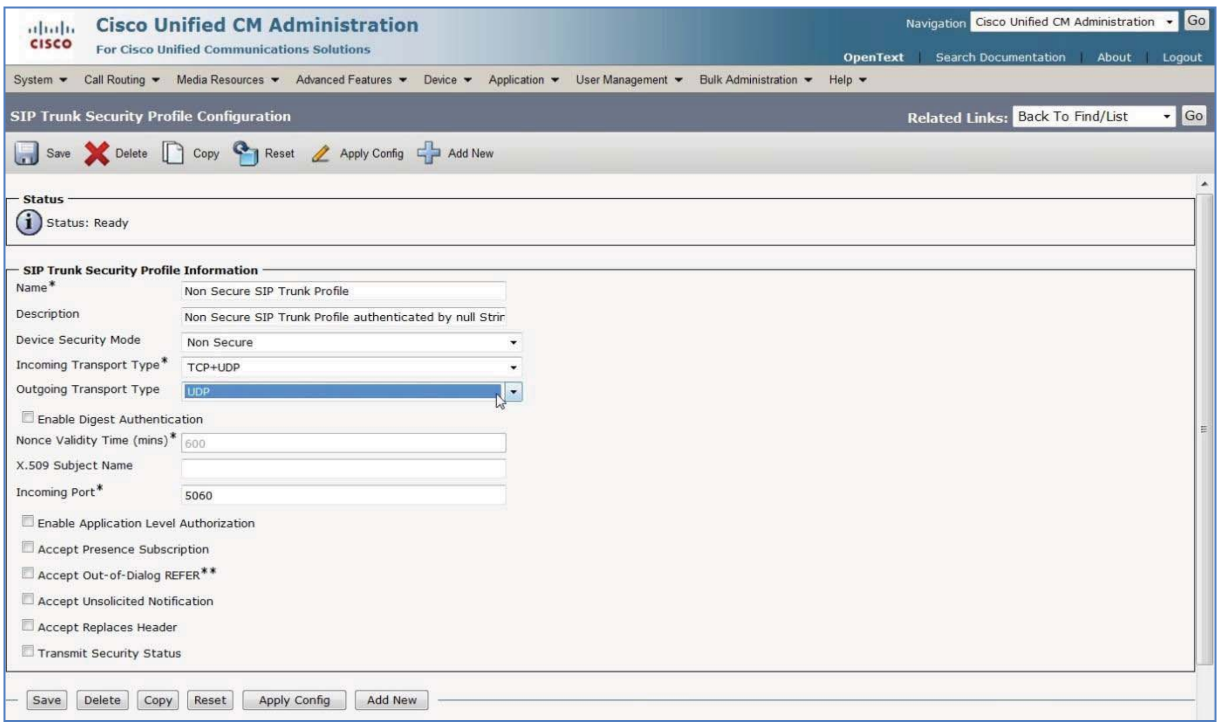


3. The following screen appears:



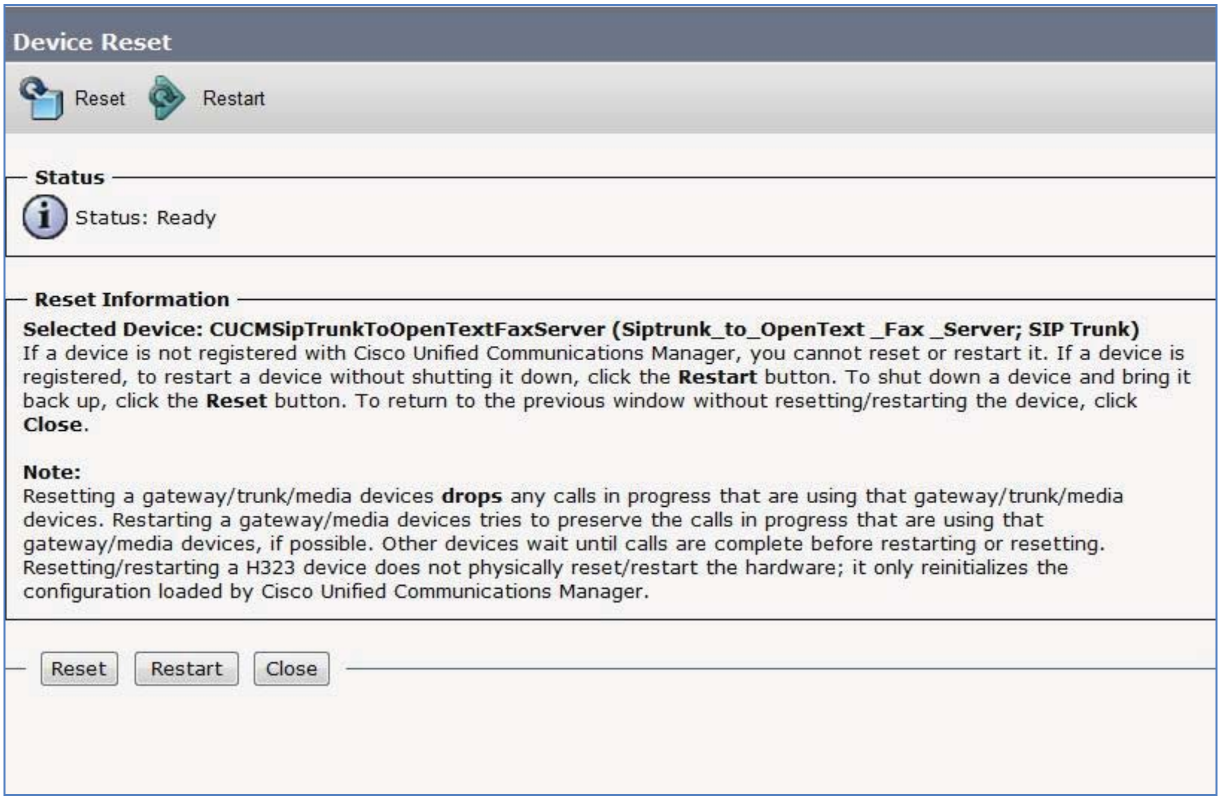
4. Click **Find** to edit an existing Sip Trunk Profile or click on **Add New** to add a new Sip Trunk Profile.

The following screen appears:



5. Change **Outgoing Transport Type** to UDP. *Note: UDP is required by OpenText RightFax.*

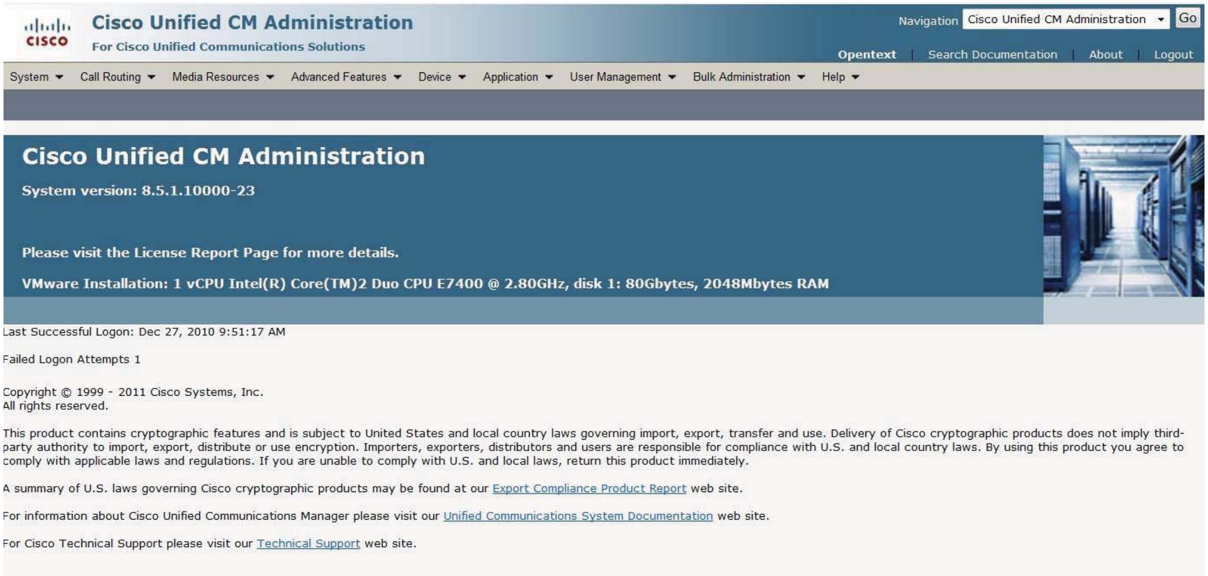
6. Press **Save**. The following screen appears:



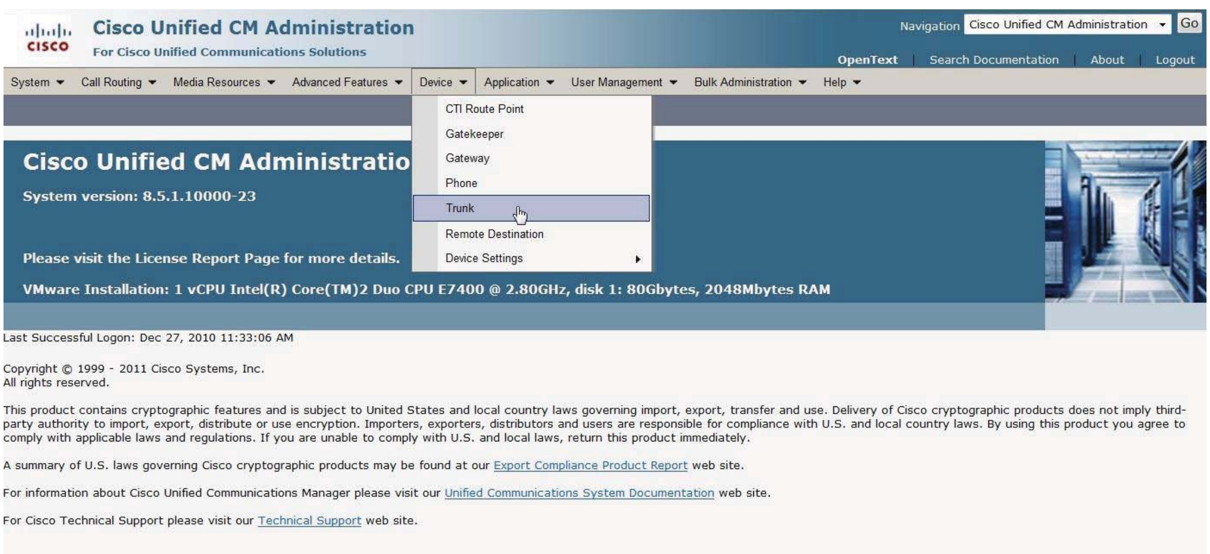
7. Press **Reset**, then press **Close.**

Configure the SIP Trunk from CUCM to OpenText RightFax

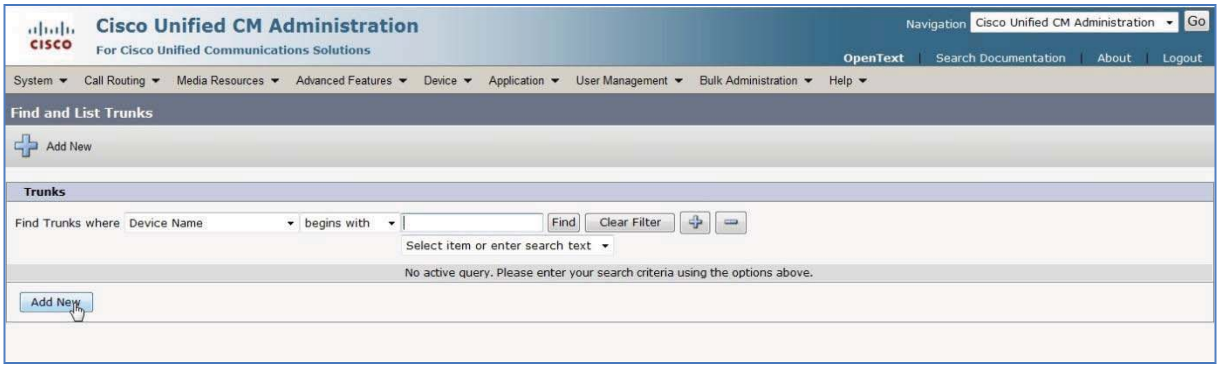
1. Using a web browser, log into the Cisco Unified CM Administration screen.



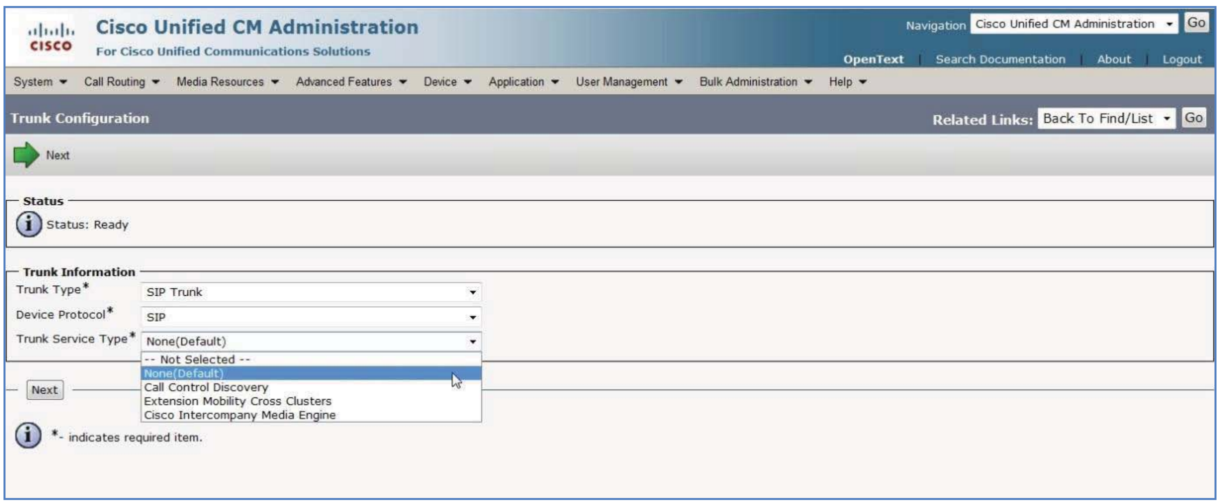
2. From the menu select **Device | Trunk.**



3. The following screen appears:

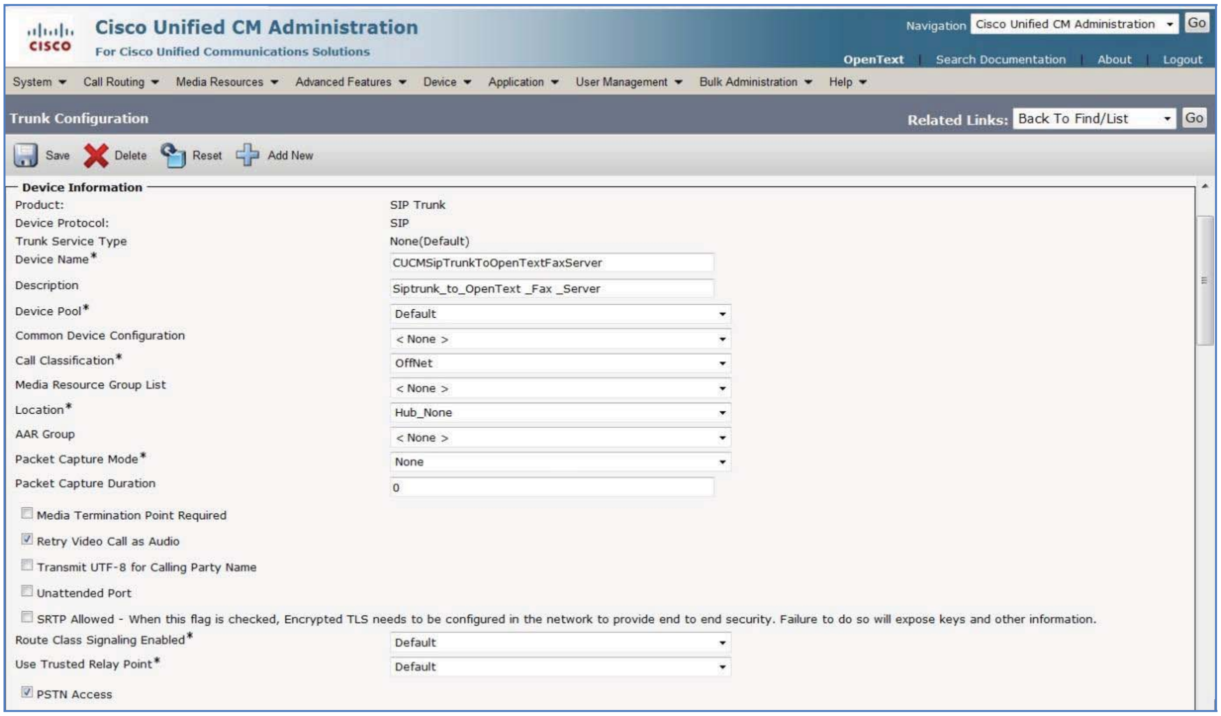


4. Press **Add New** to add a new SIP Trunk.

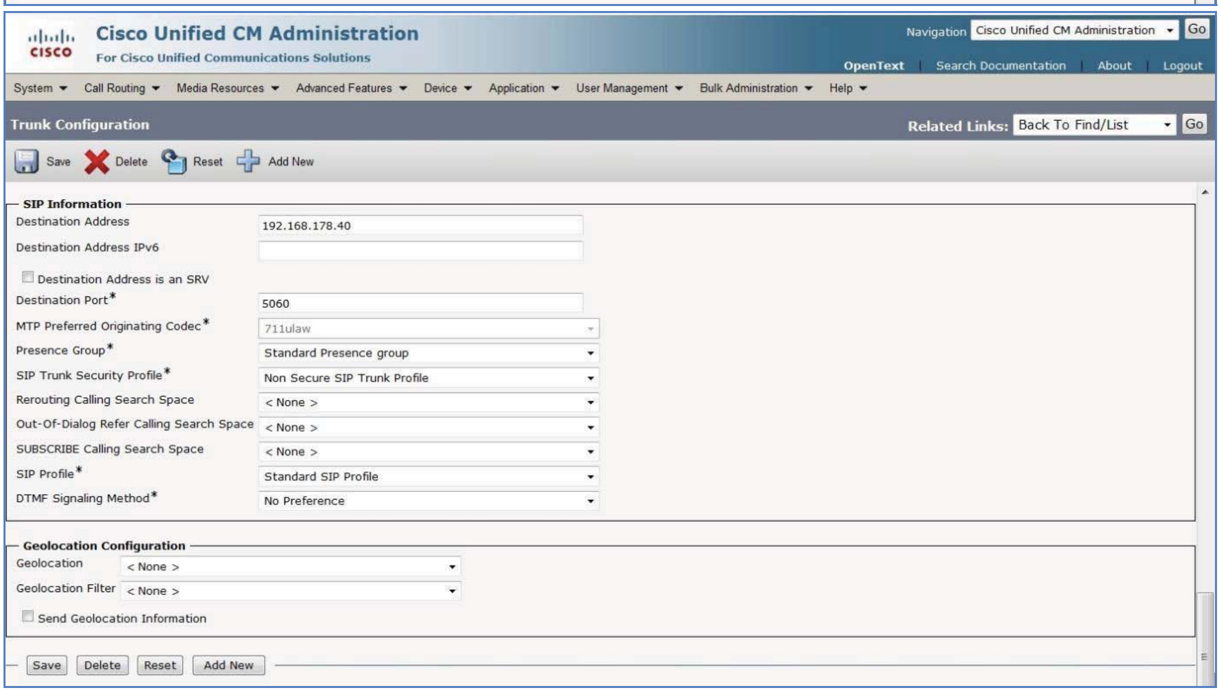


5. Select the following options and click **Next**:
    a. **Trunk Type** = SIP Trunk
    b. **Device Protocol** = SIP
    c. **Trunk Service Type** = None (Default)

6. The following screen appears:



7.

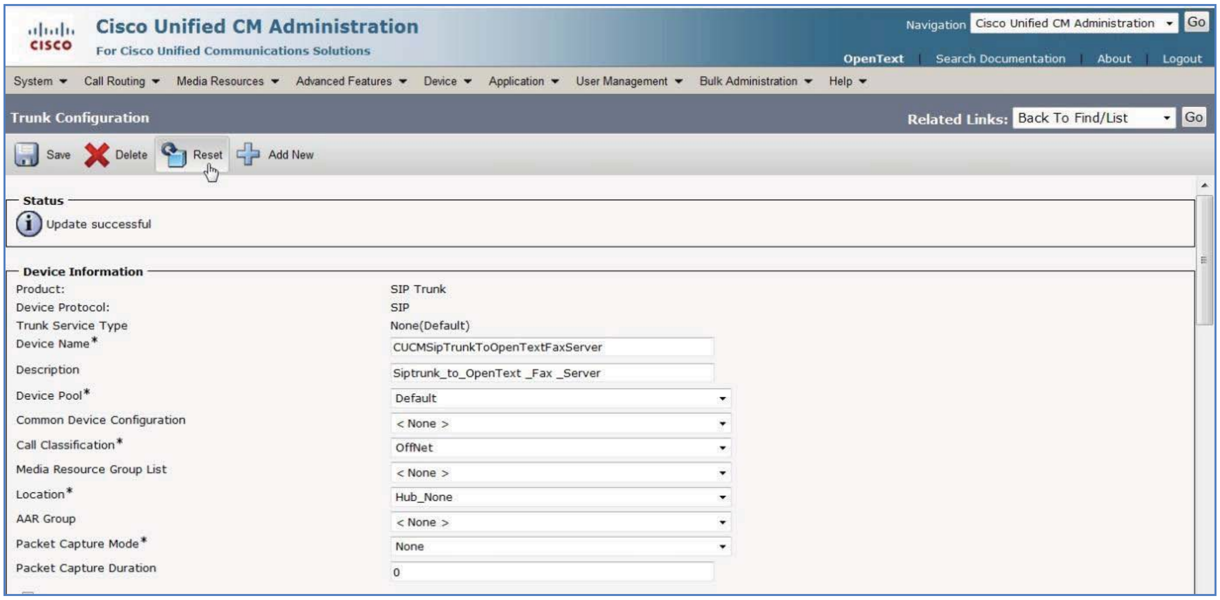


Set the following options:

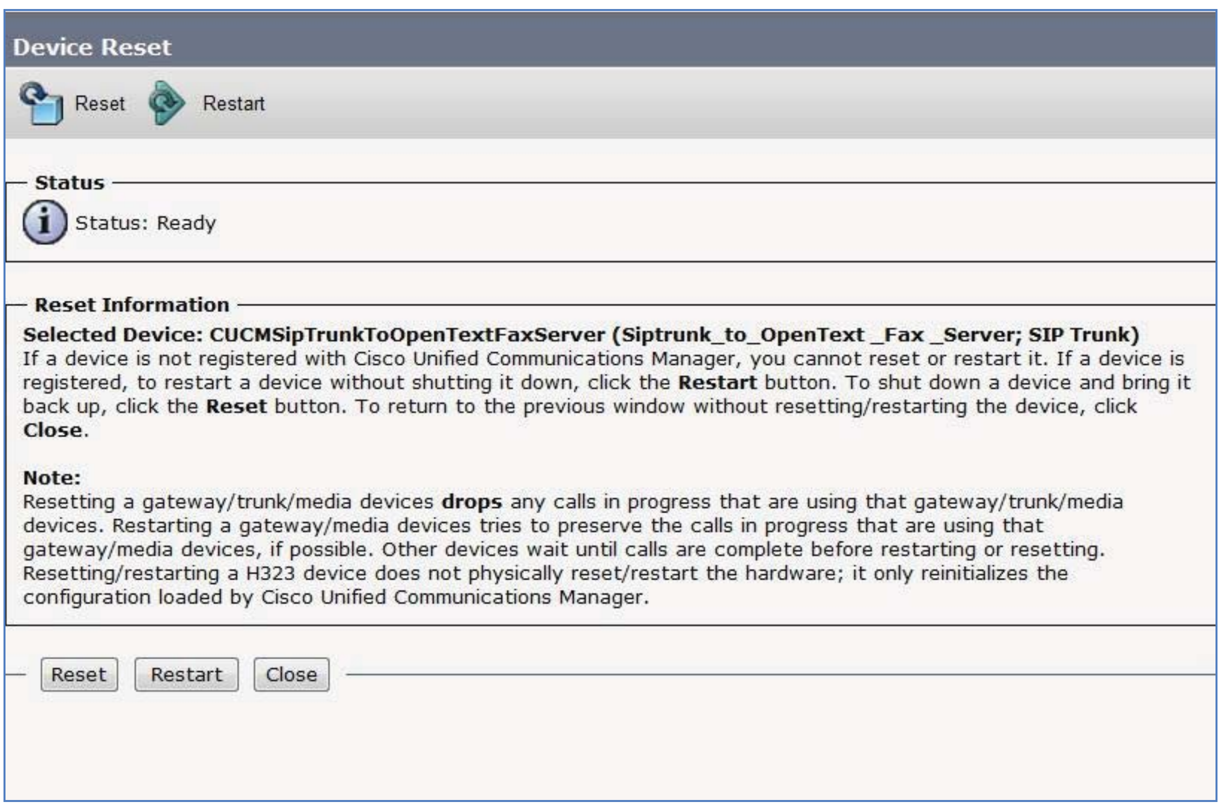
a. **Device Name**: CUCMSipTrunkToOpenTextFaxServer
b. **Device Description**: Siptrunk_to_OpenText _Fax _Server
c. **Device Pool**: Default
d. **Call Classification**: OffNet
e. **Destination Address**: 192.168.178.40 (address of OpenText RightFax)
f. **SIP Trunk Security Profile**: Non Secure SIP Trunk Profile
g. **SIP Profile**: Standard SIP Profile

8. Press **Save.**

9. Press
Reset.

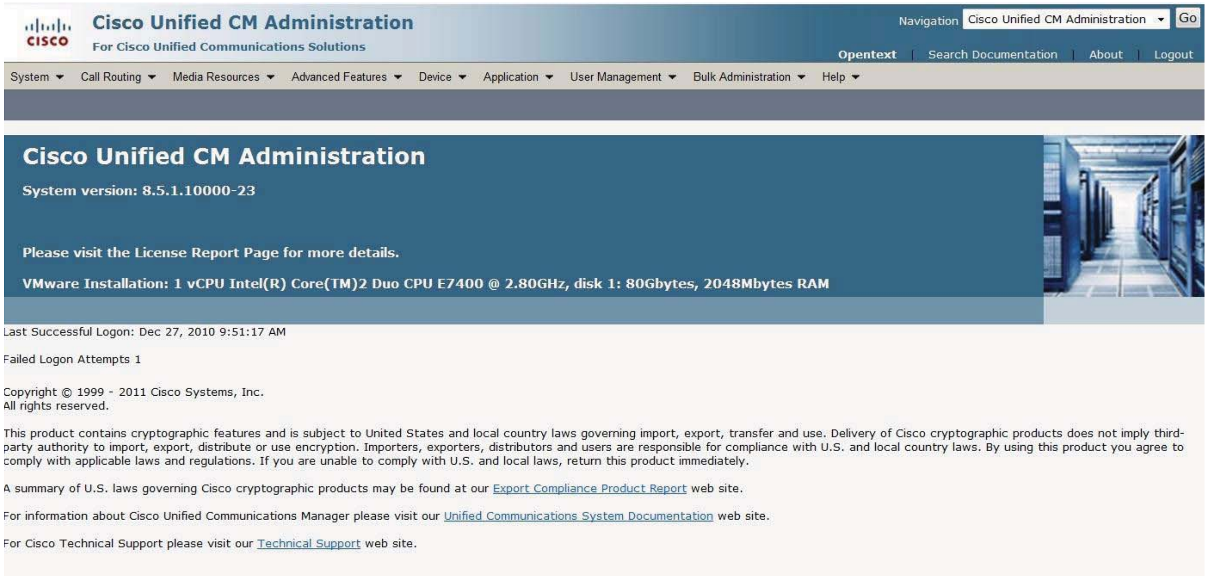


10. Press **Restart** then press **Close**.

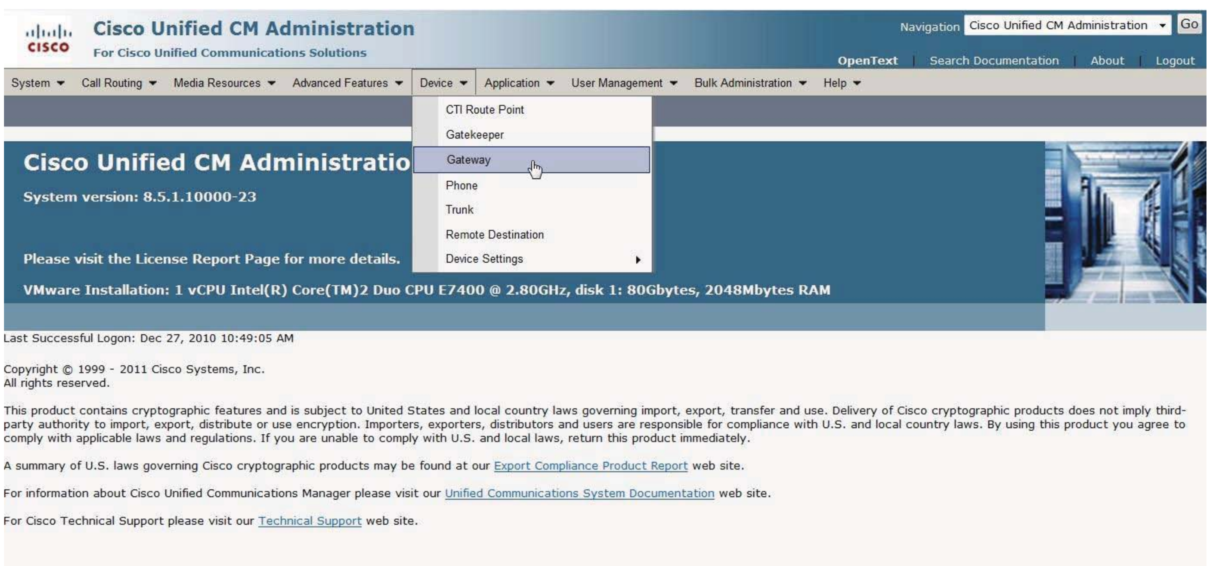


Configure MGCP Gateway

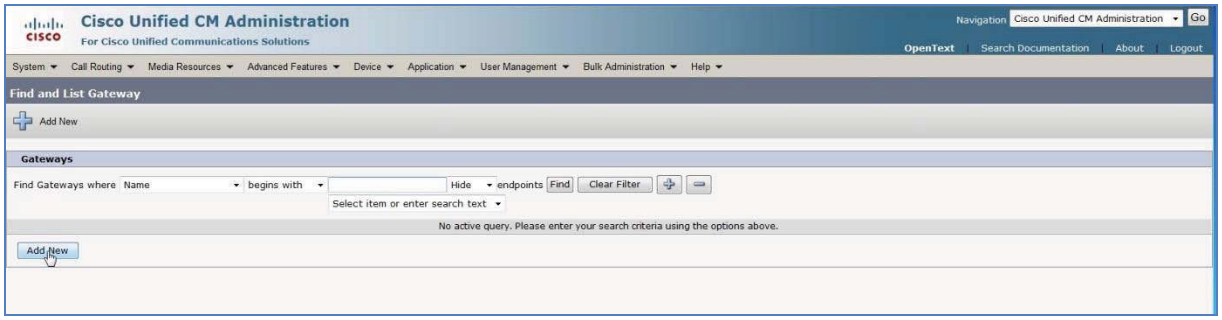
1. Using a web browser, log into the Cisco Unified CM Administration screen.



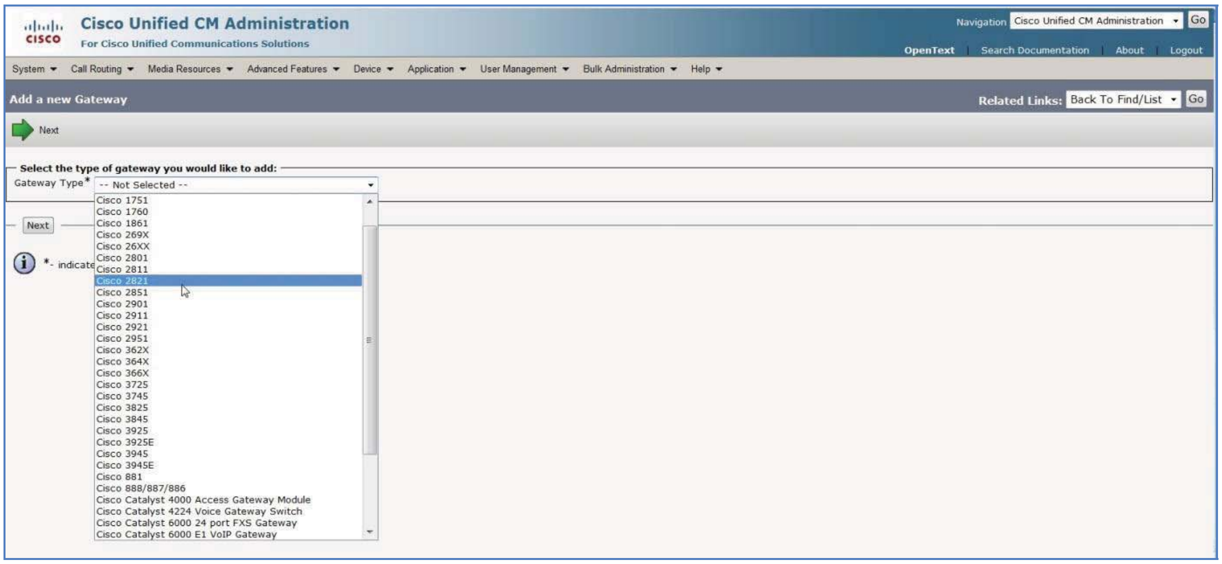
2. From the menu select **Device | Gateway**



3. Press **Add New** to add a new Gateway.

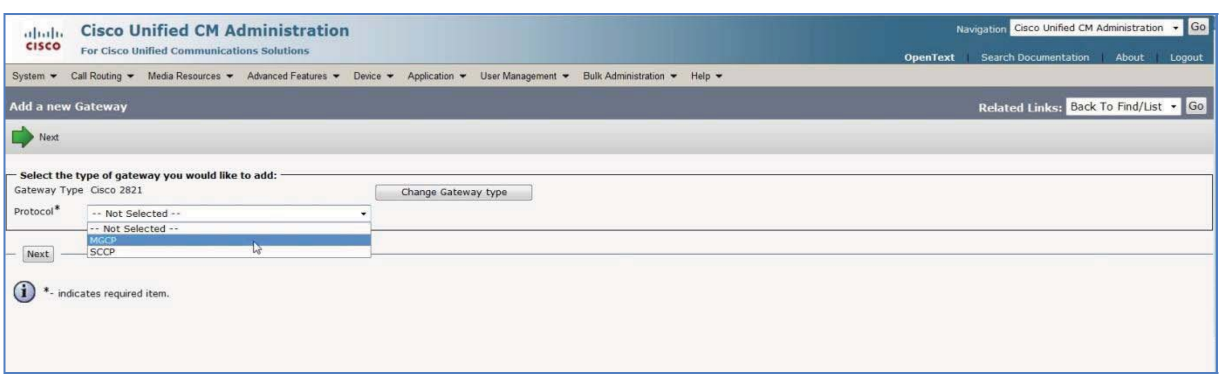


4. The following screen appears:

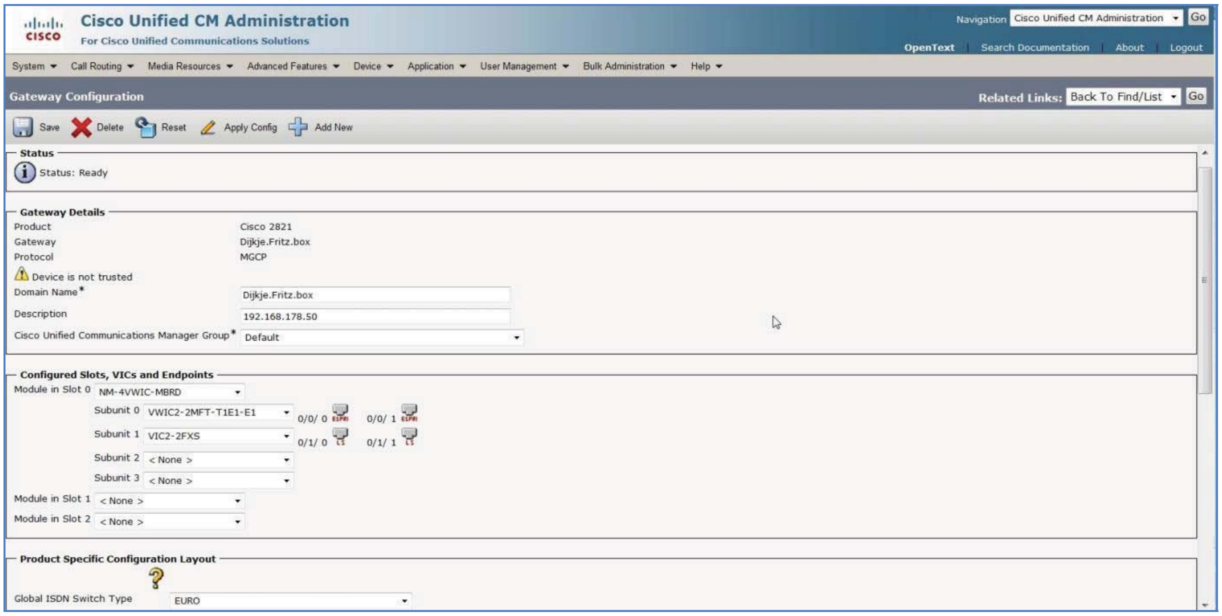


5. Select the **Gateway Type**. For MGCP gateways, choose the device type (router model or voice gateway). In this example, a Cisco 2821 router was selected. **Note:** You cannot configure Communication Manager to recognize the same device as both an MGCP and an H.323 gateway.

6. Next, set **Protocol to** MGCP and click **Next.**



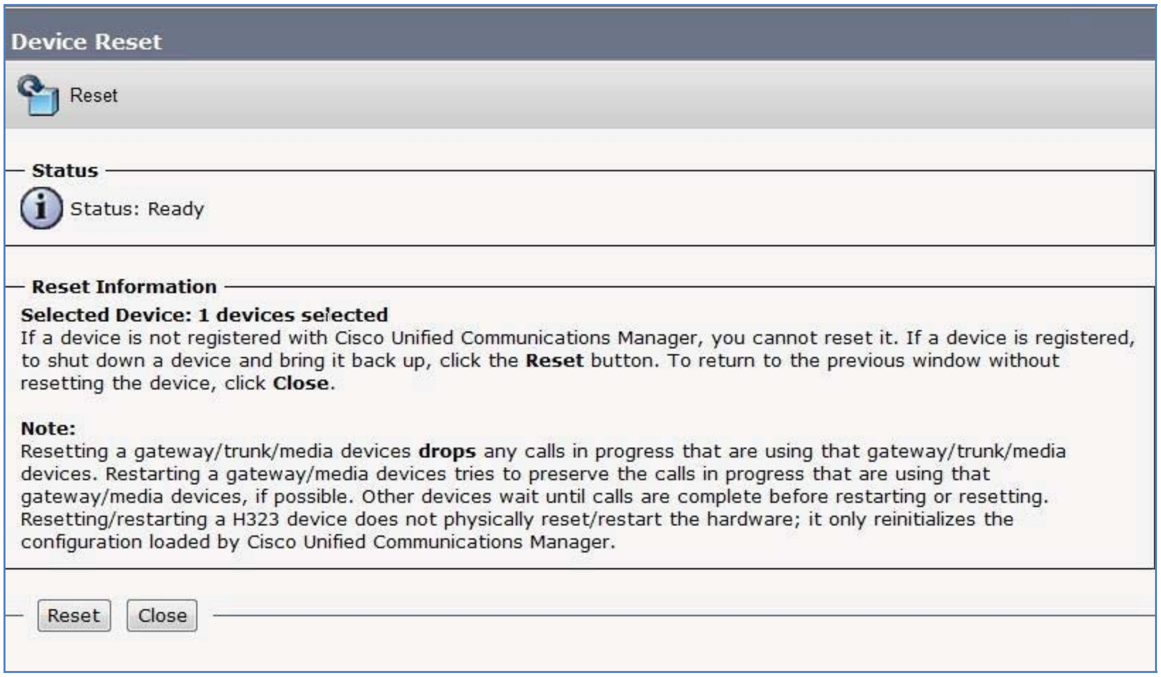
7. The **Gateway Configuration** screen appears:



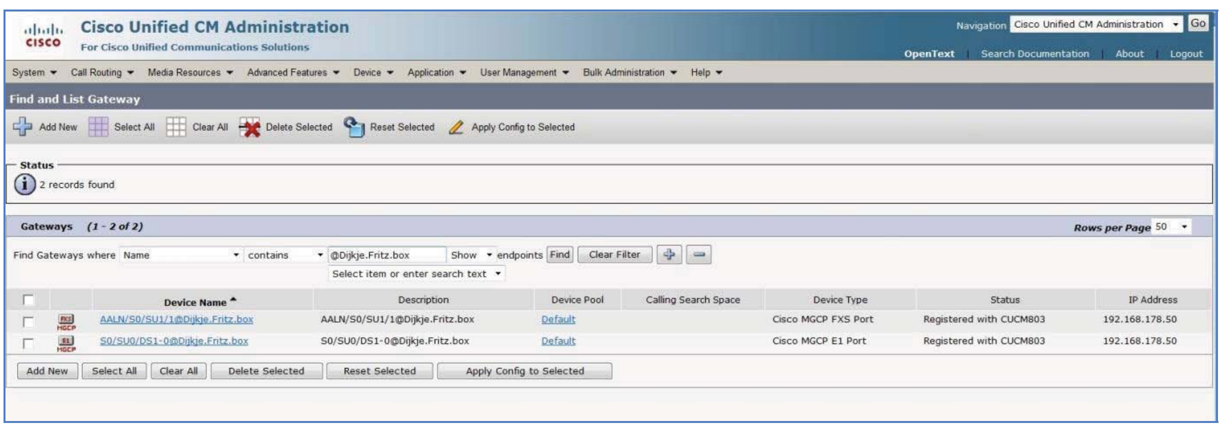
8. Under **Gateway Details,** enter the following information:
    a. **Domain Name:** Enter hostname of the router. ***Important information:***
        i. MGCP gateways are identified by *hostname*, not *IP address*.
        ii. If the router is configured with a domain name, append it to the hostname, such as Dijkje.Fritz.box.
        iii. The name is case sensitive.
    b. **Description (optional):** Enter optional description string.
    c. **Cisco Unified Communications Manager Group (required):** Choose a group, or set as Default.

9. Under **Configured Slots, VICs and Endpoints,** begin configuring endpoints.
    a. Available router slots are listed, with drop-down menu to select voice module type they contain, if any.
    b. ISR routers contain four WIC/VWIC slots that are not part of a separate module. These are listed in the drop-down menu as **"NM-4VWIC-MBRD."** Choose this option, as shown in the example, if you intend to use these slots.

10. On the next screen, reset the gateway by clicking **Reset** then click **Close**. *Note: Resetting the MGCP gateway drops all in-process calls on the gateway.*



11. To verify that the gateway is registered, go to the **Find and List Gateways** screen. Click **Find.** The gateway should be listed along with registered endpoints.

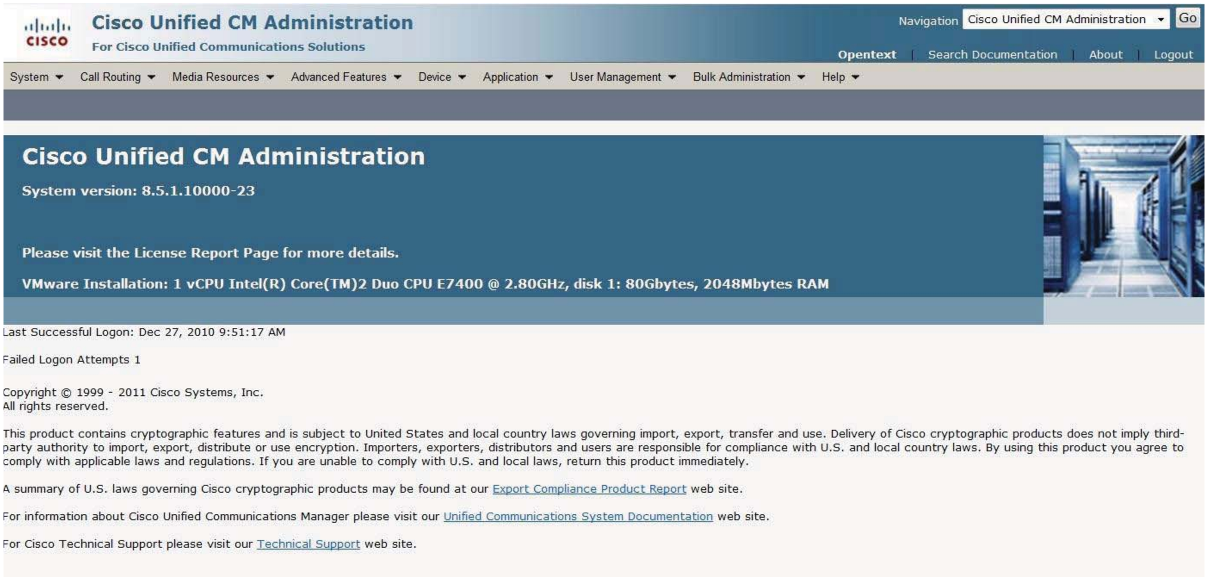


Ensure the Gateway is under MGCP control of CUCM803(c)

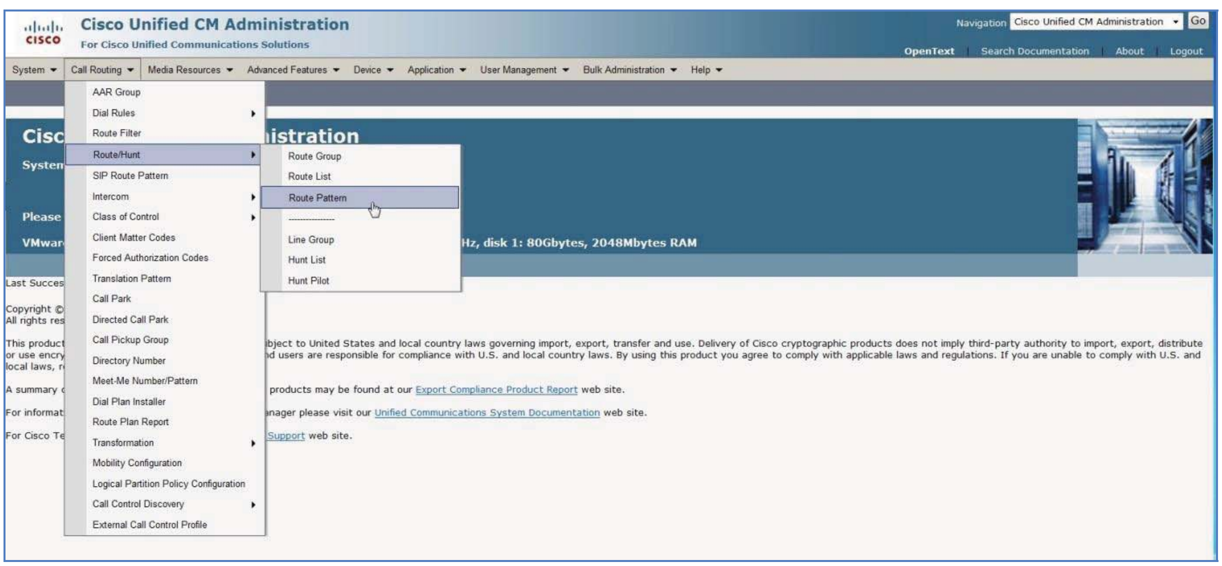
```
Dijkje#SH CCM
MGCP Domain Name: Dijkje.fritz.box
Priority        Status                   Host
============================================================
Primary         Registered               192.168.178.85
First Backup    None
Second Backup   None

Current active Call Manager:    192.168.178.85
Backhaul/Redundant link port:   2428
Failover Interval:              30 seconds
Keepalive Interval:             15 seconds
Last keepalive sent:            15:55:33 PCTime Sep 9 2010 (elapsed time:
00:00:04)
Last MGCP traffic time:         15:55:33 PCTime Sep 9 2010 (elapsed time:
00:00:04)
Last failover time:             None
Last switchback time:           None
Switchback mode:                Graceful
MGCP Fallback mode:             Not Selected
Last MGCP Fallback start time:  None
Last MGCP Fallback end time:    None
MGCP Download Tones:            Disabled
TFTP retry count to shut Ports: 2

Backhaul Link info:
    Link Protocol:      TCP
    Remote Port Number: 2428
    Remote IP Address:  192.168.178.85
    Current Link State: OPEN
    Statistics:
        Packets recvd:   2
        Recv failures:   0
        Packets xmitted: 2
        Xmit failures:   0
    PRI Ports being backhauled:
        Slot 0, VIC 0, port 0
```

Configure Call Routing (OpenText RightFax to PSTN)

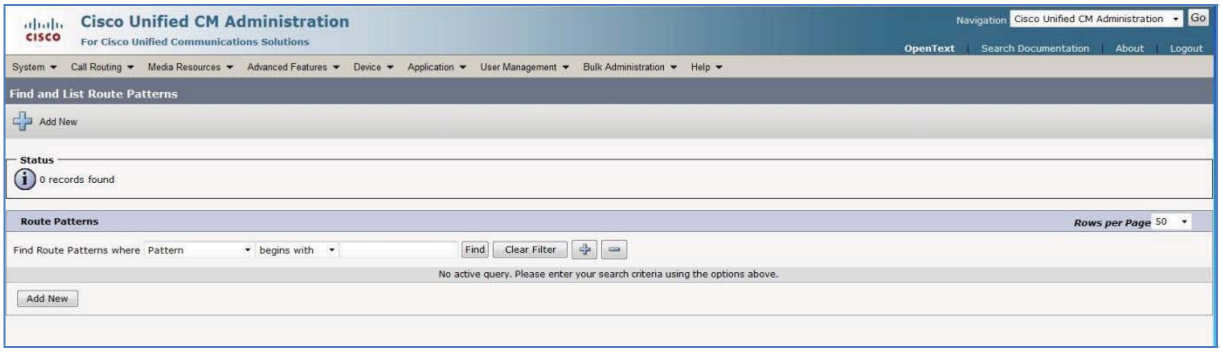
1. Using a web browser, log into the **Cisco Unified CM Administration** screen.



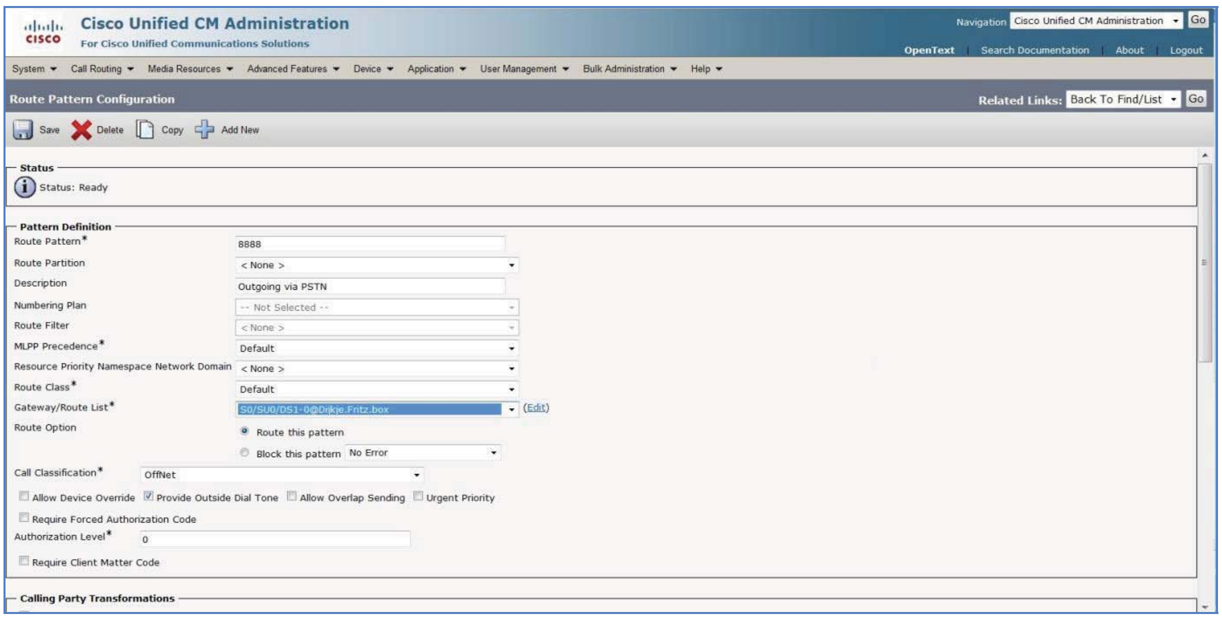
2. From the menu select Call Routing | Route / Hunt | Route Pattern.



3. Click **Add New** to add a new Route Pattern



4. The following screen appears:

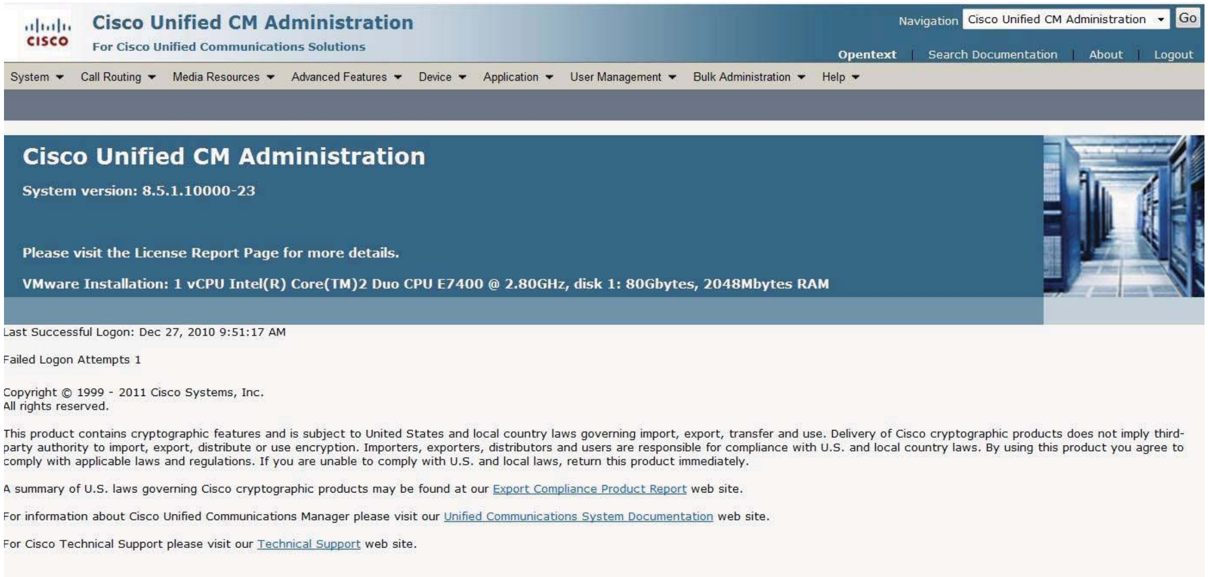


5. Set **Route Pattern** to “8888” to send faxes via the E1 (PSTN).

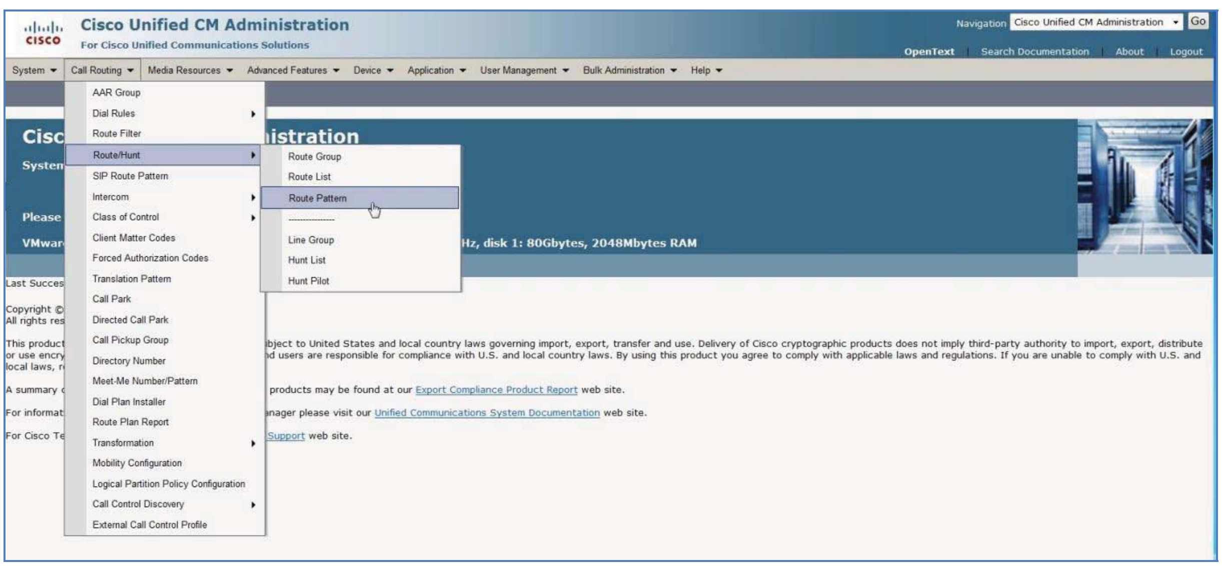
6. In this scenario, **Gateway/Route List** is S0/SUO/DS1-0@Dijkje.Fritz.box (the MGCP Trunk of the Gateway).

Configure Call Routing (PSTN to OpenText RightFax)

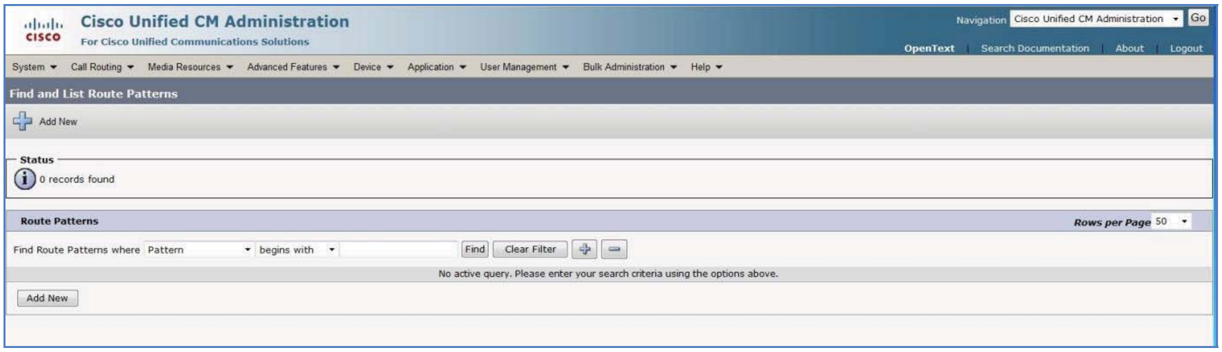
1. Using a web browser, log into the **Cisco Unified CM Administration** screen.



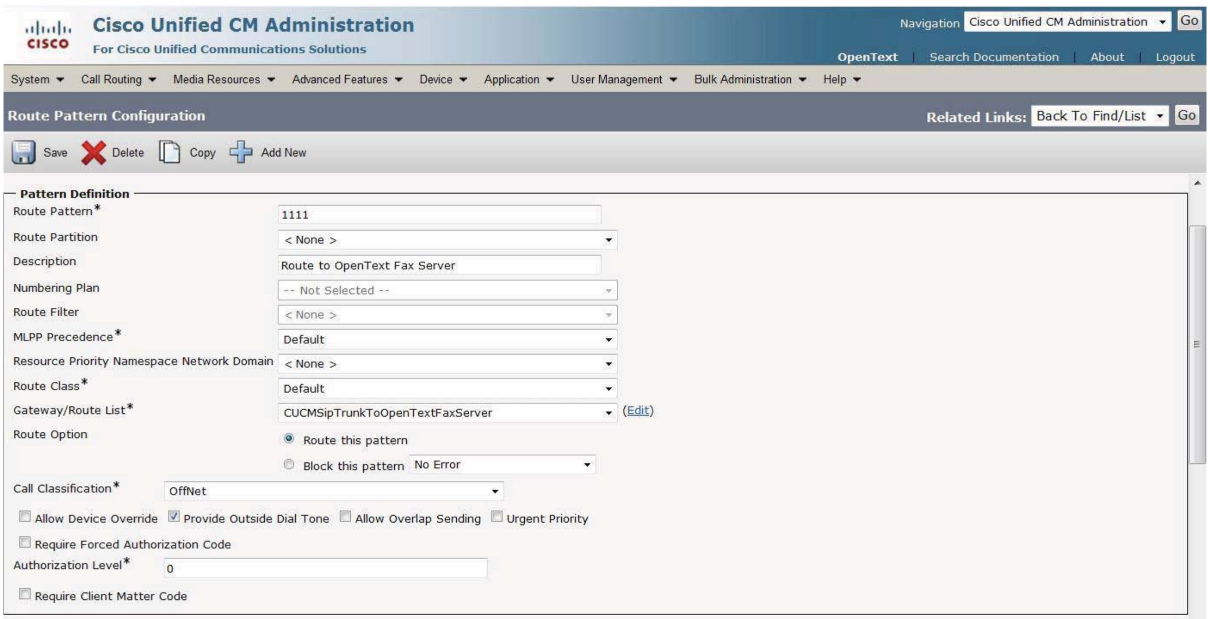
2. Select **Call Routing | Route Hunt | Route Pattern**.



3. Click on **Add New** to add a new Route Pattern



4. The following screen appears:



5. Set the following options:

   a. **Route Pattern:** "1111" (where faxes can be sent from the PSTN to OpenText RightFax via the CUCM).
   b. **Gateway/Route List**: Enter the Sip trunk created to OpenText RightFax

6. Click **Save** to save the configuration changes.

IOS overview

```
hostname Dijkje
!
no aaa new-model
clock timezone PCTime 1
network-clock-participate wic 0
no network-clock-participate aim 0
!
!
ip cef
!
!
ip domain name fritz.box
ip name-server 192.168.178.1
ip auth-proxy max-nodata-conns 3
ip admission max-nodata-conns 3
!
isdn switch-type primary-net5
!
voice-card 0
 dspfarm
!
!
voice service voip
 fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none
  sip
!
!
voice class codec 1
 codec preference 1 g711alaw
!
!
controller E1 0/0/0
 clock source internal
 pri-group timeslots 1-8,16 service mgcp
!

interface GigabitEthernet0/0
 ip ddns update dijkje
 ip address 192.168.178.50 255.255.255.0
 duplex half
 speed auto
 no keepalive
 no mop enabled
!
interface Serial0/0/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5
 isdn protocol-emulate network
 isdn incoming-voice voice
 isdn bind-l3 ccm-manager
 no cdp enable
!
interface Serial0/3/0
 no ip address
 shutdown
```

```
 clock rate 2000000
!
no ip forward-protocol nd
!
!
ip http server
ip http authentication local
ip http secure-server
!
!
!
!
control-plane
!
!
!
voice-port 0/0/0:15
!
voice-port 0/1/0
 compand-type a-law
 cptone NL
 shutdown
 description fxo00
 bearer-cap Speech
!
voice-port 0/1/1
 compand-type a-law
 cptone NL
 description FX01
 bearer-cap Speech
!
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 192.168.178.85
!
mgcp
mgcp call-agent 192.168.178.85 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp default-package fxr-package
!
mgcp profile default
!
!
dial-peer voice 1000 pots
 huntstop
 service mgcpapp
 answer-address 1000
 destination-pattern 1000
 no digit-strip
 direct-inward-dial
 port 0/1/0
!
dial-peer voice 8888 pots
 service mgcpapp
 destination-pattern 8888
 no digit-strip
 direct-inward-dial
!
```

```
gateway
 timer receive-rtp 1200
!
sip-ua
scheduler allocate 20000 1000
!
end
```

Troubleshooting guidelines

The following suggestions may assist in troubleshooting issues that arise:

- Reset the MGCP statistical counters with the **`clear mgcp statistics`** command.
- If no RTP traffic is getting through make sure **IP routing** is enabled.
- Use the **`show rtp statistics`** command, then turn on the **`debug ip udp`** command and track down the MGCP RTP packets.

```
Dijkje# show rtp statistics
RTP Statistics info:
No. CallId Xmit-pkts Xmit-bytes Rcvd-pkts  Rcvd-bytes Lost pkts  Jitter Latenc
1   17492  0x8A      0x5640     0x8A       0x5640     0x0        0x0      0x0
Dijkje# show rtp statistics
RTP Statistics info:
No. CallId Xmit-pkts Xmit-bytes Rcvd-pkts  Rcvd-bytes Lost pkts  Jitter Latenc
1   17492  0xDA      0x8840     0xDB       0x88E0     0x0        0x160    0x0
```

- If an RSIP message is not received by the call agent, make sure the **`mgcp call-agent`** command or the MGCP profile **`call-agent`** command is configured with the correct call agent name (or IP address) and UDP port number. Use the **`show mgcp`** command or the **`show mgcp profile`** command to display this information:

```
Dijkje# show mgcp
MGCP Admin State ACTIVE, Oper State ACTIVE - Cause Code NONE
MGCP call-agent: 192.168.178.85  Initial protocol service is MGCP, v. 1.0
...
MGCP gateway port: 2727, MGCP maximum waiting delay 3000
...
Dijkje# show mgcp profile
MGCP Profile default
Description: None
Call-agent: 192.168.178.85 2427 Initial protocol service is MGCP 0.1
Tsmax timeout is 20 sec, Tdinit timeout is 15 sec
Tdmin timeout is 15 sec, Tdmax timeout is 600 sec
Tcrit timeout is 4 sec, Tpar timeout is 16 sec
Thist timeout is 30 sec, MWI timeout is 16 sec
Ringback tone timeout is 180 sec, Ringback tone on connection timeout is 180 sec
Network congestion tone timeout is 180 sec, Busy tone timeout is 30 sec
Network busy tone timeout is 0 sec
Dial tone timeout is 16 sec, Stutter dial tone timeout is 16 sec
Ringing tone timeout is 180 sec, Distinctive ringing tone timeout is 180 sec
Continuity1 tone timeout is 3 sec, Continuity2 tone timeout is 3 sec
```

```
Reorder tone timeout is 30 sec,  Persistent package is ms-package
Max1 DNS lookup: ENABLED, Max1 retries is 5
Max2 DNS lookup: ENABLED, Max2 retries is 7
Source Interface: NONE...
```

- To verify connections and endpoints, use the **show mgcp** command:

```
Dijkje# show mgcp connection
Endpoint  Call_ID(C) Conn_ID(I) (P)ort (M)ode (S)tate (C)odec (E)vent[SIFL]
(R)esult[EA]
1. S0/DS1-1/5      C=F123AB,5,6  I=0x3  P=16506,16602  M=3  S=4  C=1 E=2,0,0,2
R=0,0
2. S0/DS1-1/6      C=F123AB,7,8  I=0x4  P=16602,16506  M=3  S=4  C=1 E=0,0,0,0
R=0,0
Dijkje# show mgcp endpoint
Interface E1 0/0/0

             ENDPOINT-NAME    V-PORT     SIG-TYPE   ADMIN
     S0/SU0/ds1-0/1@Dijkje   0/0/0:15        none   up
     S0/SU0/ds1-0/2@Dijkje   0/0/0:15        none   up
     S0/SU0/ds1-0/3@Dijkje   0/0/0:15        none   up
     S0/SU0/ds1-0/4@Dijkje   0/0/0:15        none   up
     S0/SU0/ds1-0/5@Dijkje   0/0/0:15        none   up
     S0/SU0/ds1-0/6@Dijkje   0/0/0:15        none   up
     S0/SU0/ds1-0/7@Dijkje   0/0/0:15        none   up
     S0/SU0/ds1-0/8@Dijkje   0/0/0:15        none   up


Interface E1 0/0/1

             ENDPOINT-NAME    V-PORT     SIG-TYPE   ADMIN
```

- If an MGCP message is rejected, it may be because the remote media gateway does not support SDP mandatory parameters (the *o*=, *s*=, and *t*= lines). If this is the case, configure the **mgcp sdp simple** command to send SDP messages without those parameters.
- If there are problems with voice quality, make sure that **cptone** (voice-port configuration) command is set for the correct country code.
- Capturing RTP packets from a sniffer may help isolate the problem. You may be able to decide such questions as whether the payload type or timestamps are set correctly.
- To check operation of interfaces, use the **show interface** command.
- To view information about activity on the T1 or E1 line, use the **show controllers** command. Alarms, line conditions, and other errors are displayed. The data

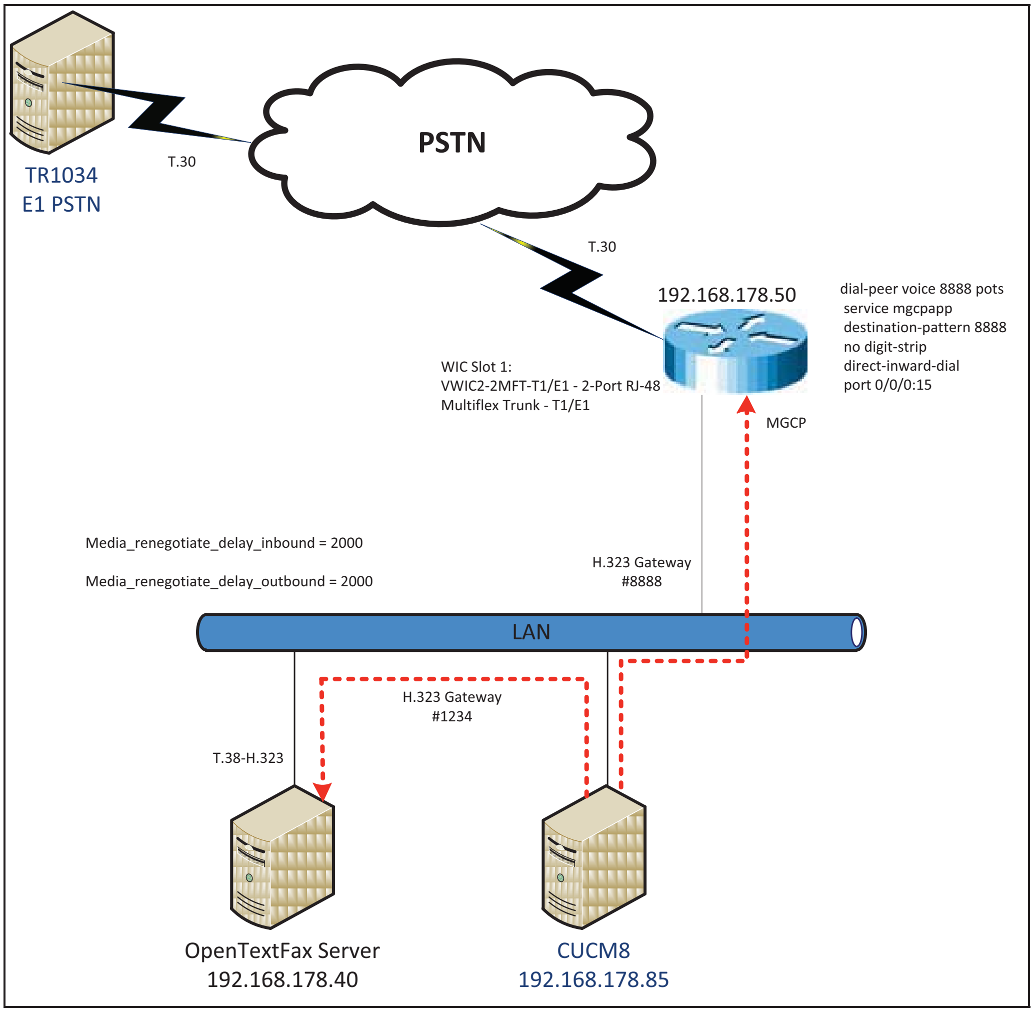
is updated every 10 seconds. Every 15 minutes, the cumulative data is stored and retained for 24 hours.

- When necessary, enable debug traces for errors, events, media, packets, and parser. The command **`debug mgcp packets`** can be used to monitor message flow in general. Note that there is always a performance penalty when using debug commands. The sample output below shows the use of the optional **`input-hex`** keyword to enable display of hexadecimal values.

```
Dijkje# debug mgcp {all | errors | events | packets {input-hex}| parser}
Dijkje# debug mgcp packets input-hex
Media Gateway Control Protocol input packets in hex value debugging is on
MGCP Packet received -
DLCX 49993 * MGCP 0.1
MGCP Packet received in hex -
44 4C 43 58 20 34 39 39 39 33 20 2A 20 4D 47 43 50 20 30 2E 31 A
send_mgcp_msg, MGCP Packet sent ---> </nowiki>
250 49993
```

Scenario 4: H.323-to-MGCP Configuration

Network System Configuration – MGCP / H.323 Configuration



Network Addresses

Device #	Device Make, Model, and Description	Device IP Address
1	OpenText RightFax	192.168.178.40
2	CUCM 8.5.10000-23	192.168.178.85
3	Cisco 2800 Integrated Service Router	192.168.178.50

Dialing Plan Overview

To call the SR140 from a POTS phone, dial 1234

- POTS (dial 1234—E1—>
- Gateway (dial 1234@192.168.178.85)—H.323—>
- CUCM8.5.10000-23 (dial 1234@192.168.178.40)—H.323—>
- OpenText RightFax.

To call the POTS lines of the Gateway, dial 8888@192.168.178.85

- OpenText RightFax (8888@192.168.178.85)—H.323—>
- CUCM8.5.10000-23 (dial 8888@192.168.178.50)—H.323—>
- Gateway (dial 8888)—E1—>
- POTS

OpenText RightFax SR140 Setup Notes

In this scenario, Dialogic SR140 is required non-default values. For RightFax version 9.4 FP1 SR2 (Dialogic SDK 6.3.0 and later), the following parameters must be set under T.38 Parameters:

- Media Renegotiate Delay Inbound, msec = 2000
  - Callctrl.cfg value = Media_renegotiate_delay_inbound
- Media Renegotiate Delay Inbound, msec = 2000
  - Callctrl.cfg value = Media_renegotiate_delay_outbound

Dialogic® Brooktrout® TR1034 Fax PSTN Setup Notes

For the sample test configuration, the TR1034 was configured using the default values, consult the Dialogic® Brooktrout® Fax Products Installation and Configuration Guide for details.

Cisco 2800 Gateway Setup Notes

For the sample test configuration, the Cisco 2800 Gateway was configured the Cisco IOS command-line Interface. The specific items configured include:

- Enable T.38 support
- Configure line card interface
- Configure MGCP
- Configure Dial-Peers – POTS

Enable T.38 support

The following lines allow H.323 and T.38 fax calls.

```
voice service voip
 fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none

h323
 session transport udp
 h245 tunnel disable
```

Note: `session transport` must contain `udp`.

Configure line card interface

```
controller E1 0/0/0
 clock source internal
 pri-group timeslots 1-8,16 service mgcp

interface Serial0/0/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5
 isdn protocol-emulate network
 isdn incoming-voice voice
 isdn bind-l3 ccm-manager
 no cdp enable
```

Configure MGCP

When enabling MGCP, first configure the following basic router information:

- Hostname
- IP addressing
- Routing information

The next steps to configure MGCP are:

- Enable MGCP
- Specify how to reach the call agent
- Specify that the call agent is a Cisco Communications Manager.

Enter the following commands in **Global Configuration Mode** to allow MGCP calls:

```
ccm-manager mgcp
```

!Note: The following command enables music on hold so off-net callers receive streaming music as multicast, rather than unicast:

```
 ccm-manager music-on-hold
ccm-manager config server 192.168.178.85
!
mgcp
mgcp call-agent 192.168.178.85 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp default-package fxr-package
!
mgcp profile default
```

Notes:

- 192.168.178.85 is the IP address of the CUCM.
- Verify that
    - **`mgcp fax t38 inhibit`** does not exist, as it disables T.38

Configure Dial-Peers – POTS

Next, you must bind MGCP to the voice ports:

- Configure a dial peer for each voice port
- Binding MGCP to it using the application `MGCPAPP` command. *Note: This command is case sensitive in some IOS releases. If you are unsure, use all capital letters.*

The following allows the phone “8888* to be dialed out through the POTS lines:

```
dial-peer voice 8888 pots
 service mgcpapp
 destination-pattern 8888
 no digit-strip
 direct-inward-dial
 port 0/0/0:15

interface Serial0/0/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5
 isdn protocol-emulate network
 isdn incoming-voice voice
 no cdp enable
```

CUCM 8.5 Setup Notes – MGCP / SIP Configuration

Configuration of CUCM 8.5 consists of the following steps:

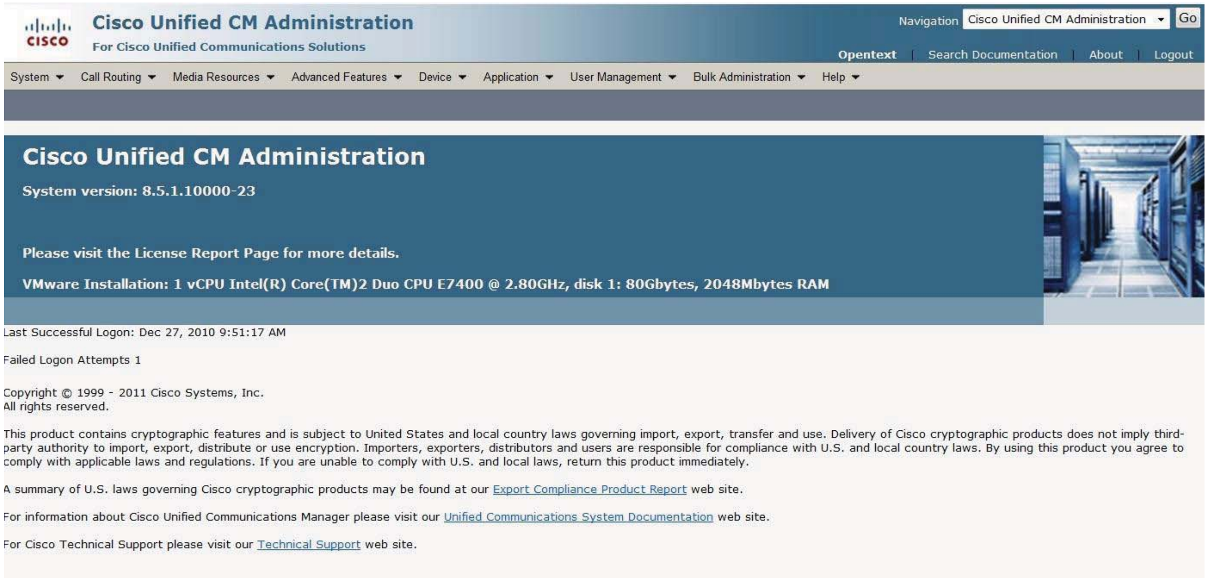
- Configure SIP Trunk Security Profile
- Configure Sip Trunk from CUCM to OpenText RightFax
- Configure MGCP Gateway

The following items are included at the end of the section:

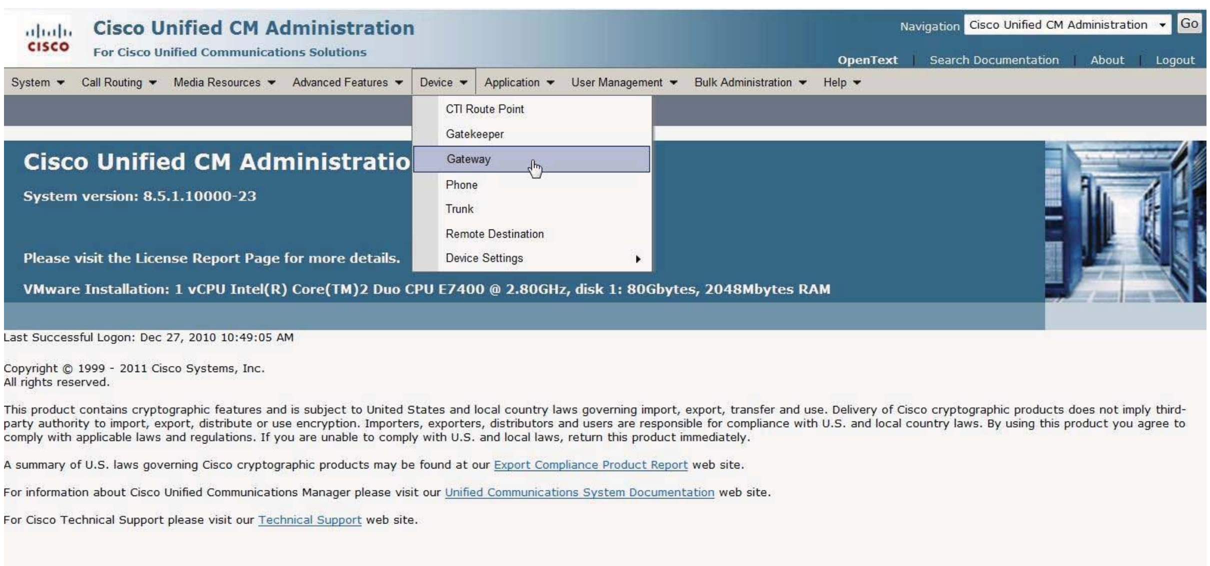
- IOS overview
- Troubleshooting guidelines

Configure OpenText RightFax Gateway

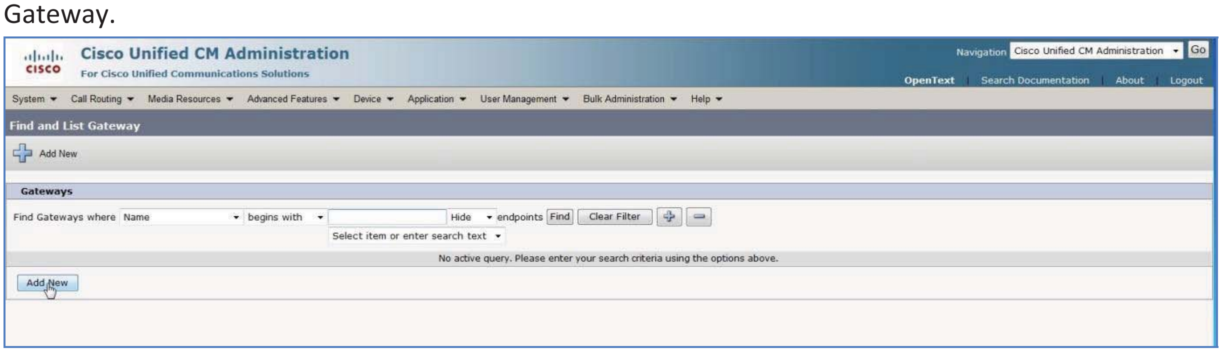
1. Using a web browser, log into the Cisco Unified CM Administration screen.



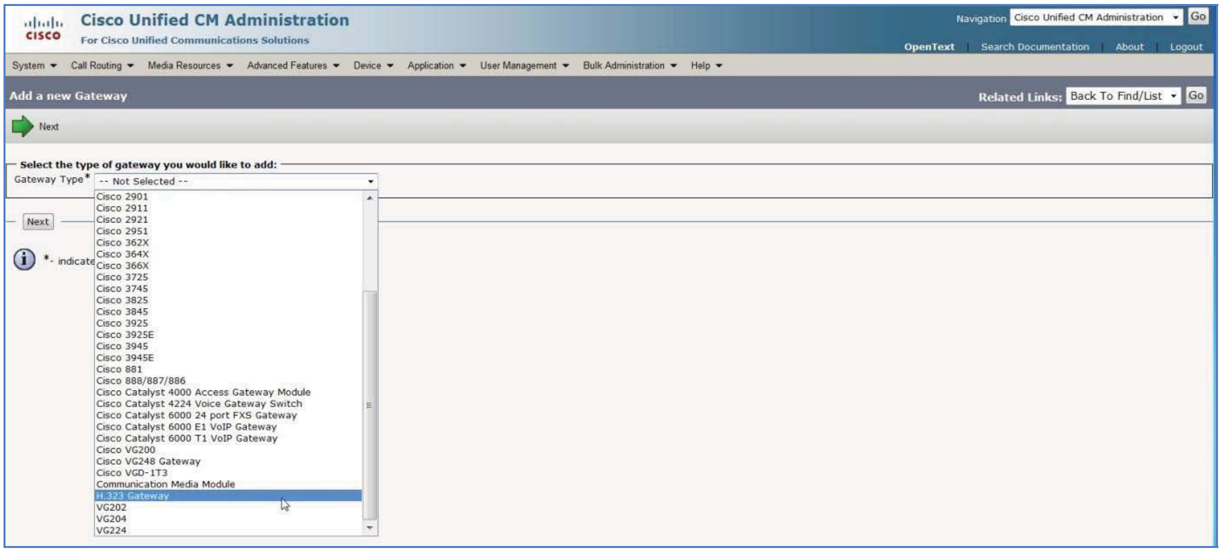
2. From the menu select **Device | Gateway**



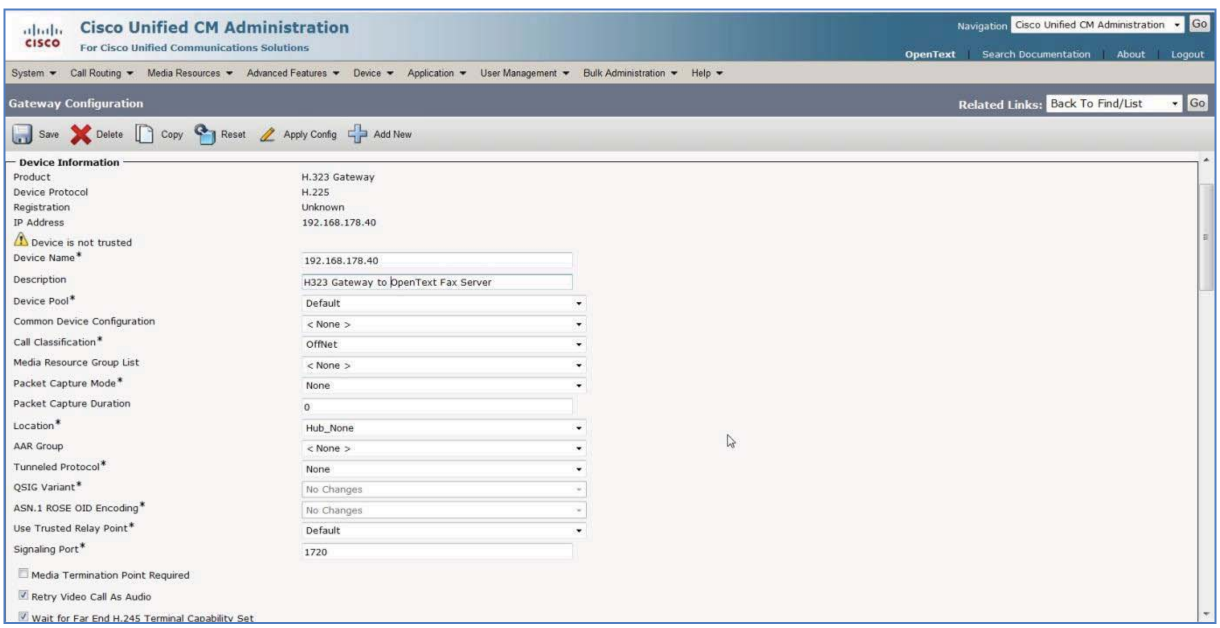
3. Press **Add New** to add a new H.323
   Gateway.



4. Select **H.323 Gateway** and press **Next**.



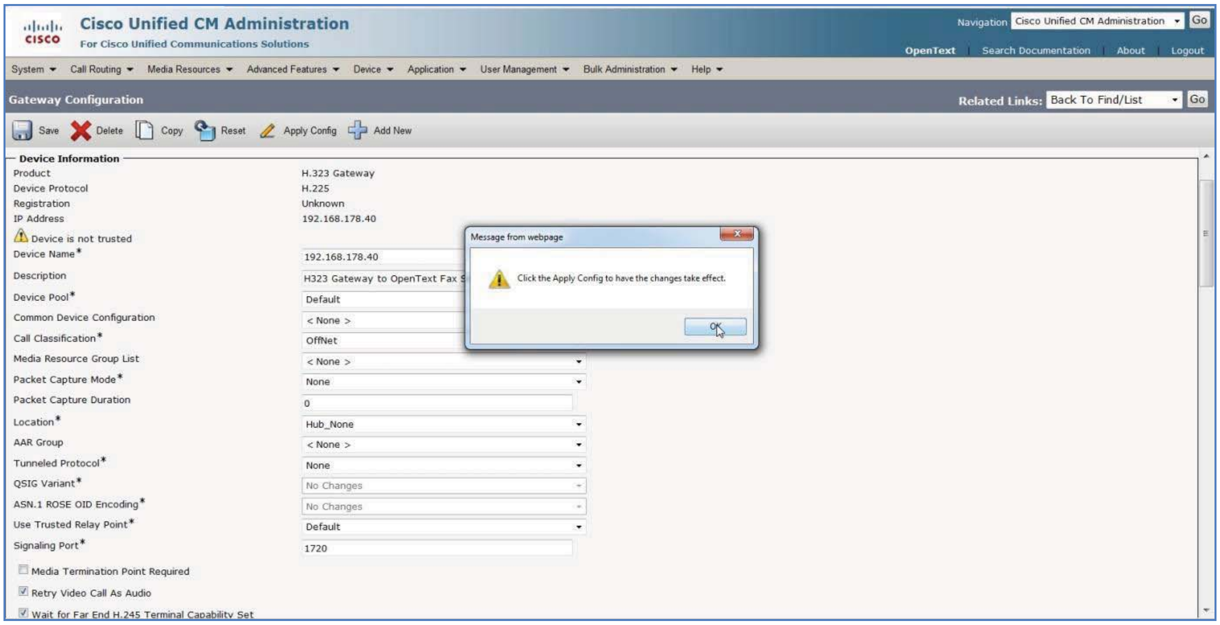
5. The following screen appears:



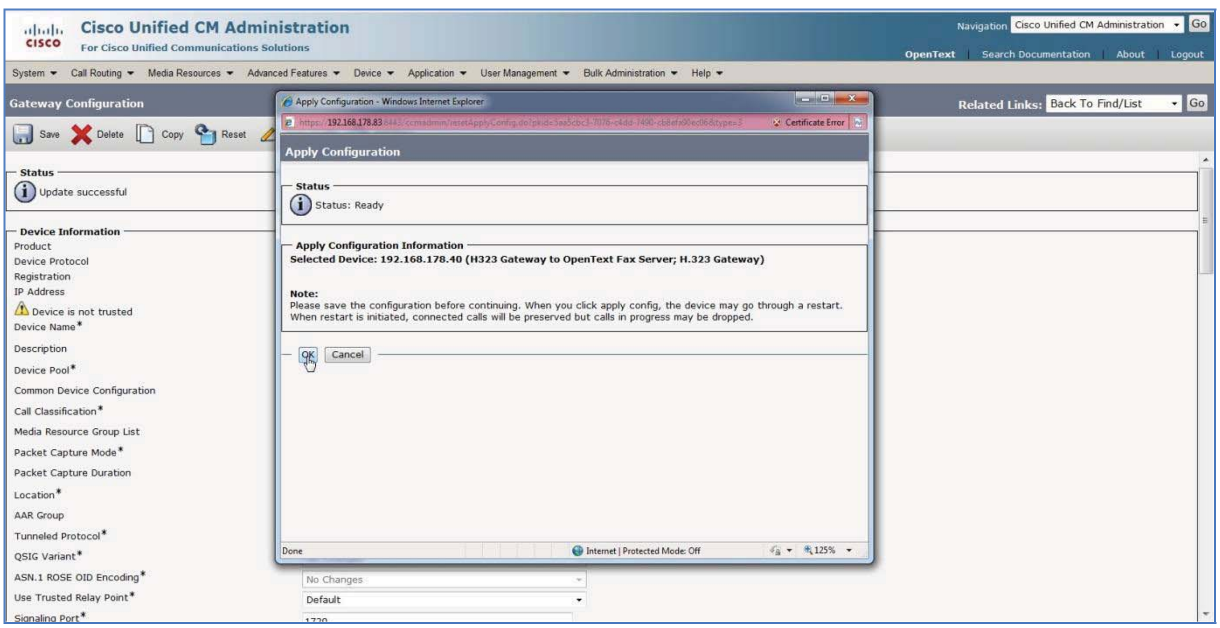
6. Set the following options:

   a. **Device Name**: 192.168.178.40 (address of OpenText RightFax)
   b. **Device Description**: H323 Gateway to OpenText RightFax
   c. **Device Pool**: Default
   d. **Call Classification**: OffNet

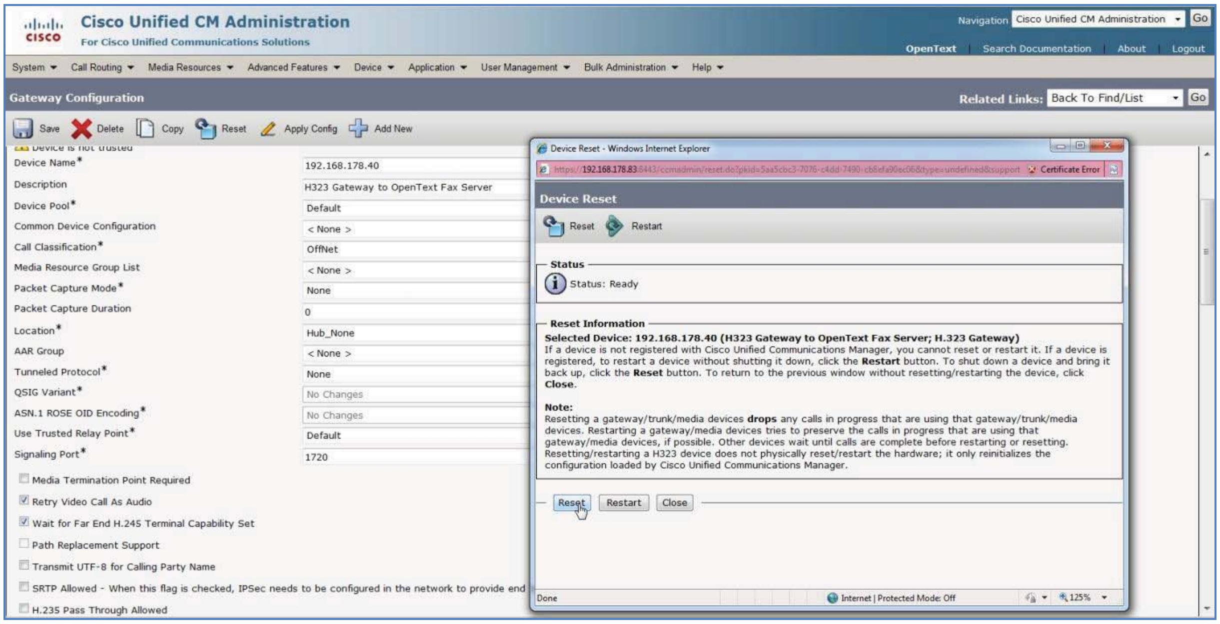
7. Press **Save.**



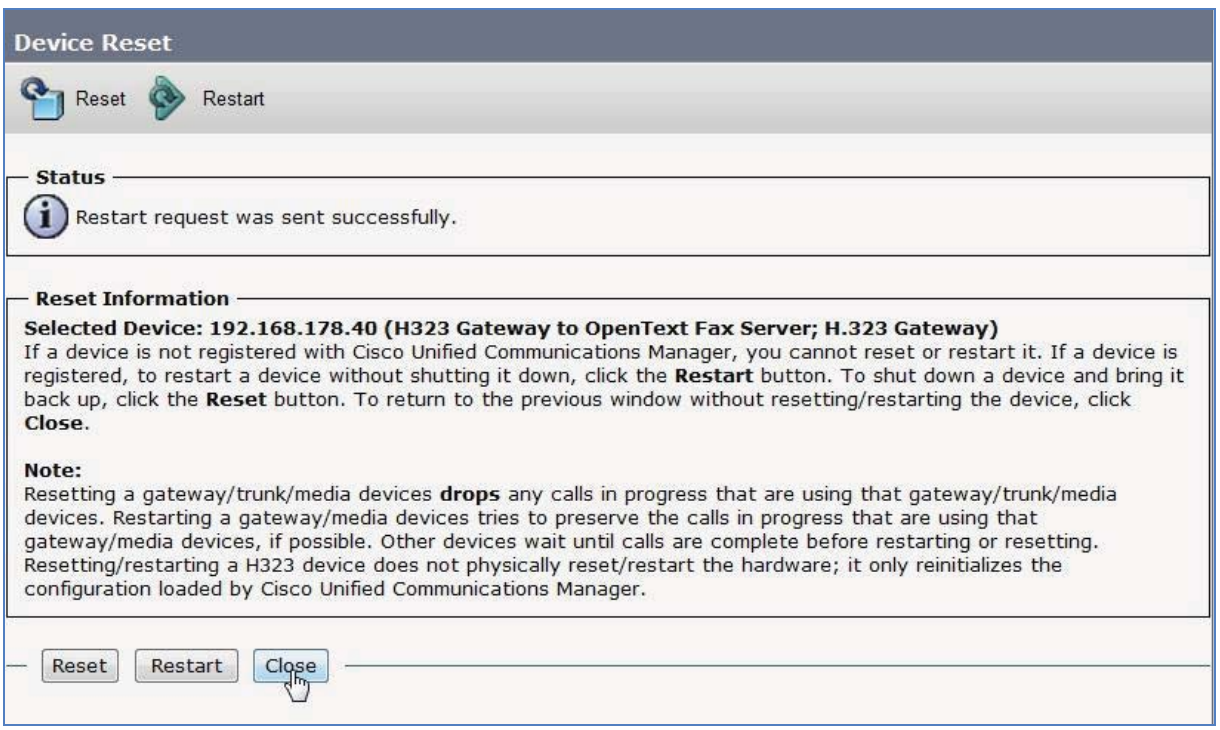
8. Click **OK,** then click **Apply Config.**



9. Click **OK** and click **Reset**.

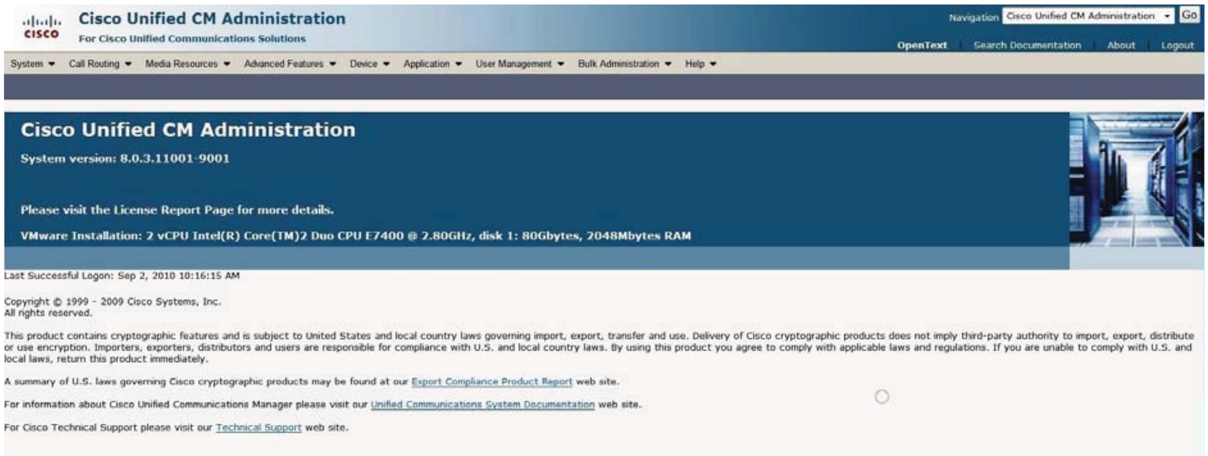


10. Click **Restart** and **Close.**

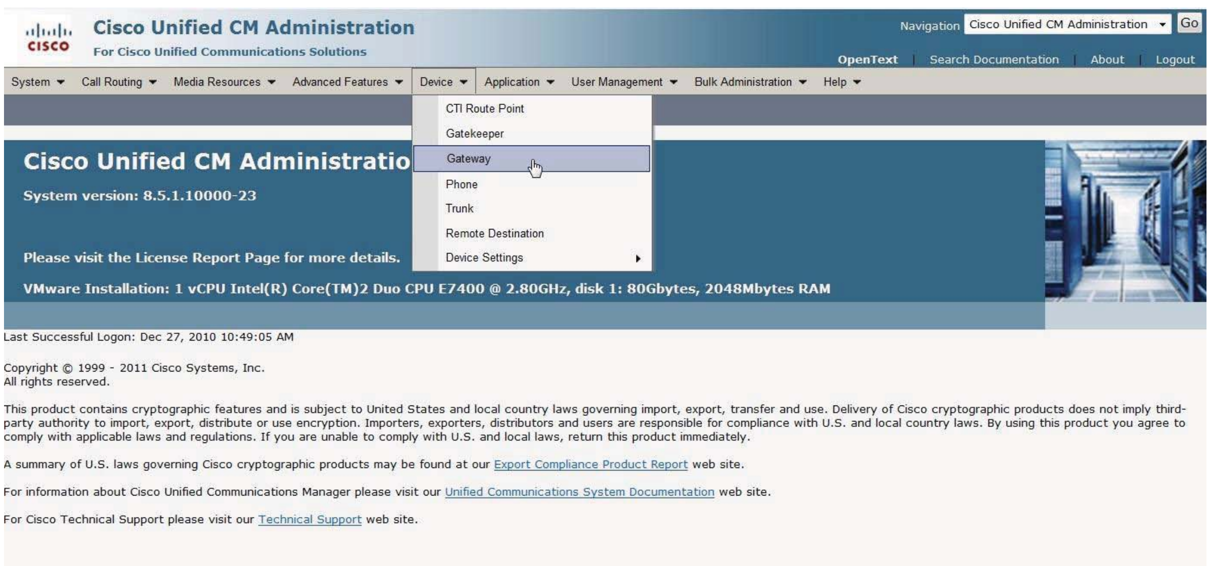


Configure MGCP Gateway

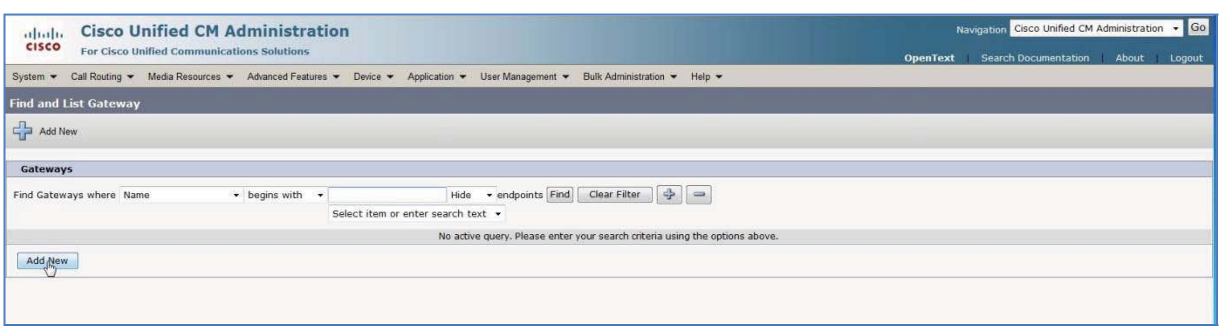
1. Using a web browser, log into the **Cisco Unified CM Administration** screen.



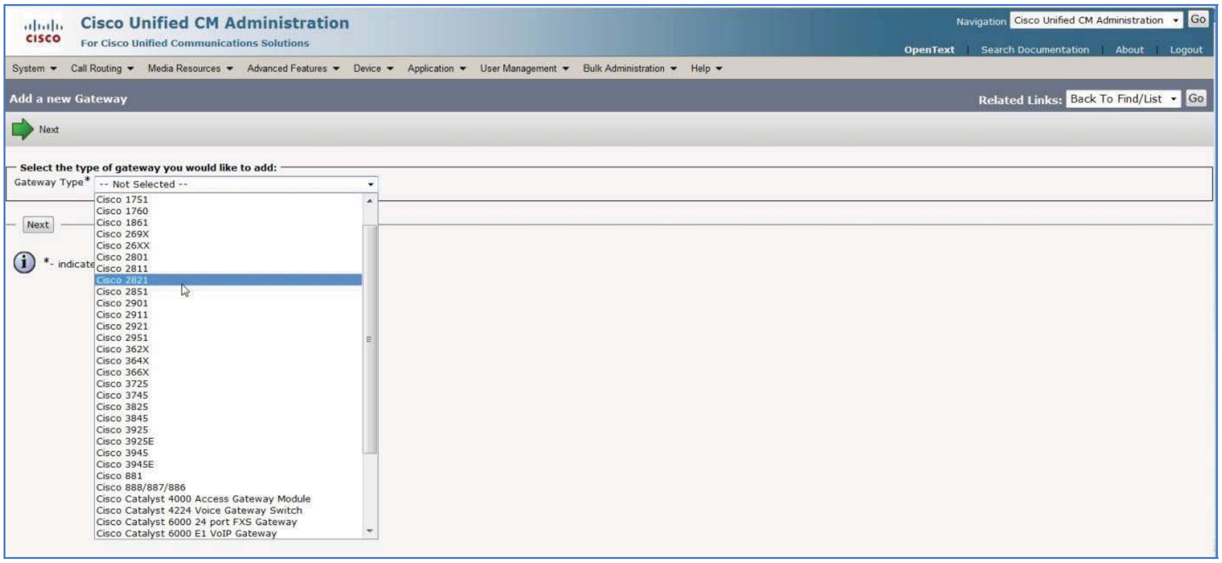
2. From the menu select **Device | Gateway.**



3. Press **Add New** to add a new H.323 Gateway.

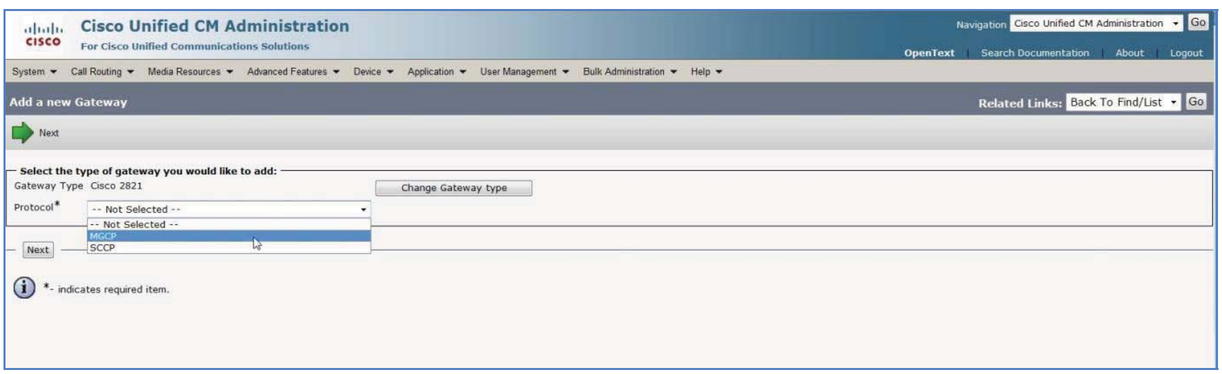


4. The following screen appears:

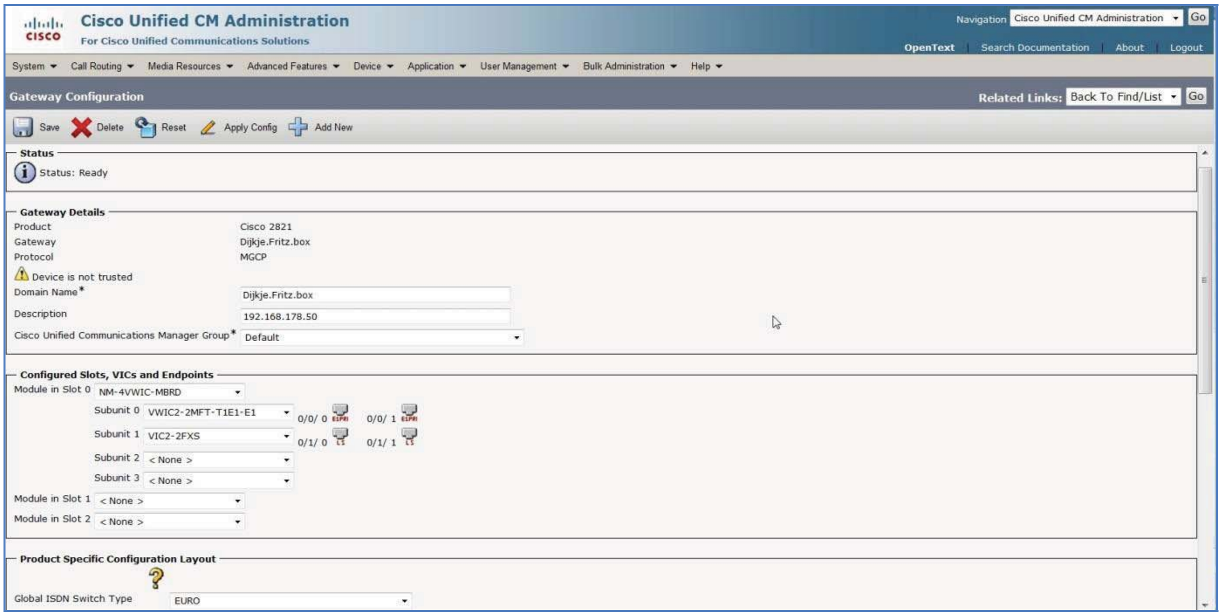


5. Select the **Gateway Type**. For MGCP gateways, choose the device type (router model or voice gateway). In this example, a Cisco 2821 router was selected. **Note:** You cannot configure Communication Manager to recognize the same device as both an MGCP and an H.323 gateway.

6. Next, set **Protocol to** MGCP and click **Next.**



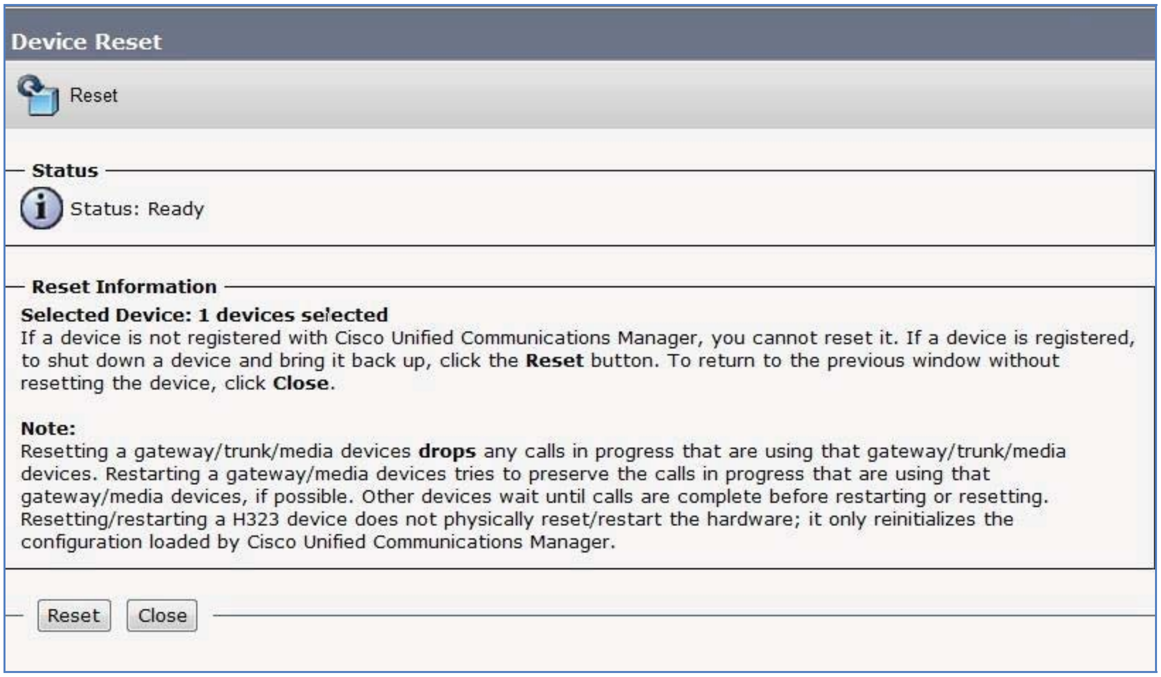
7. The **Gateway Configuration** screen appears:



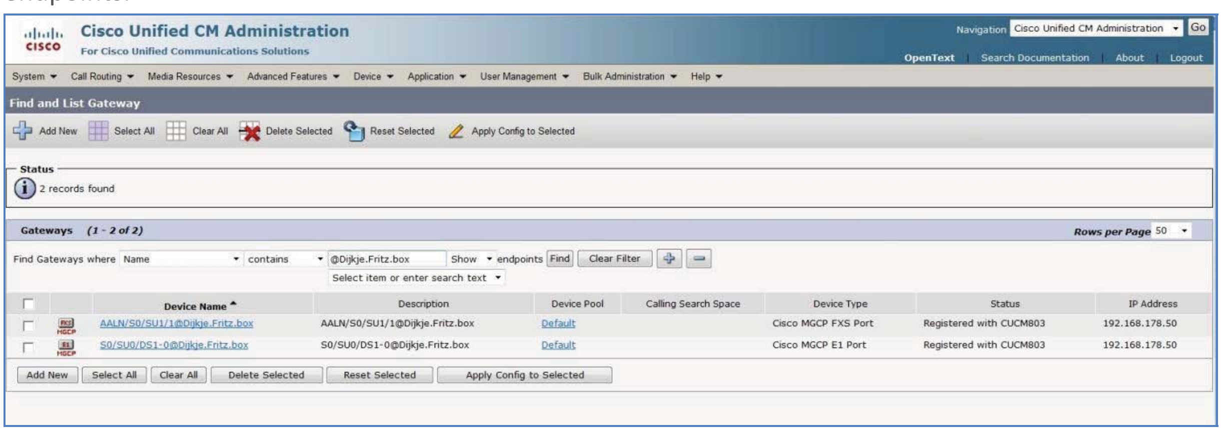
8. Under **Gateway Details,** enter the following information:
   a. **Domain Name:** Enter hostname of the router. ***Important information:***
      i. MGCP gateways are identified by *hostname*, not *IP address*.
      ii. If the router is configured with a domain name, append it to the hostname, such as Dijkje.Fritz.box.
      iii. The name is case sensitive.
   b. **Description (optional):** Enter optional description string.
   a. **Cisco Unified Communications Manager Group (required):** Choose a group, or set as Default.

9. Under **Configured Slots, VICs and Endpoints,** begin configuring endpoints.
   c. Available router slots are listed, with drop-down menu to select voice module type they contain, if any.
   d. ISR routers contain four WIC/VWIC slots that are not part of a separate module. These are listed in the drop-down menu as **"NM-4VWIC-MBRD."** Choose this option, as shown in the example, if you intend to use these slots.

10. On the next screen, reset the gateway by clicking **Reset** then click **Close**. *Note: Resetting the MGCP gateway drops all in-process calls on the gateway.*



11. To verify that the gateway is registered, go to the **Find and List Gateway**s screen. Click **Find.** The gateway should be listed along with registered endpoints.



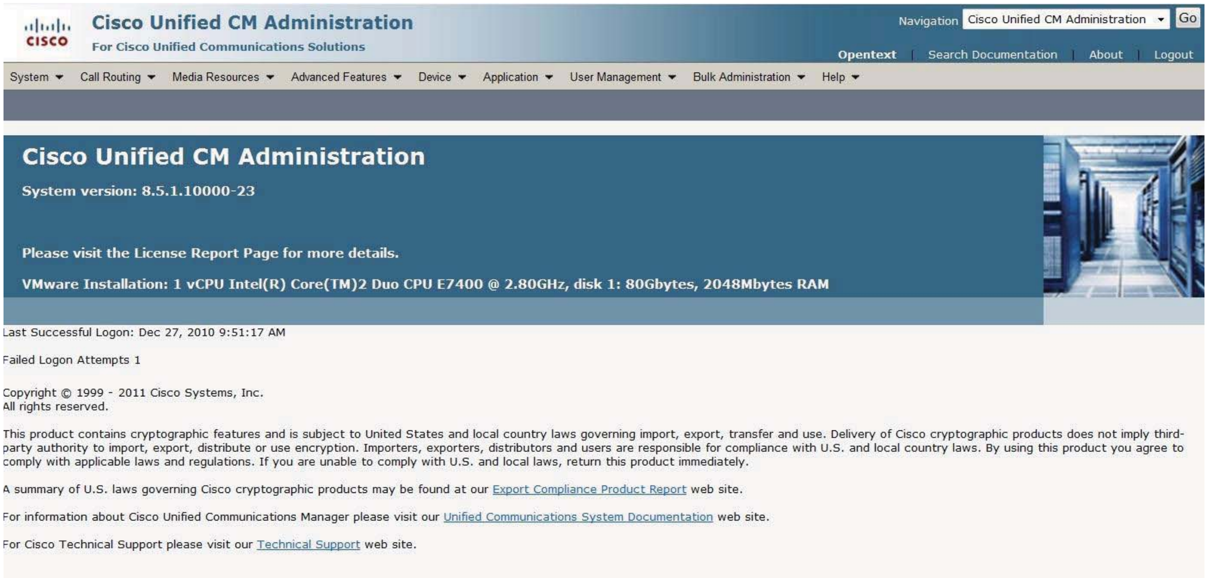
Ensure the Gateway is under MGCP control of CUCM803(c)

```
Dijkje#SH CCM
MGCP Domain Name: Dijkje.fritz.box
Priority         Status                   Host
============================================================
Primary          Registered               192.168.178.85
First Backup     None
Second Backup    None

Current active Call Manager:     192.168.178.85
Backhaul/Redundant link port:    2428
Failover Interval:               30 seconds
```

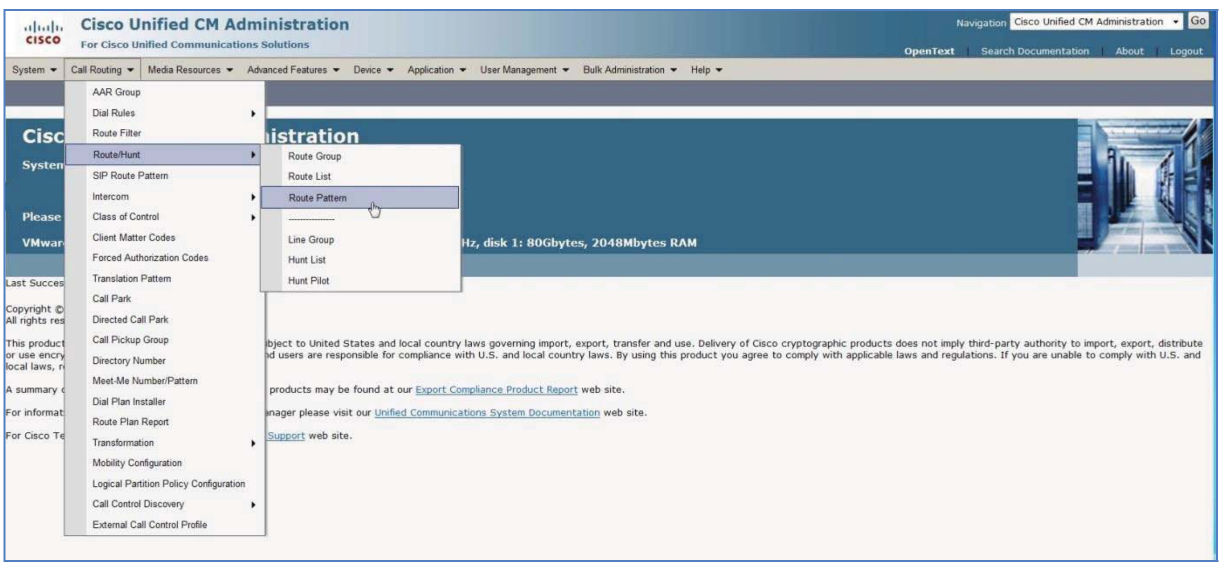
```
Keepalive Interval:              15 seconds
Last keepalive sent:             16:57:26 PCTime Sep 9 2010 (elapsed time:
00:00:04)
Last MGCP traffic time:          16:57:26 PCTime Sep 9 2010 (elapsed time:
00:00:04)
Last failover time:              None
Last switchback time:            None
Switchback mode:                 Graceful
MGCP Fallback mode:              Not Selected
Last MGCP Fallback start time:   None
Last MGCP Fallback end time:     None
MGCP Download Tones:             Disabled
TFTP retry count to shut Ports: 2

Backhaul Link info:
    Link Protocol:      TCP
    Remote Port Number: 2428
    Remote IP Address:  192.168.178.85
    Current Link State: OPEN
    Statistics:
        Packets recvd:   2
        Recv failures:   0
        Packets xmitted: 2
        Xmit failures:   0
    PRI Ports being backhauled:
        Slot 0, VIC 0, port 0
```

Configure Call Routing OpenText RightFax to PSTN

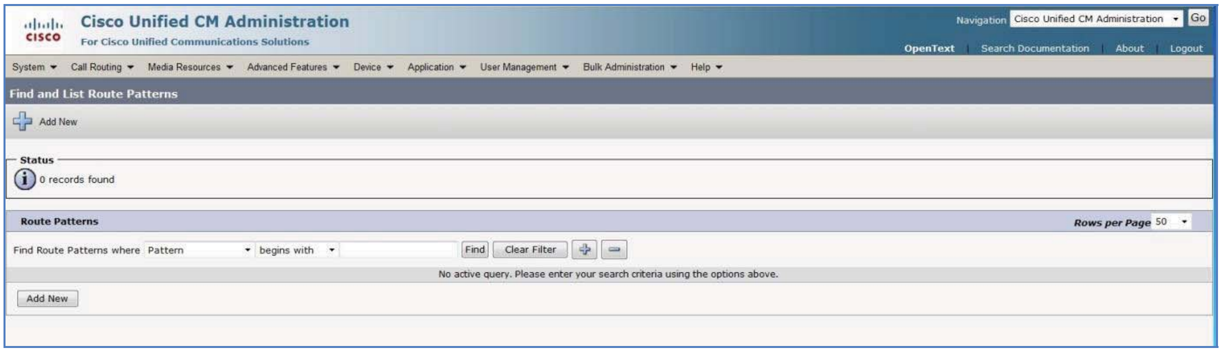
1. Using a web browser, log into the **Cisco Unified CM Administration** screen.



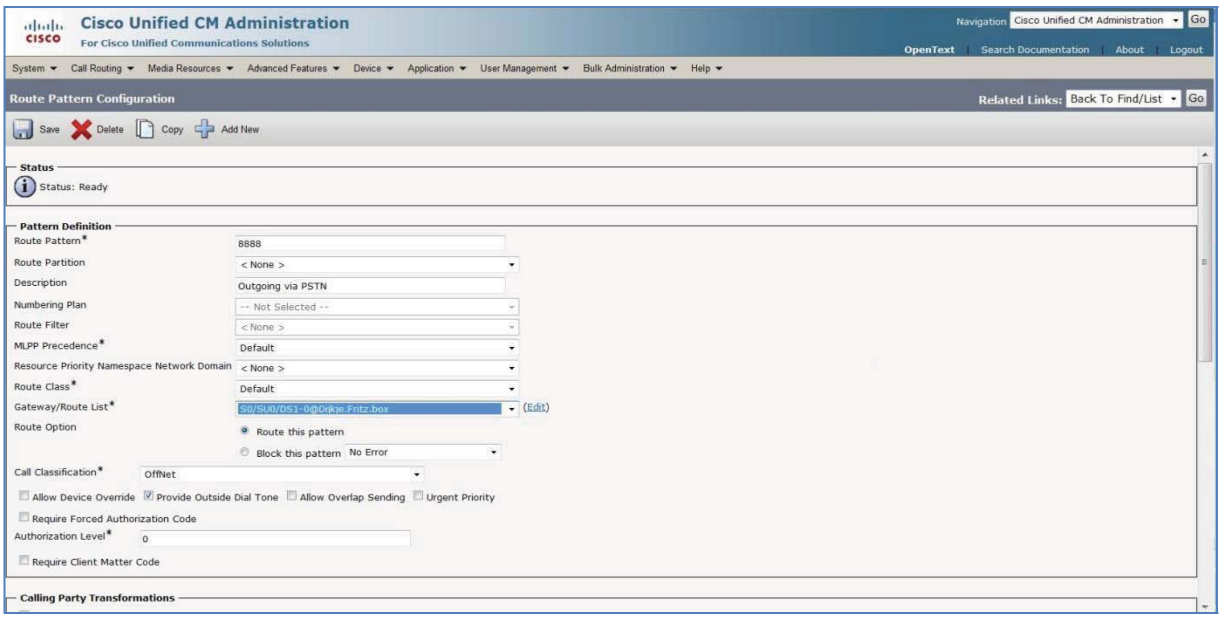
2. From the menu select Call Routing | Route / Hunt | Route Pattern.



3. Click on **Add New** to add a new Route Pattern.



4. The following screen appears:

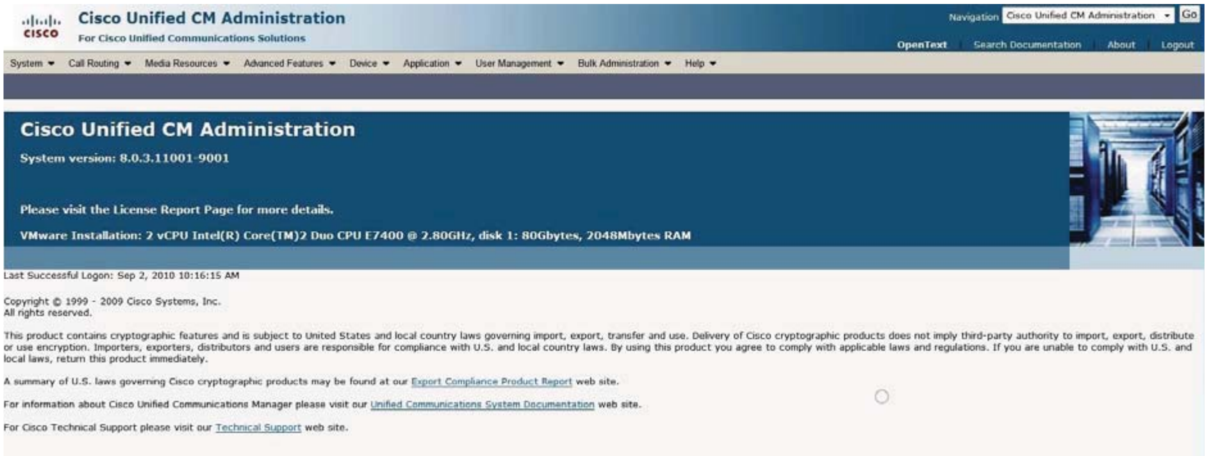


5. Set **Route Pattern** to “8888” to send faxes via the E1 (PSTN).

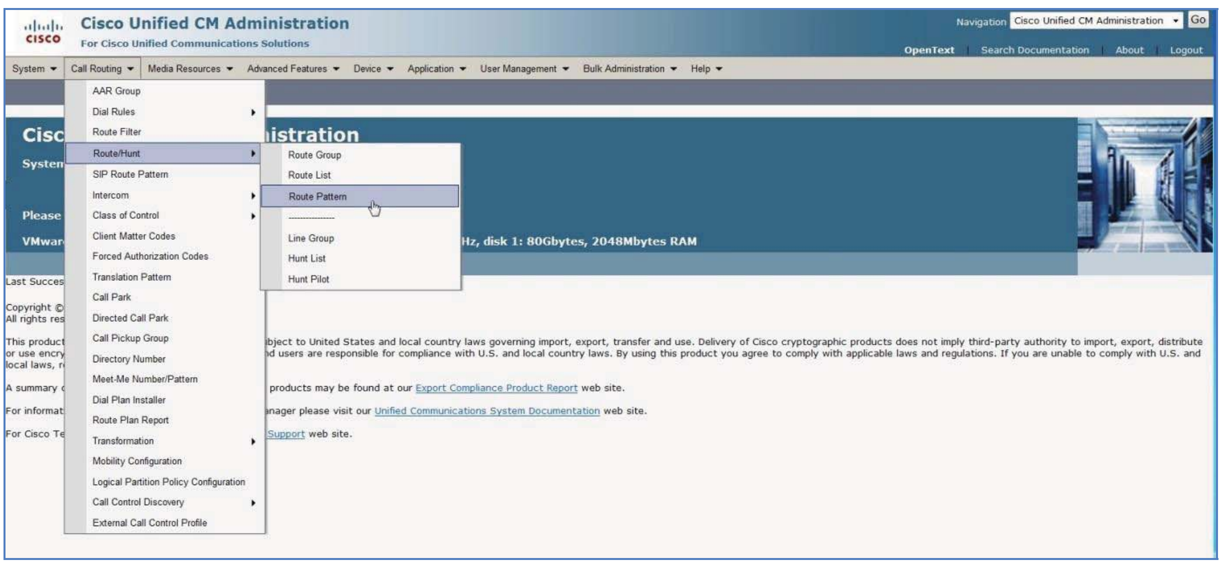
6. In this scenario, **Gateway/Route List** is S0/SUO/DS1-0@Dijkje.Fritz.box (the MGCP Trunk of the Gateway).

Configure Call Routing (PSTN to OpenText RightFax)

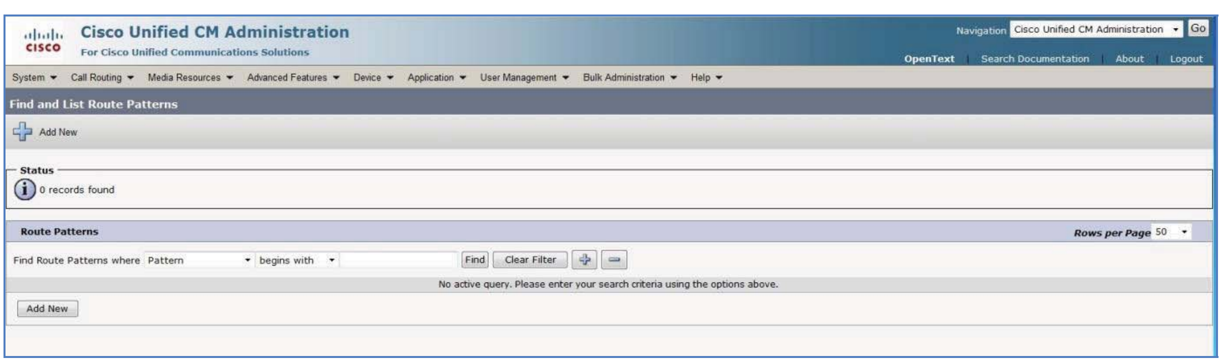
7. Using a web browser, log into the **Cisco Unified CM Administration** screen.



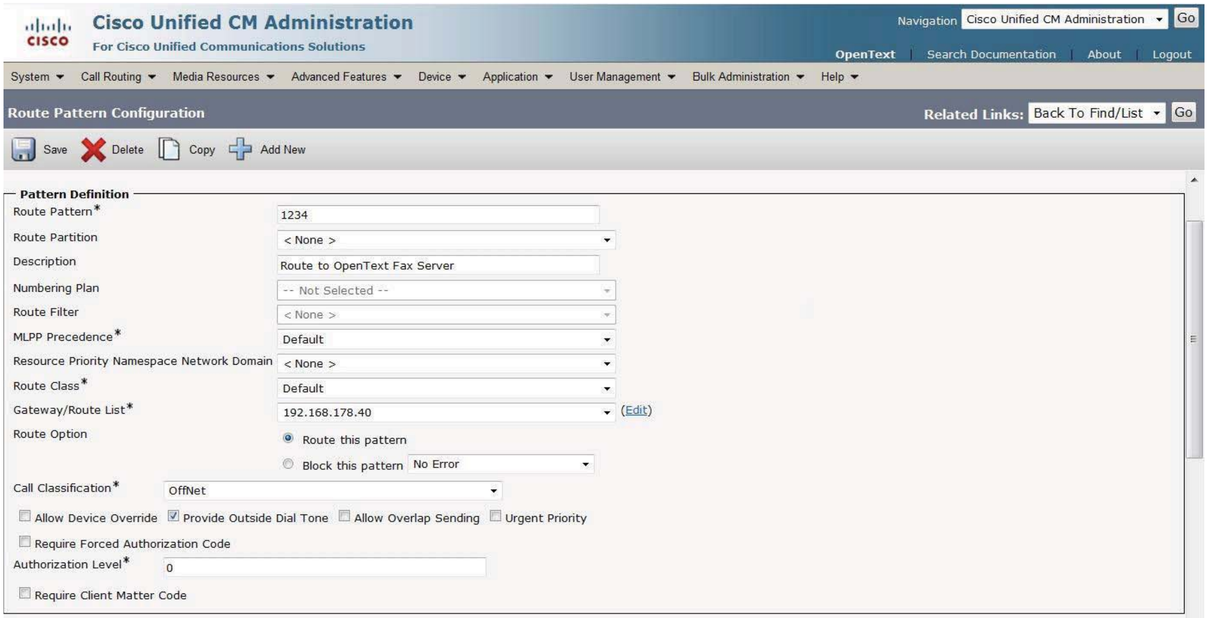
8. Select **CallRouting | Route Hunt | Route Pattern**.



9. Click on **Add New** to add a new Route Pattern



10. The following screen appears:



11. Set the following options:

    a. **Route Pattern:** "1234" (where faxes can be sent from the PSTN to OpenText RightFax via the CUCM).
    b. **Gateway/Route List**: Enter the IP address of OpenText RightFax.

12. Click **Save** to save the configuration changes.

IOS overview

```
hostname Dijkje
!
no aaa new-model
clock timezone PCTime 1
network-clock-participate wic 0
no network-clock-participate aim 0
!
!
ip cef
!
!
ip domain name fritz.box
ip name-server 192.168.178.1
ip auth-proxy max-nodata-conns 3
ip admission max-nodata-conns 3
!
isdn switch-type primary-net5
!
voice-card 0
 dspfarm
!
!
voice service voip
 fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none

!
!
voice class codec 1
 codec preference 1 g711alaw
!
!
controller E1 0/0/0
 clock source internal
 pri-group timeslots 1-8,16 service mgcp
!

interface GigabitEthernet0/0
 ip ddns update dijkje
 ip address 192.168.178.50 255.255.255.0
 duplex half
 speed auto
 no keepalive
 no mop enabled
!
interface Serial0/0/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5
 isdn protocol-emulate network
 isdn incoming-voice voice
 isdn bind-l3 ccm-manager
 no cdp enable
!
interface Serial0/3/0
 no ip address
 shutdown
```

```
 clock rate 2000000
!
no ip forward-protocol nd
!
!
ip http server
ip http authentication local
ip http secure-server
!
!
!
!
control-plane
!
!
!
voice-port 0/0/0:15
!
voice-port 0/1/0
 compand-type a-law
 cptone NL
 shutdown
 description fxo00
 bearer-cap Speech
!
voice-port 0/1/1
 compand-type a-law
 cptone NL
 description FX01
 bearer-cap Speech
!
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 192.168.178.85
!
mgcp
mgcp call-agent 192.168.178.85 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp default-package fxr-package
!
mgcp profile default
!
!
dial-peer voice 1000 pots
 huntstop
 service mgcpapp
 answer-address 1000
 destination-pattern 1000
 no digit-strip
 direct-inward-dial
 port 0/1/0
!
dial-peer voice 8888 pots
 service mgcpapp
 destination-pattern 8888
 no digit-strip
 direct-inward-dial
!
```

```
!
gateway
 timer receive-rtp 1200
!
sip-ua
scheduler allocate 20000 1000
!
end
```

Troubleshooting guidelines

The following suggestions may assist in troubleshooting issues that arise:

- Reset the MGCP statistical counters with the **`clear mgcp statistics`** command.
- If no RTP traffic is getting through make sure **IP routing** is enabled.
- Use the **`show rtp statistics`** command, then turn on the **`debug ip udp`** command and track down the MGCP RTP packets.

```
Dijkje# show rtp statistics
RTP Statistics info:
No. CallId Xmit-pkts Xmit-bytes Rcvd-pkts  Rcvd-bytes Lost pkts  Jitter Latenc
1   17492  0x8A      0x5640     0x8A       0x5640     0x0        0x0      0x0
Dijkje# show rtp statistics
RTP Statistics info:
No. CallId Xmit-pkts Xmit-bytes Rcvd-pkts  Rcvd-bytes Lost pkts  Jitter Latenc
1   17492  0xDA      0x8840     0xDB       0x88E0     0x0        0x160    0x0
```

- If an RSIP message is not received by the call agent make sure that the **`mgcp call-agent`** command or the MGCP profile **`call-agent`** command is configured with the correct call agent name or IP address and UDP port. Use the **`show mgcp`** command or the **`show mgcp profile`** command to display this information:

```
Dijkje# show mgcp
MGCP Admin State ACTIVE, Oper State ACTIVE - Cause Code NONE
MGCP call-agent: 192.168.178.85  Initial protocol service is MGCP, v. 1.0
...
MGCP gateway port: 2727, MGCP maximum waiting delay 3000
...
Dijkje# show mgcp profile
MGCP Profile default
Description: None
Call-agent: 192.168.178.85 2427 Initial protocol service is MGCP 0.1
Tsmax timeout is 20 sec, Tdinit timeout is 15 sec
Tdmin timeout is 15 sec, Tdmax timeout is 600 sec
Tcrit timeout is 4 sec, Tpar timeout is 16 sec
Thist timeout is 30 sec, MWI timeout is 16 sec
Ringback tone timeout is 180 sec, Ringback tone on connection timeout is 180 sec
Network congestion tone timeout is 180 sec, Busy tone timeout is 30 sec
Network busy tone timeout is 0 sec
Dial tone timeout is 16 sec, Stutter dial tone timeout is 16 sec
Ringing tone timeout is 180 sec, Distinctive ringing tone timeout is 180 sec
Continuity1 tone timeout is 3 sec, Continuity2 tone timeout is 3 sec
Reorder tone timeout is 30 sec,  Persistent package is ms-package
Max1 DNS lookup: ENABLED, Max1 retries is 5
Max2 DNS lookup: ENABLED, Max2 retries is 7
Source Interface: NONE...
```

- To verify connections and endpoints, use the **show mgcp** command:

```
Dijkje# show mgcp connection
Endpoint  Call_ID(C) Conn_ID(I) (P)ort (M)ode (S)tate (C)odec (E)vent[SIFL]
(R)esult[EA]
1. S0/DS1-1/5      C=F123AB,5,6  I=0x3  P=16506,16602  M=3  S=4  C=1 E=2,0,0,2
R=0,0
2. S0/DS1-1/6      C=F123AB,7,8  I=0x4  P=16602,16506  M=3  S=4  C=1 E=0,0,0,0
R=0,0
Dijkje# show mgcp endpoint
Interface E1 0/0/0

             ENDPOINT-NAME    V-PORT     SIG-TYPE   ADMIN
     S0/SU0/ds1-0/1@Dijkje   0/0/0:15        none    up
     S0/SU0/ds1-0/2@Dijkje   0/0/0:15        none    up
     S0/SU0/ds1-0/3@Dijkje   0/0/0:15        none    up
     S0/SU0/ds1-0/4@Dijkje   0/0/0:15        none    up
     S0/SU0/ds1-0/5@Dijkje   0/0/0:15        none    up
     S0/SU0/ds1-0/6@Dijkje   0/0/0:15        none    up
     S0/SU0/ds1-0/7@Dijkje   0/0/0:15        none    up
     S0/SU0/ds1-0/8@Dijkje   0/0/0:15        none    up


Interface E1 0/0/1

             ENDPOINT-NAME    V-PORT     SIG-TYPE   ADMIN
```

- If an MGCP message is rejected, it might be because the remote media gateway does not support SDP mandatory parameters (the o=, s=, and t= lines). If this is the case, configure the **mgcp sdp simple** command to send SDP messages without those parameters.
- If there are problems with voice quality, make sure that **cptone** (voice-port configuration) command is set for the correct country code.
- Capturing RTP packets from a sniffer may help isolate the problem. You may be able to decide such questions as whether the payload type or timestamps are set correctly.
- To check operation of interfaces, use the **show interface** command.
- To view information about activity on the T1 or E1 line, use the **show controllers** command. Alarms, line conditions, and other errors are displayed. The data is updated every 10 seconds. Every 15 minutes, the cumulative data is stored and retained for 24 hours.
- When necessary, enable debug traces for errors, events, media, packets, and parser. The command **debug mgcp packets** can be used to monitor message flow in general. Note that there is always a performance penalty when using debug commands. The sample output

below shows the use of the optional **`input-hex`** keyword to enable display of hexadecimal values.

```
Dijkje# debug mgcp {all | errors | events | packets {input-hex}| parser}
Dijkje# debug mgcp packets input-hex
Media Gateway Control Protocol input packets in hex value debugging is on
MGCP Packet received -
DLCX 49993 * MGCP 0.1
MGCP Packet received in hex -
44 4C 43 58 20 34 39 39 39 33 20 2A 20 4D 47 43 50 20 30 2E 31 A
send_mgcp_msg, MGCP Packet sent ---> </nowiki>
250 49993
```