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SIP Trunking Configuration Guide
for
Cisco Unified Communications Manager
Version 8.6.2.20000-2
With CUBE 15.2.1T1
c3900e-universalk9-mz.SPA.152-1.T1.bin

Document Revision 2.1

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1 Audience

This document is intended for the SIP trunk customer's technical staff and Value Added Retailer (VAR) having installation and operational responsibilities.

2 Introduction

This Configuration Guide describes configuration steps for Cox SIP trunking with the Cisco Unified Communications Manager (CUCM) 8.6.2 and the Cisco Unified Border Element (CUBE) for connectivity to Cox's SIP Trunking service. Trunking provides both inbound and outbound call services replacing traditional ISDN PRI services.Cox SIP Cox SIP trunking is a scalable and efficient IP trunking telecommunication solution for your business that provides all the traditional services such as Direct Inward Dialing, Hunting, Calling Name, Calling Number, Local/Long Distance and Business Continuity options, including:

- Burstable Trunk Capacity – Dynamically increases call capacity during peak busy periods so your customers never receive a busy signal.
- Call Forward Always – On the trunk group pilot number for all calls in case of an outage (i.e., flood, fire, loss of power, etc.).
- Call Forward Not Reachable – On the trunk group pilot number that operates on a per-call contingency basis to forward the call to any PSTN number (i.e., call center or alternate office location) during temporary call completion impairments.
- Route Exhaustion – Automatic reroute of trunk group calls to any PSTN phone number (i.e., a call center) if calls can't be completed to the PBX.
- Support for geo-redundant PBX deployments and automatic reroute of SIP trunks to the backup customer data center

All calls are routed over Cox's national fiber network with guaranteed Quality of Service (QoS); calls never traverse the Internet

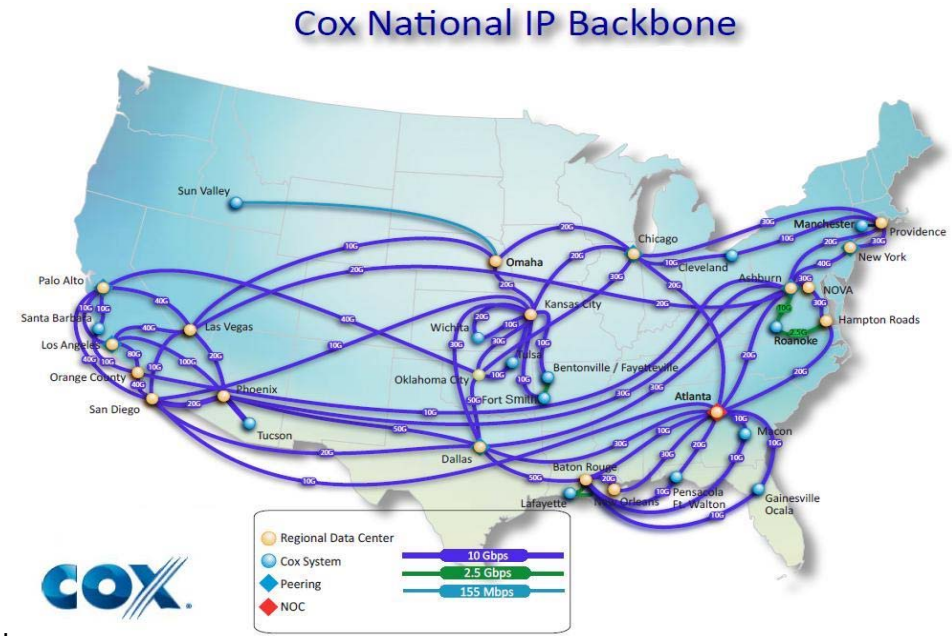

Figure 1 - Cox Fiber Network

2.1 tekVizion Labs

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3 SIP Trunking Network Components

The network for the SIP trunk reference configuration is illustrated below and is representative of a CUCM with CUBE configuration

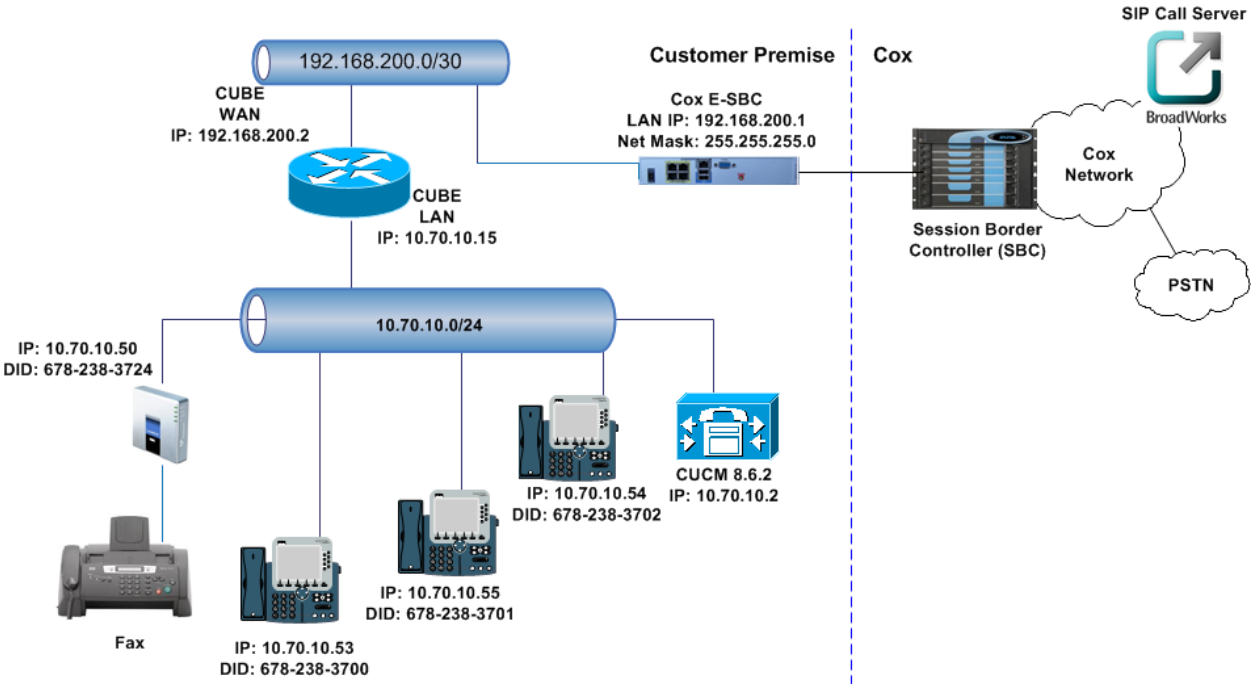


Figure 2 - SIP Trunk Lab Reference Network

Note: The CUCM does not offer DHCP server for dynamic IP address assignment for the SIP phones; however, the Cox Enterprise Session Border Controller (E-SBC) requires a static LAN IP address that must be manually assigned by the LAN network administrator. The DHCP server is provisioned on the Ethernet switch. The DHCP's IP address pool is constrained so that the E-SBC can be assigned an IP address outside of the pool.

The lab network consists of the following components:

- CUCM PBX for voice features, SIP proxy and SIP trunk termination.
- CUBE for routing between internal networks and to the E-SBC on the external network
- Various SIP phones on the local LAN.
- The Cox E-SBC is the Edgewater Networks (www.edgewaternetworks.com) EdgeMarc appliance. The EdgeMarc is the service demarcation point between customer's LAN network and Cox's WAN network and provides firewall/NAT traversal, B2BUA and SIP Application-level gateway. The EdgeMarc has diverse routes to a primary and secondary Acme SBC.
- Acme Packet Net-Net 9200 Session Border Controllers (SBC).

3.1 Hardware Components

- CUCM installed on Cisco 7800 Series
- Cisco IOS gateway (3900 series) running CUBE
- EdgeMarc 5300LF2 E-SBC Relaese 11.6.5
- Cisco IP Phones (7970, 7975)
- Analog Fax Machine

3.2 Software Requirements

- Cisco Unified Communications Manager 8.6.2.20000-2
- Cisco Unified Border Element (CUBE) 15.2.1T1 (c3900e-universalk9-mz.SPA.152-1.T1.bin)
- EdgeMarc 5300LF2 E-SBC Release 11.6.5

4 Features

4.1 Features Supported

- Basic calls using G.711ulaw
- Calling Party Number Presentation
- Anonymous call
- Call Transfer
- Call Forwarding
- Call Hold and Resume
- Call Pickup
- Call Waiting
- DND
- Call Park
- Hunt groups (Simultaneous and Sequential Ring)
- Three-Way Calling
- PBX Account Codes
- PBX Authorization Codes
- Fax Receive
- Pass Thru Fax
- E911 Call
- RFC2833 transcoding
- PBX-Defined Caller ID (spoofing)

4.2 Features Not Supported

- Dial-Up Modem
- T.38 Fax

5 Caveats and Limitations

- **Cox Required Configuration:**
  To enable Early Media with RFC2833 In-band DTMF transport, the Cisco Unified Border Element (CUBE) must be configured with the following two commands.:
  - allow-connections sip to sip
  - early-offer forced
- T.38 fax may be supported by the Linksys ATA SPA2102, but at this time the device would not offer T.38. G711 fax is successful.
- At the time of this writing, Blind Transfer has not passed using G711. It will work when configured with G729, but with G711, there is no talk path. A trouble ticket has been opened with ACME and the results are pending.
- Modem test did not pass. Test originated from PBX, and receiving side did not connect. This is most likely a lab environment artifact.
- Unable to test Auto Attendant with CUCM.

6 Configuration

6.1 Configuration Checklist

In this section we present an overview of the steps that are required to configure CUCM and CUBE for SIP Trunking as well as the PBX features that were tested.

Table 1 – PBX Configuration Steps

Step	Description	Reference
Step 1	System IP Address	Section 6.3.1
Step 2	Cisco Unified Border Element (CUBE) Configuration	Section 6.3.2
Step 3	Cisco Unified Communications Manager Configuration	Section 6.3.3
Step 4	Enterprise Parameters	Section 6.3.4
Step 5	Service Parameters	Section 6.3.5
Step 6	Device Pools	Section 6.3.6
Step 7	SIP Trunks	Section 6.3.7
Step 8	SIP Profiles	Section 6.3.8
Step 9	SIP Trunk Security Profile	Section 6.3.9
Step 10	Route Pattern	Section 6.3.10
Step 11	Line Groups	Section 6.3.11
Step 12	Hunt Lists	Section 6.3.12
Step 13	Hunt Pilot Numbers	Section 6.3.13
Step 14	Client Matter Codes	Section 6.3.14
Step 15	Forced Authorization Codes	Section 6.3.15
Step 16	Call Park Numbers	Section 6.3.16
Step 17	Call Pickup Groups	Section 6.3.17
Step 18	Directory Numbers	Section 6.3.18
Step 19	Route Plan Report	Section 6.3.19
Step 20	Unity Voice Mail Ports	Section 6.3.20
Step 21	Message Waiting Indicator	Section 6.3.21
Step 22	Voice Mail Pilot	Section 6.3.22
Step 23	Voice Mail Profile	Section 6.3.23
Step 24	IP Phone Service	Section 6.3.24
Step 25	Linksys ATA	Section 6.3.25

6.2 IP Address Worksheet

The specific values listed in the table below and in subsequent sections are used in the lab configuration described in this document, and are for **illustrative purposes only**. The customer must obtain and use the values for your deployment.

Table 2 – IP Addresses

Component	Cox Lab Value	Customer Value
EdgeMarc E-SBC		
• LAN IP Address	192.168.200.1	
• LAN Subnet Mask	255.255.255.0	
CUCM IP PBX		
• System IP Address This is the IP address of the CUCM. This IP address is typically on the same subnet as the LAN IP Address of the E-SBC. If this is not the case, then Layer 3 routing must be in place.	10.70.10.2	
• CUBE LAN	10.70.10.15	
• CUBE WAN	192.168.200.2	
• Default Gateway The Default Gateway must be the LAN Network default Gateway. This will allow the administrator to log in via his\her workstation if the workstation is on a different network	10.70.10.1	
• DNS This is the DNS server for the Enterprise network. Cox Communications does not supply DNS services.	10.64.1.3	

6.3 CUCM with CUBE Detailed Configuration Steps

Equipment used for configuration setup:

- CUCM 8.6.2.20000-2 installed on Cisco 7800 Series
- CUBE installed on Cisco 3900 Series with c3900e-universalk9-mz.SPA.152-1.T1.bin
- Cisco IP Phones (7970, 7975)

6.3.1 System IP Address

The IP Address of the CUCM server is 10.70.10.2 with a subnet mask of 255.255.255.0

6.3.2 Cisco Unified Border Element (CUBE) Configuration

Cisco Unified Border Element (CUBE) routers are utilized to hand off SIP calls to the Cox E-SBC. The CUBE feature set allowed for modification of key SIP headers using manipulation rules to format both SIP INVITE and SIP RE-INVITE messages to proceed properly.

The following sip profile was used to inject “user=phone” into the INVITE and REINVITE headers.

```
voice class sip-profiles 1
request INVITE sip-header SIP-Req-URI modify " SIP/2.0" ";user=phone SIP/2.0"
request REINVITE sip-header SIP-Req-URI modify " SIP/2.0" ";user=phone SIP/2.0"
request INVITE sip-header Contact modify ">" ";user=phone>"
request REINVITE sip-header Contact modify ">" ";user=phone>"
request INVITE sip-header To modify ">" ";user=phone>"
request REINVITE sip-header To modify ">" ";user=phone>"
request INVITE sip-header From modify ">" ";user=phone>"
request REINVITE sip-header From modify ">" ";user=phone>"
```

The SIP-SIP calling, in-call signaling, and SIP profile activation was enabled using the following commands.

voice service voip
 ip address trusted list
  ipv4 10.70.10.2[1]
  ipv4 192.168.200.1[2]
 address-hiding[3]
 mode border-element
 allow-connections sip to sip
 no supplementary-service sip refer
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  asserted-id pai[4]
  privacy pstn[5]
  early-offer forced[6]
  midcall-signaling passthru[7]

When the CUCM is configured with the CUBE, the trunk is pointed to the LAN side of the CUBE with the sip-server definition

sip-ua
 sip-server ipv4:192.168.200.1:5060

[1] This is the IP address of the CUCM
[2] This is the IP address of the E-SBC LAN
[3] Hides the signaling and media peer addresses from the endpoints
[4] To enable the translation to PAID headers in the outgoing header at a global level
[5] To support the User Privacy Policy on the CUCM, this flag is set on the CUBE to preserve the P-Asserted-Identity and Privacy header on the outgoing SIP INVITE
[6] This feature alters the default configuration of the CUBE from not distinguishing SIP Delayed-Offer to Early-Offer call flows, to forcing the CUBE to generate an Early-Offer with the configured codecs for an incoming Delayed-Offer INVITE. This is required for RFC 2833 in-band DTMF from the CUCM to interwork with Cox’s service
[7] This command is for SIP-to-SIP calls only

Dial-Peers are used to provide both inbound and outbound call legs. Each call utilizes two peers. In the event the primary path is not available, preference 1 dial-peer is used. Preference 0 is the default and therefore does not appear in Cisco IOS

```
dial-peer voice 1 voip[8]
 destination-pattern 67823837..
 session protocol sipv2
 session target ipv4:10.70.10.2
 voice-class codec 1
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
 ip qos dscp cs5 media
 ip qos dscp cs4 signaling
 no vad
!
dial-peer voice 10 voip[9]
 destination-pattern 1[2-9]..[2-9].....
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
 ip qos dscp cs5 media
 ip qos dscp cs4 signaling
 no vad
!
dial-peer voice 11 voip[10]
 translation-profile outgoing PSTN_Outgoing
 preference 1
 destination-pattern [2-9]..[2-9].....
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
 ip qos dscp cs5 media
 ip qos dscp cs4 signaling
 no vad

dial-peer voice 12 voip
 translation-profile outgoing PSTN_Outgoing
 preference 1
 destination-pattern 011T
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
```

[8] This dial-peer is for all incoming calls
[9] This dial-peer is for outgoing long distance calls
[10] This dial-peer is for outgoing local calls

```
 ip qos dscp cs5 media
 ip qos dscp cs4 signaling
 no vad

dial-peer voice 13 voip
 description **0 or 00 calls to Local or International Operator**
 translation-profile outgoing PSTN_Outgoing
 preference 1
 destination-pattern 0T
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
 ip qos dscp cs5 media
 ip qos dscp cs3 signaling
 no vad

dial-peer voice 14 voip
 description **CCA*North American-10-Digit*Service Numbers**
 translation-profile outgoing PSTN_Outgoing
 preference 1
 destination-pattern [2-9]11
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
 ip qos dscp cs3 signaling
 no vad

dial-peer voice 2003 voip
 description **CCA*North American-10-Digit*Service Numbers**
 translation-profile outgoing PSTN_Outgoing
 preference 1
 destination-pattern [2-9]11
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
 ip qos dscp cs3 signaling
 no vad


dial-peer voice 2005 voip
 description **0 or 00 calls to Local or International Operator**
 translation-profile outgoing PSTN_Outgoing
 preference 1
```

```
 destination-pattern 0T
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
 ip qos dscp cs3 signaling
 no vad

dial-peer voice 2006 voip
 description **Dial Carrier Access Code**
 translation-profile outgoing PSTN_Outgoing
 preference 1
 destination-pattern 101....1[2-9]..[2-9].....
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 no voice-class sip early-offer forced
 dtmf-relay rtp-nte
 ip qos dscp cs3 signaling
 no vad
```

The Codec Preference determines the priority of the entries

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

6.3.3 Cisco Unified Communications Manager (CUCM)

The following sections detail the configuration of the CUCM.

6.3.4 Enterprise Parameters

The Enterprise Parameters Configuration section defines the settings for IP phone service. The Cluster ID parameter value will be a custom value. All of the remaining values are default settings.

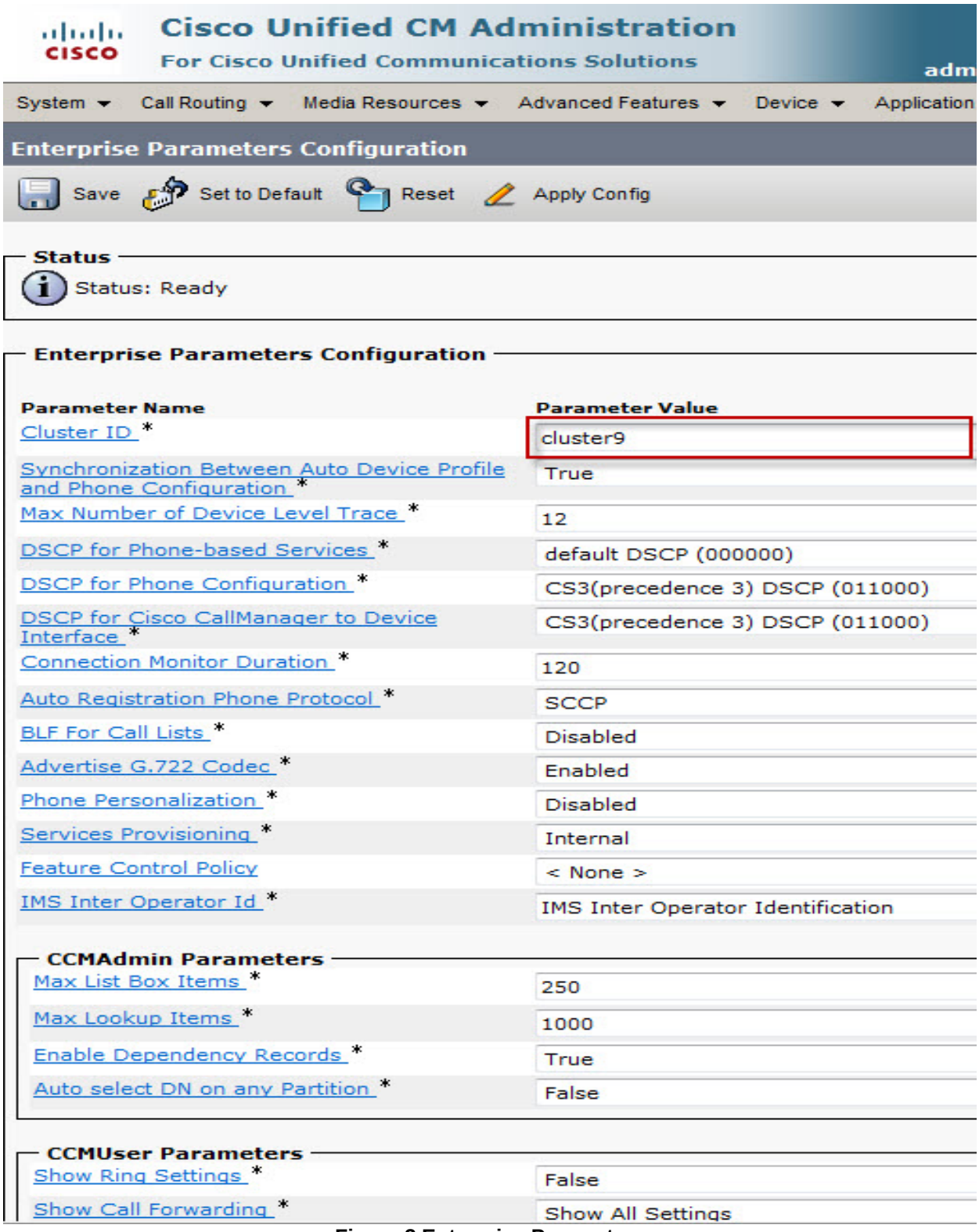


Figure 3 Enterprise Parameters

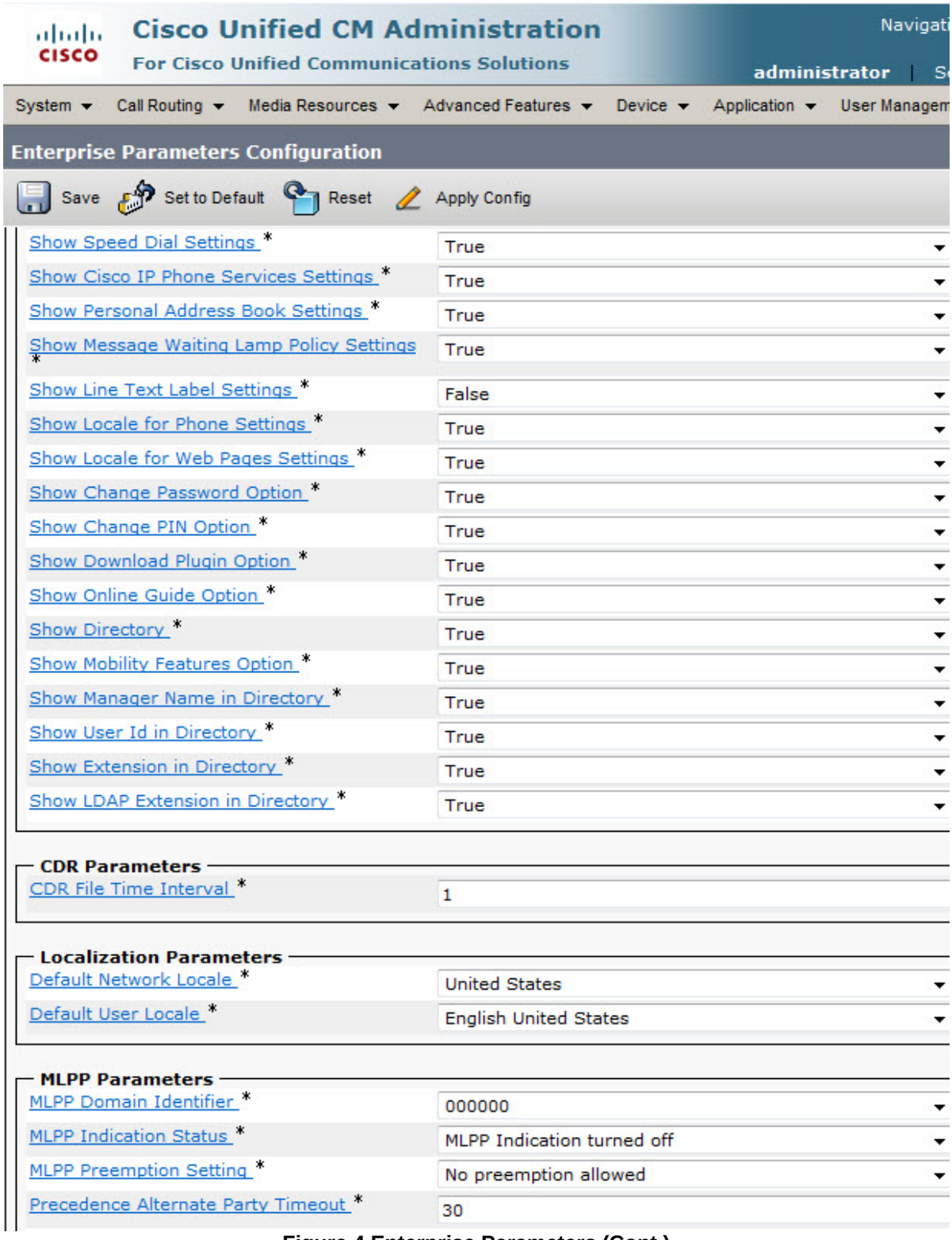


Figure 4 Enterprise Parameters (Cont.)

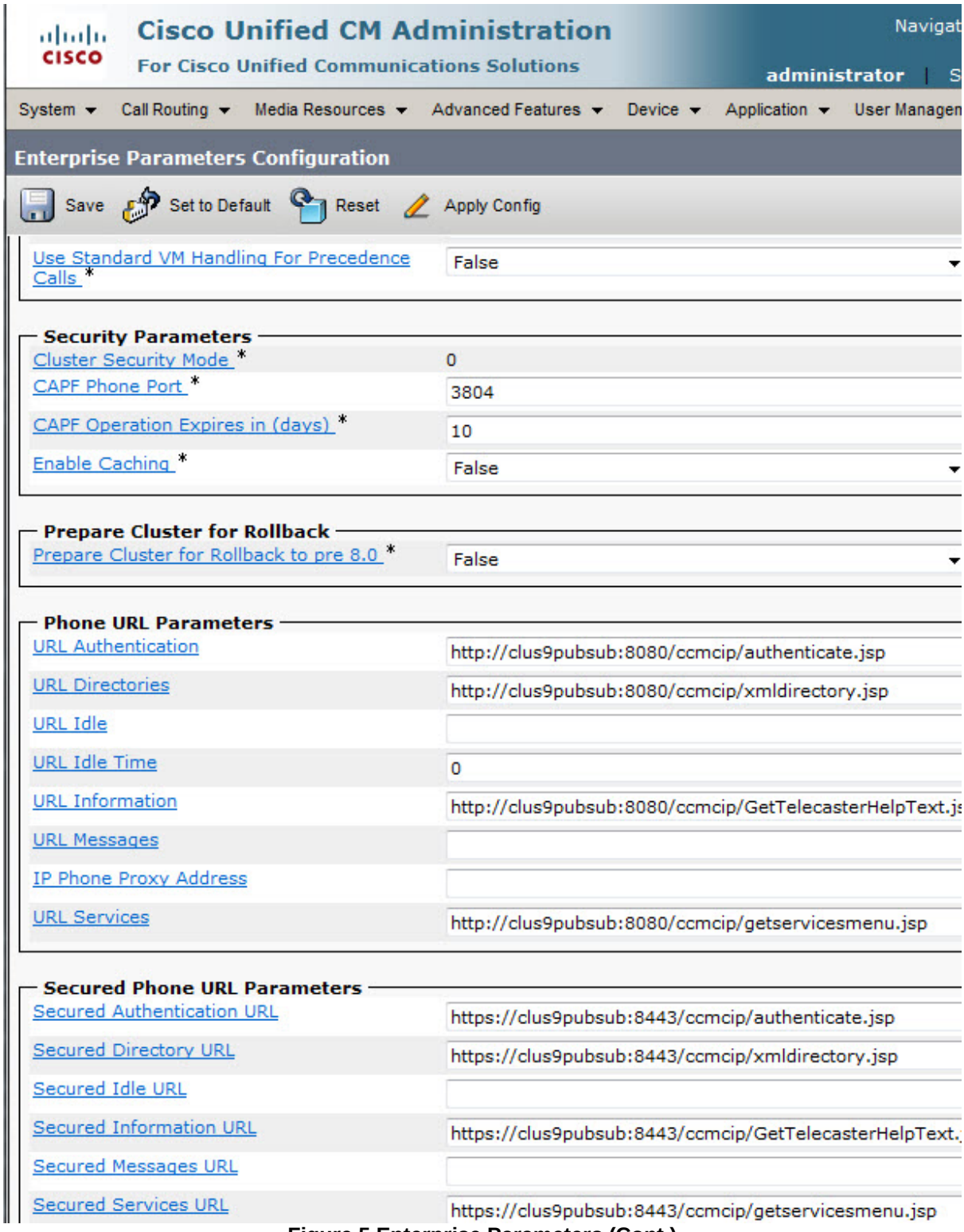


Figure 5 Enterprise Parameters (Cont.)

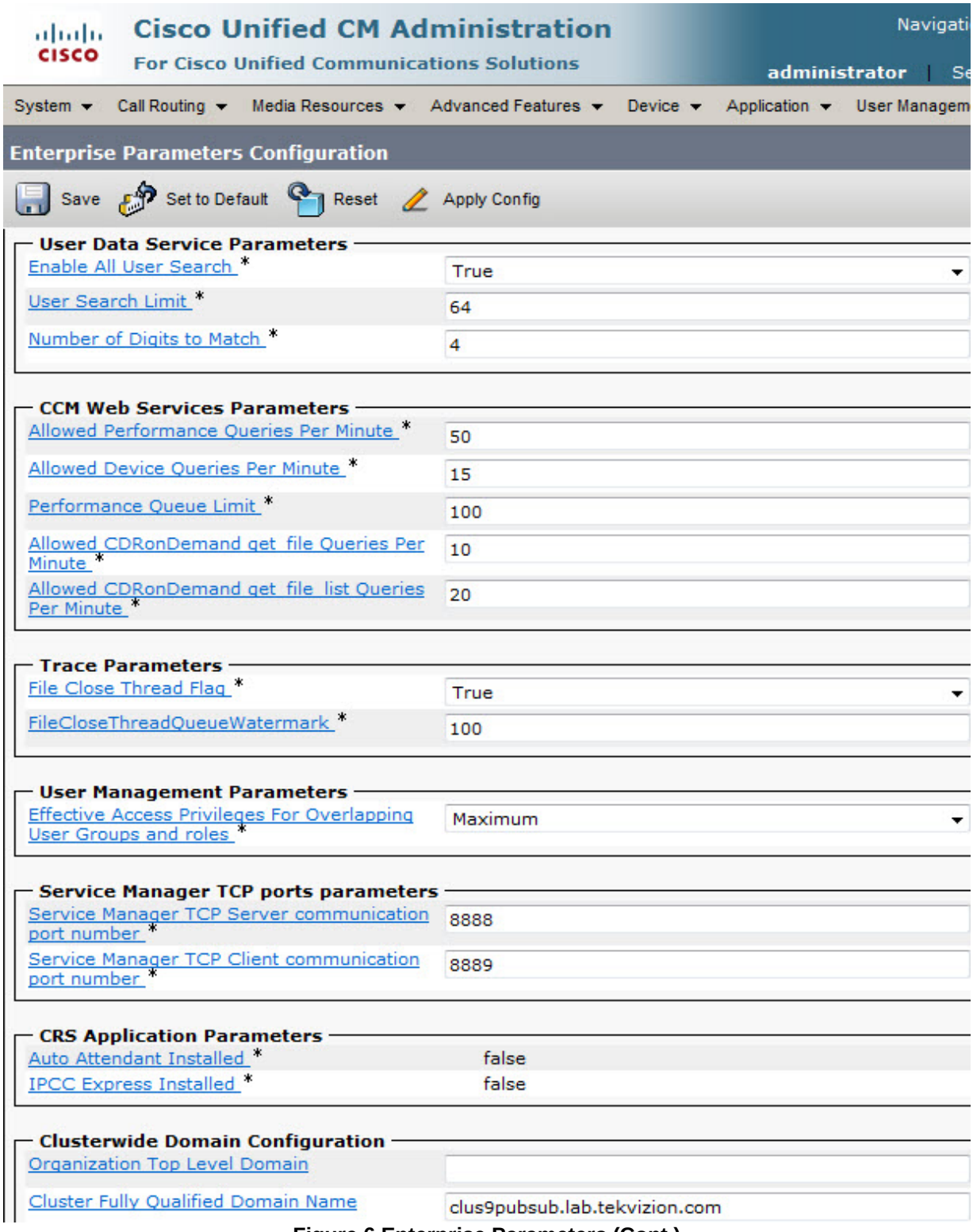


Figure 6 Enterprise Parameters (Cont.)

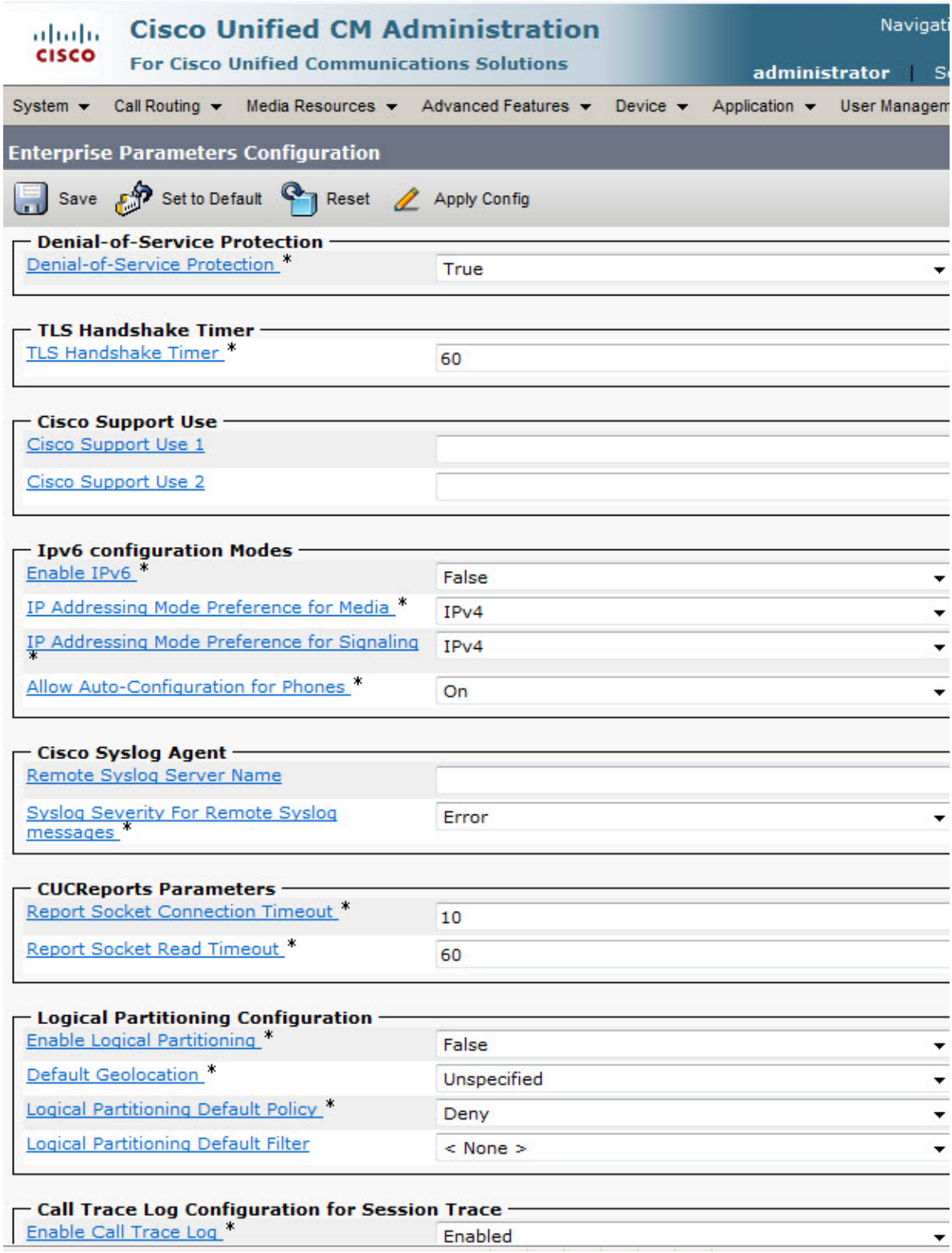


Figure 7 Enterprise Parameters (Cont.)

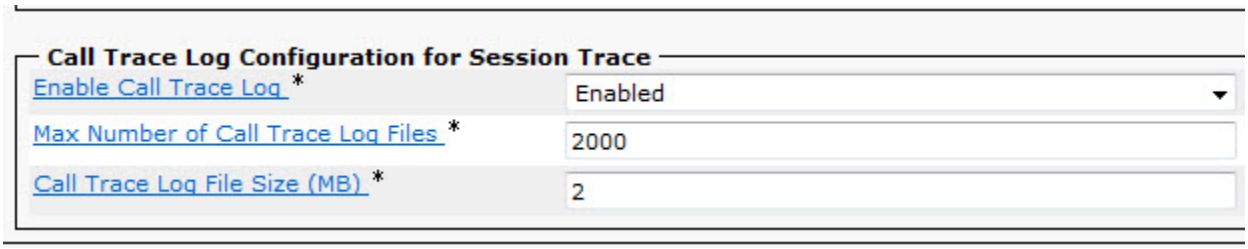


Figure 8 Enterprise Parameters (Cont.)

6.3.5 Service Parameters

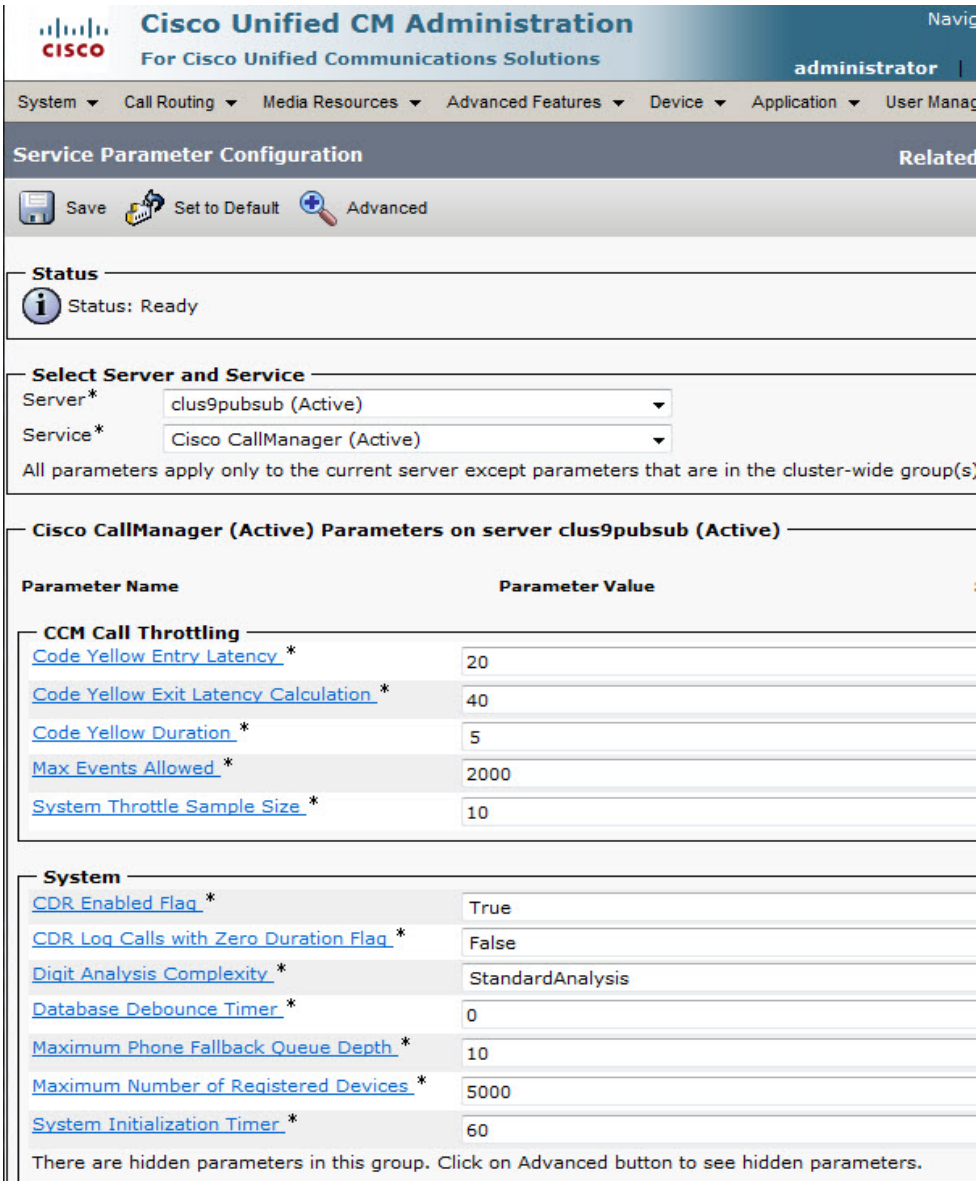


Figure 9 Service Parameters

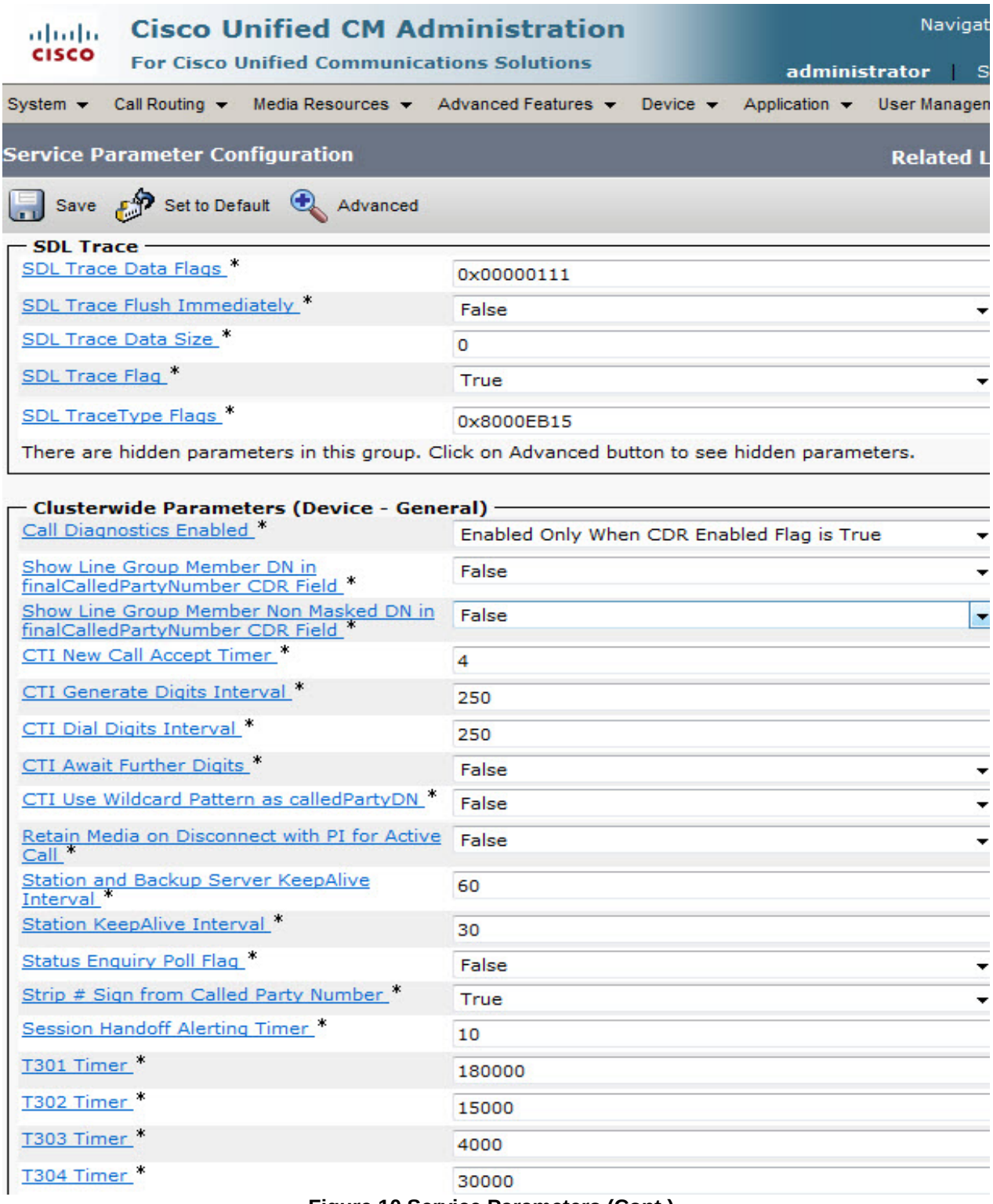


Figure 10 Service Parameters (Cont.)

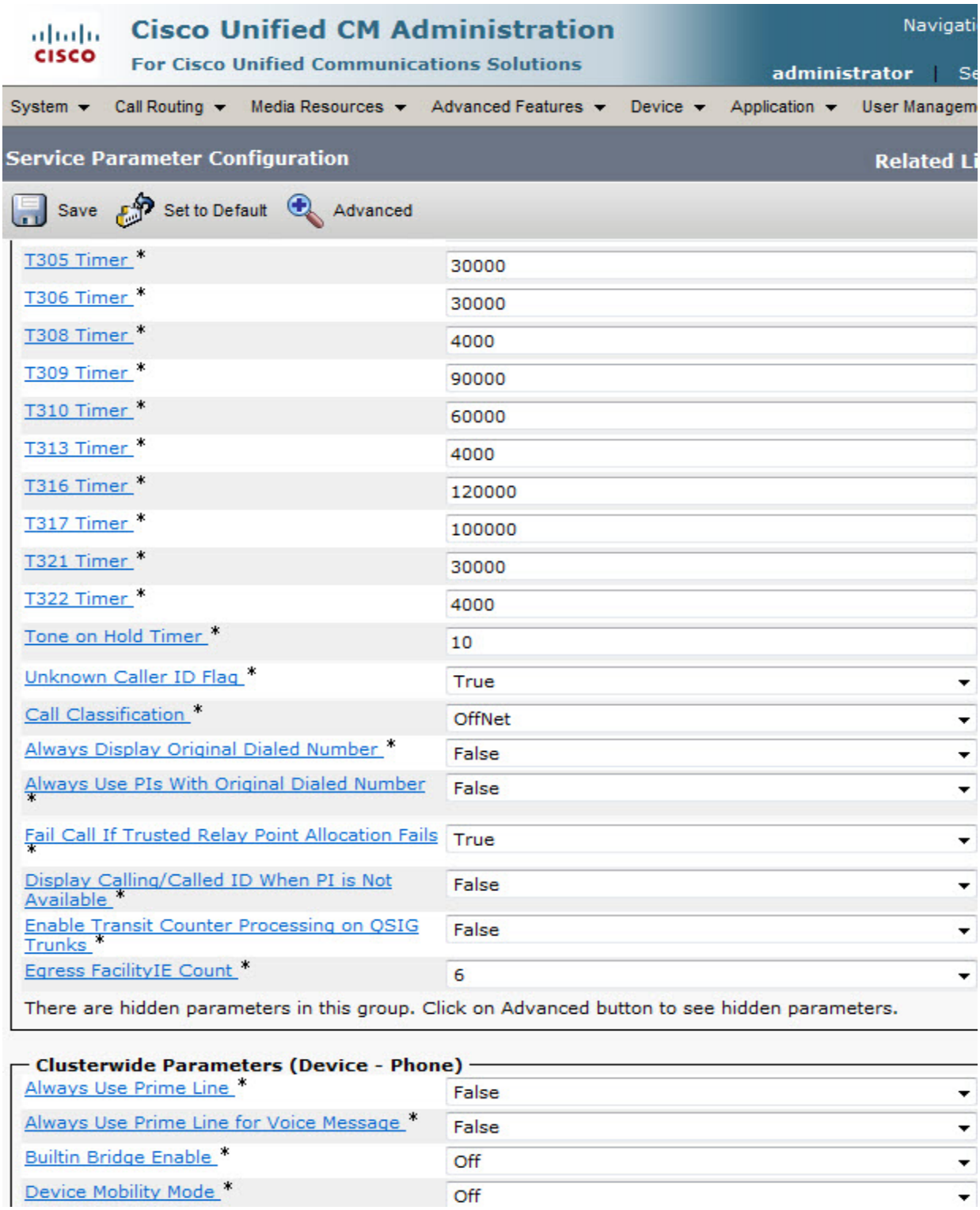


Figure 11 Service Parameters (Cont.)

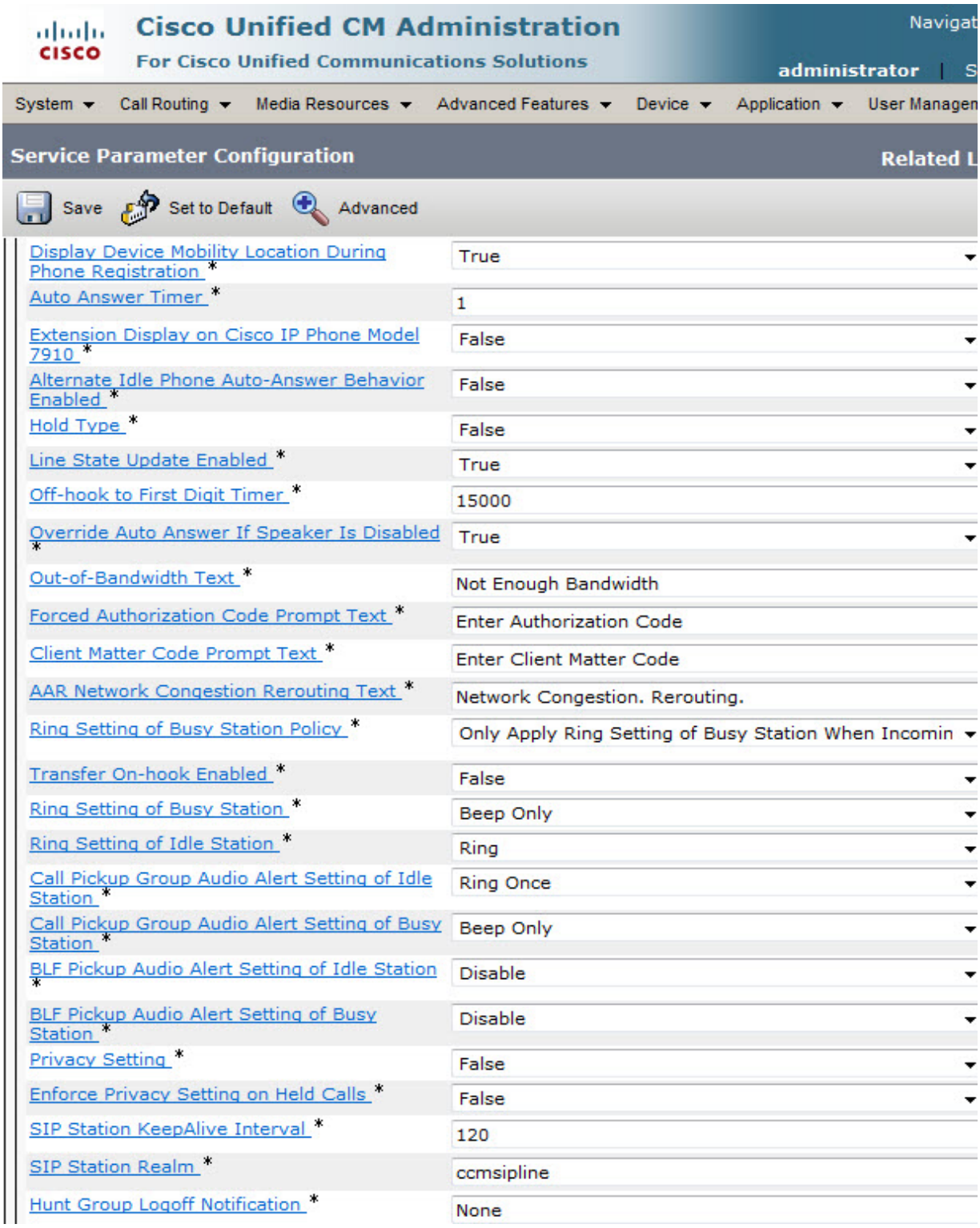


Figure 12 Service Parameters (Cont.)

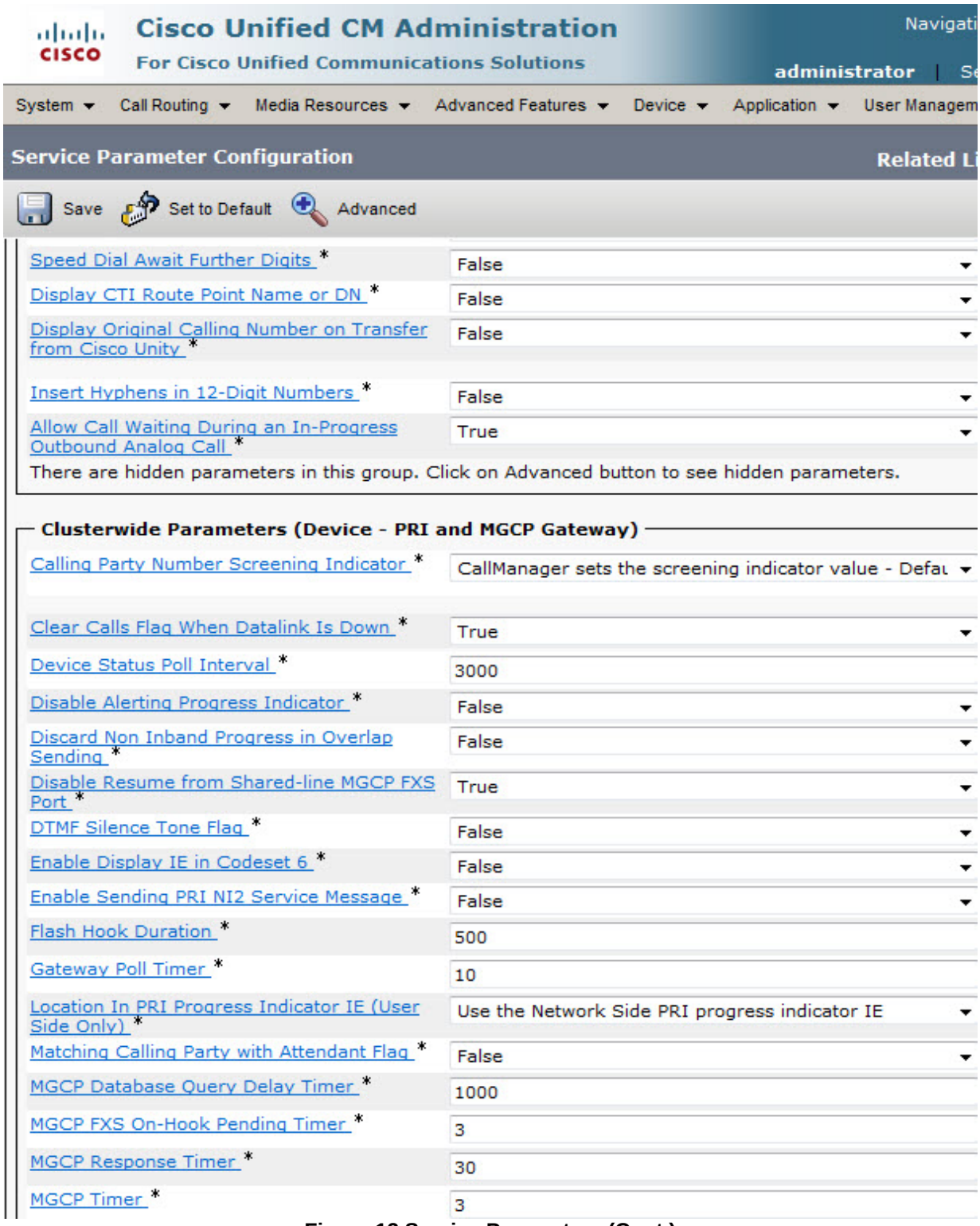


Figure 13 Service Parameters (Cont.)

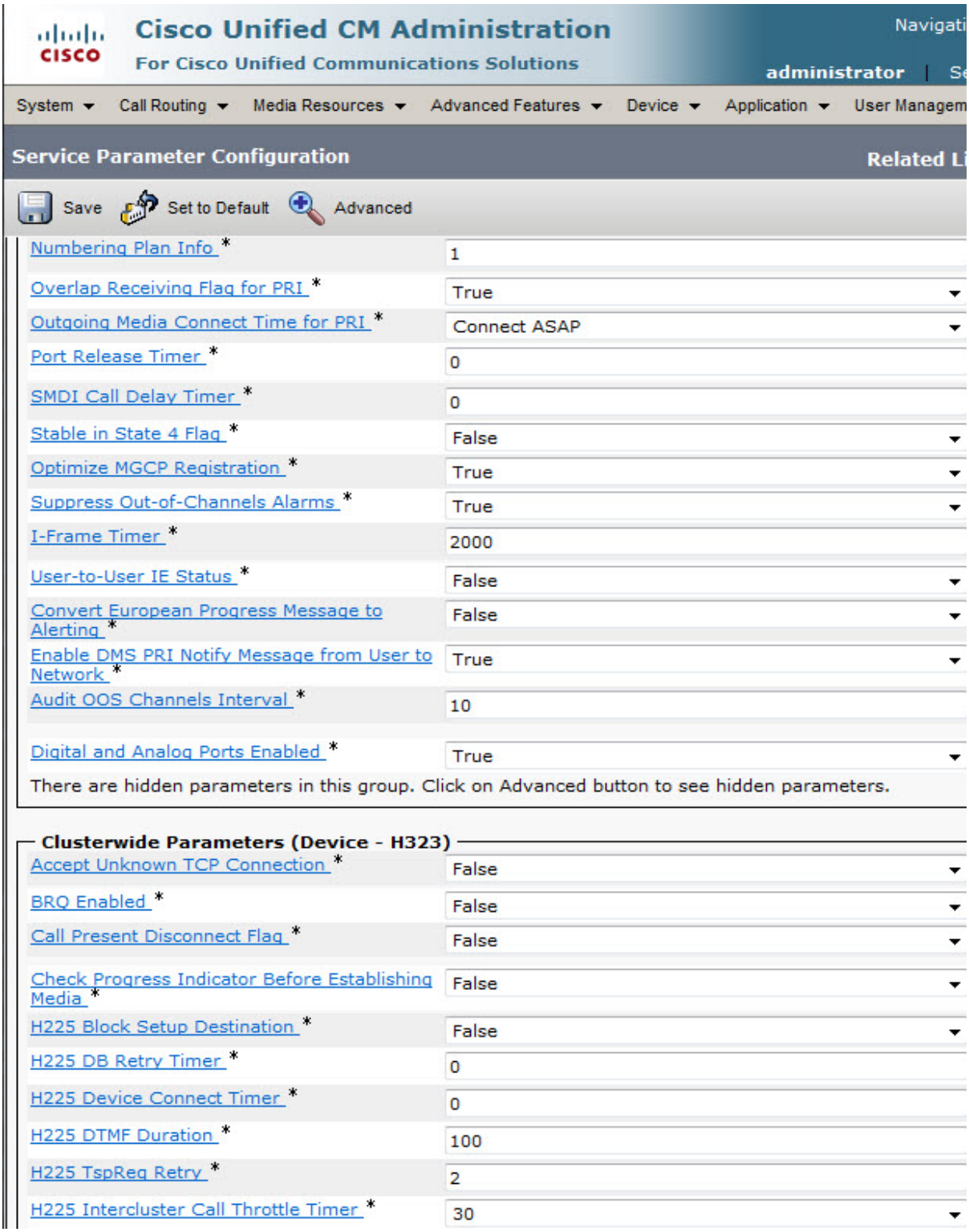


Figure 14 Service Parameters (Cont.)

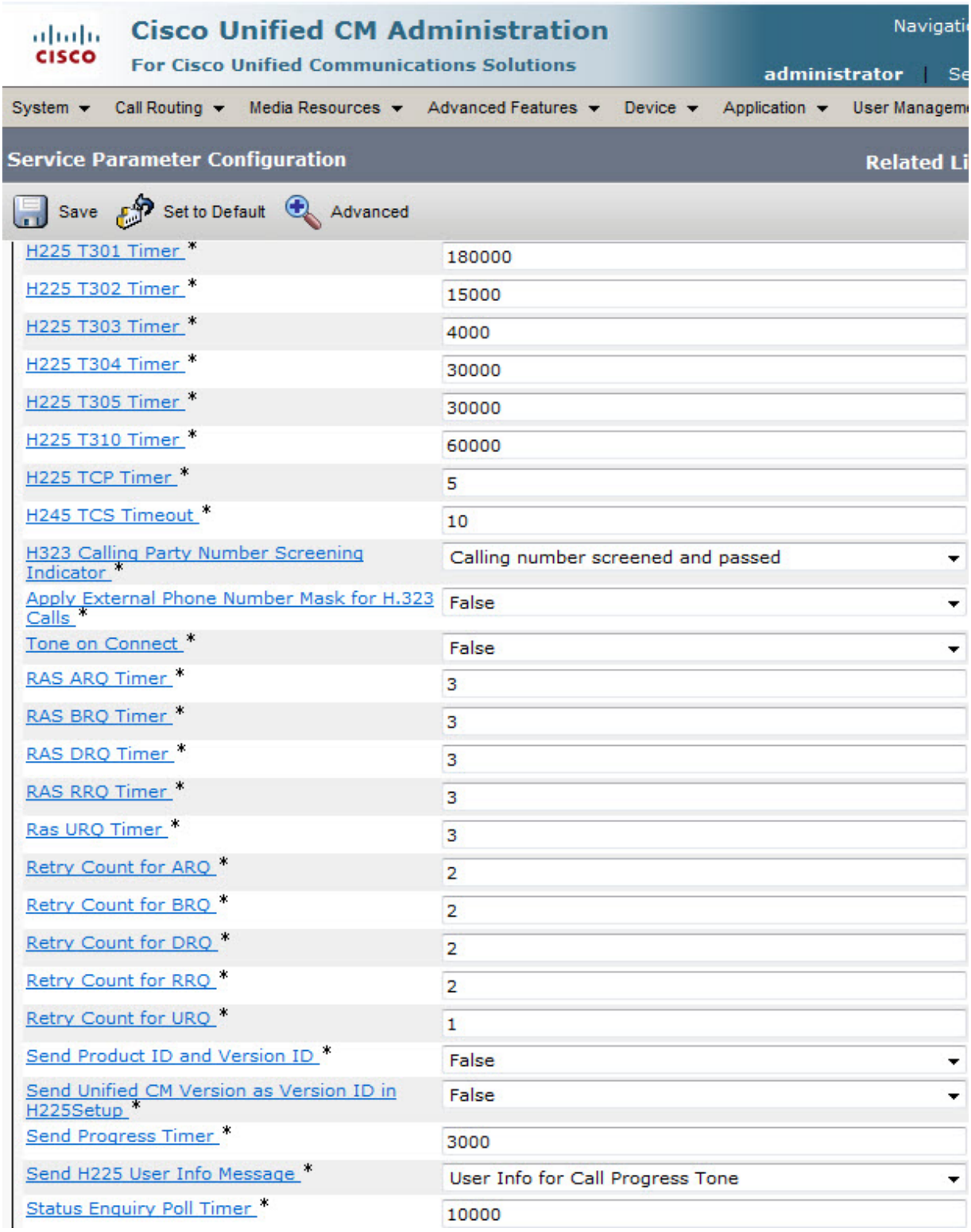


Figure 15 Service Parameters (Cont.)

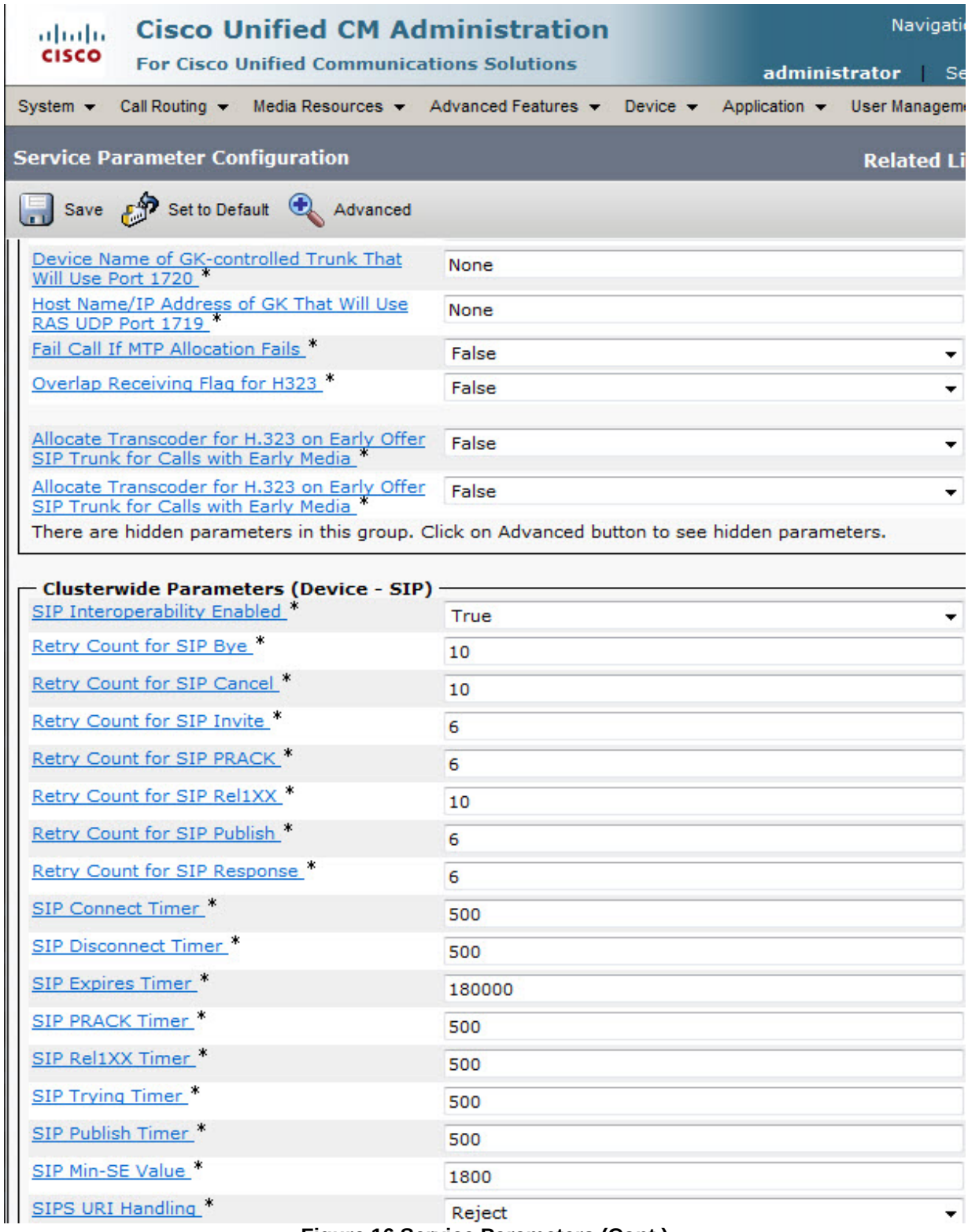


Figure 16 Service Parameters (Cont.)

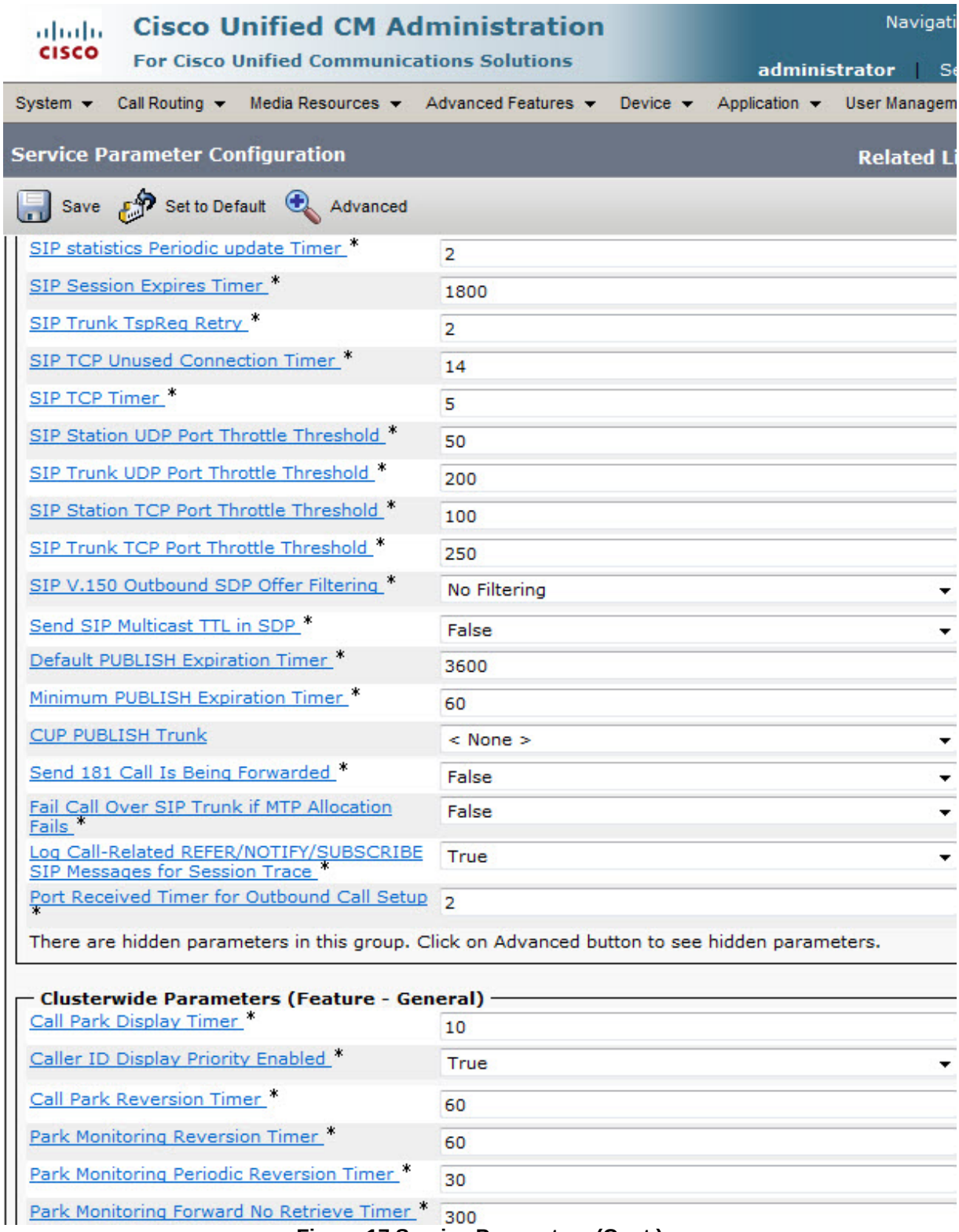


Figure 17 Service Parameters (Cont.)

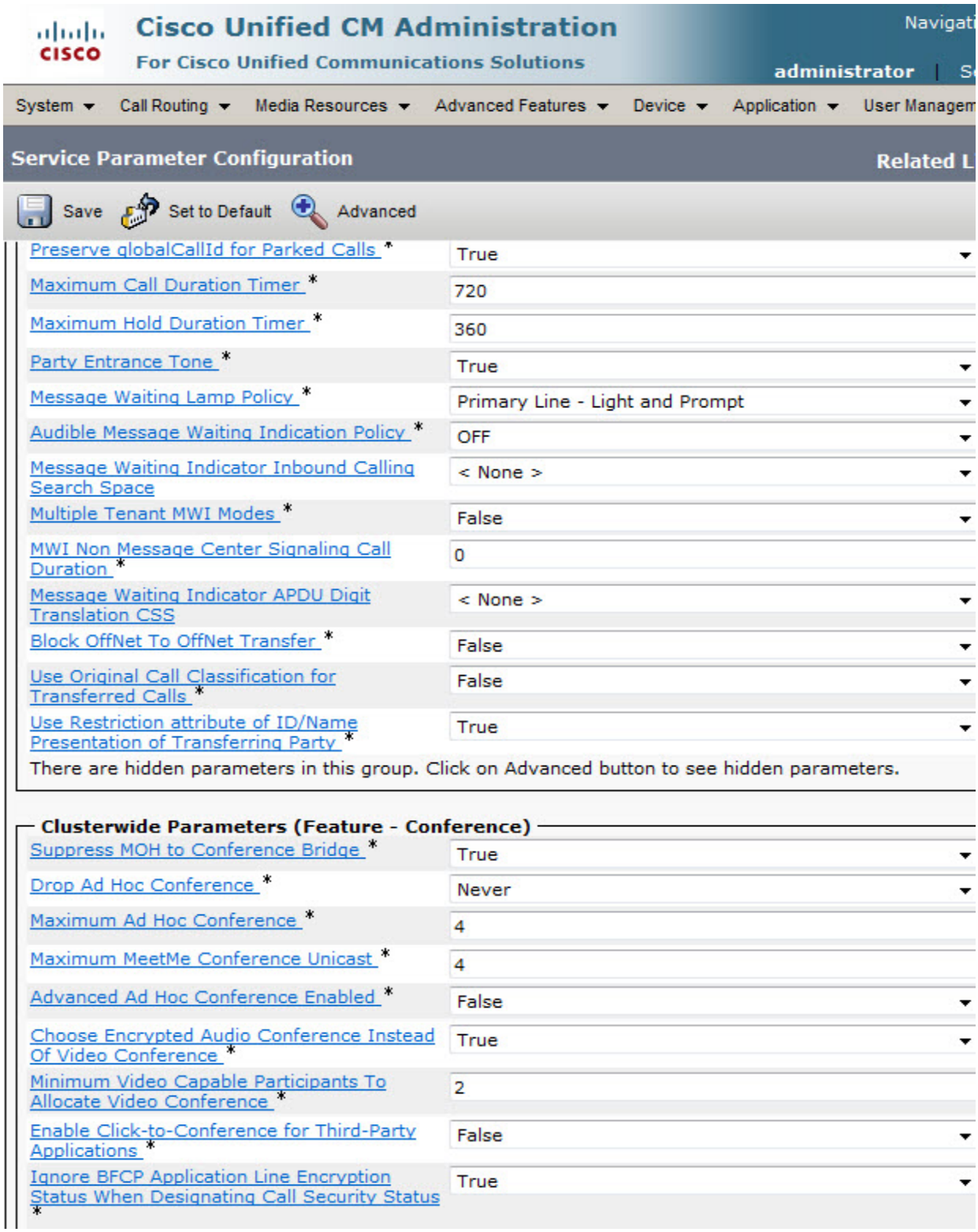


Figure 18 Service Parameters (Cont.)

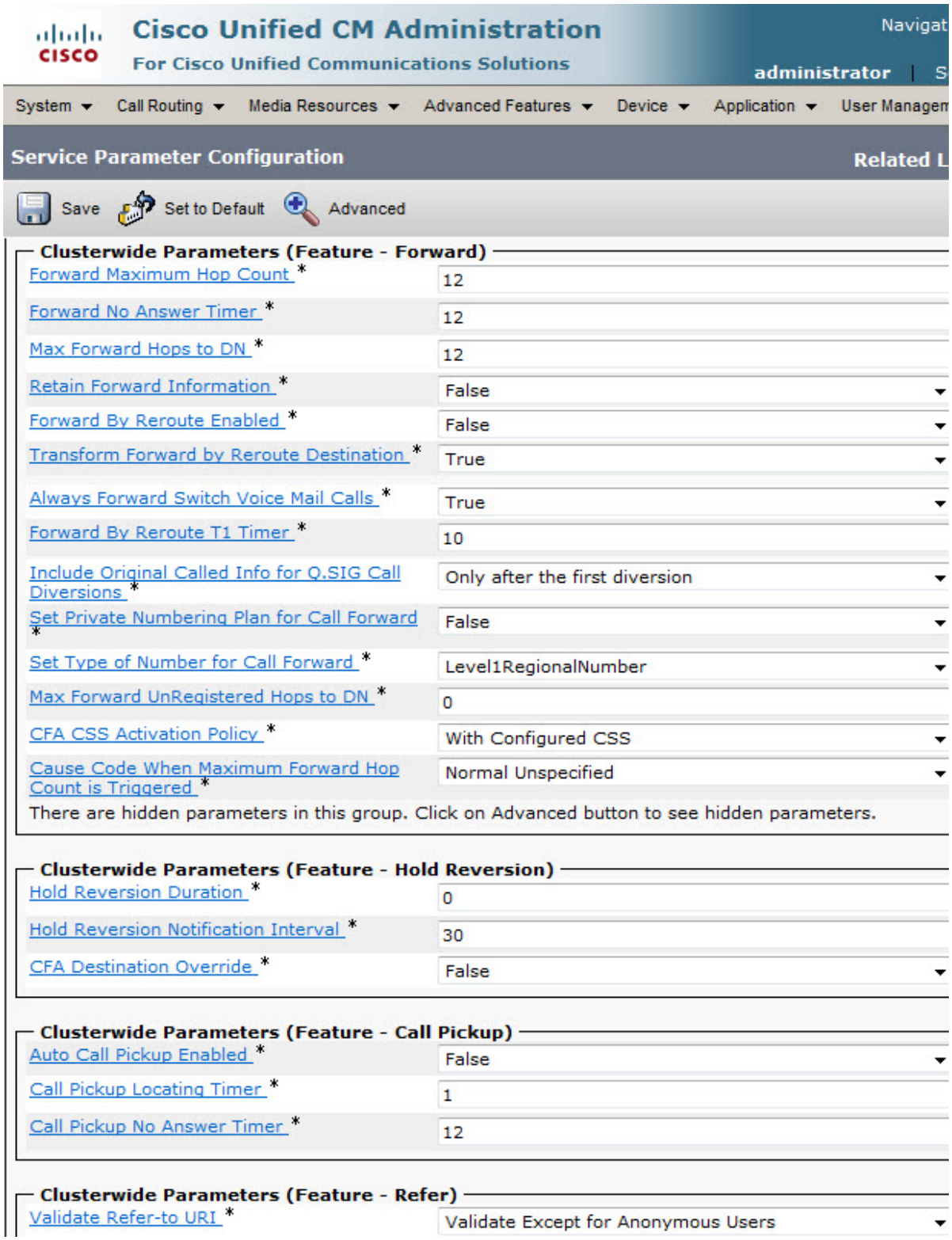


Figure 19 Service Parameters (Cont.)

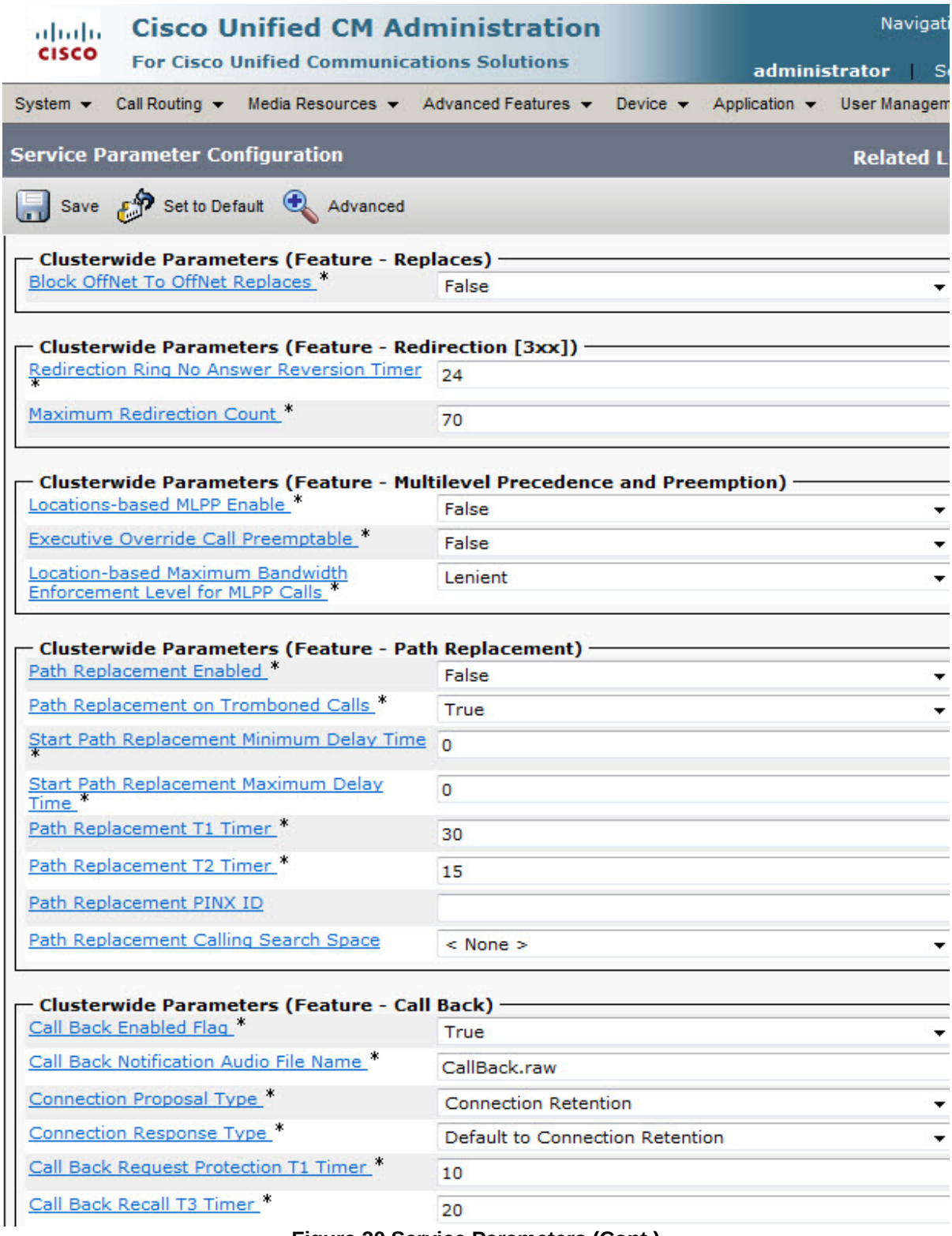


Figure 20 Service Parameters (Cont.)

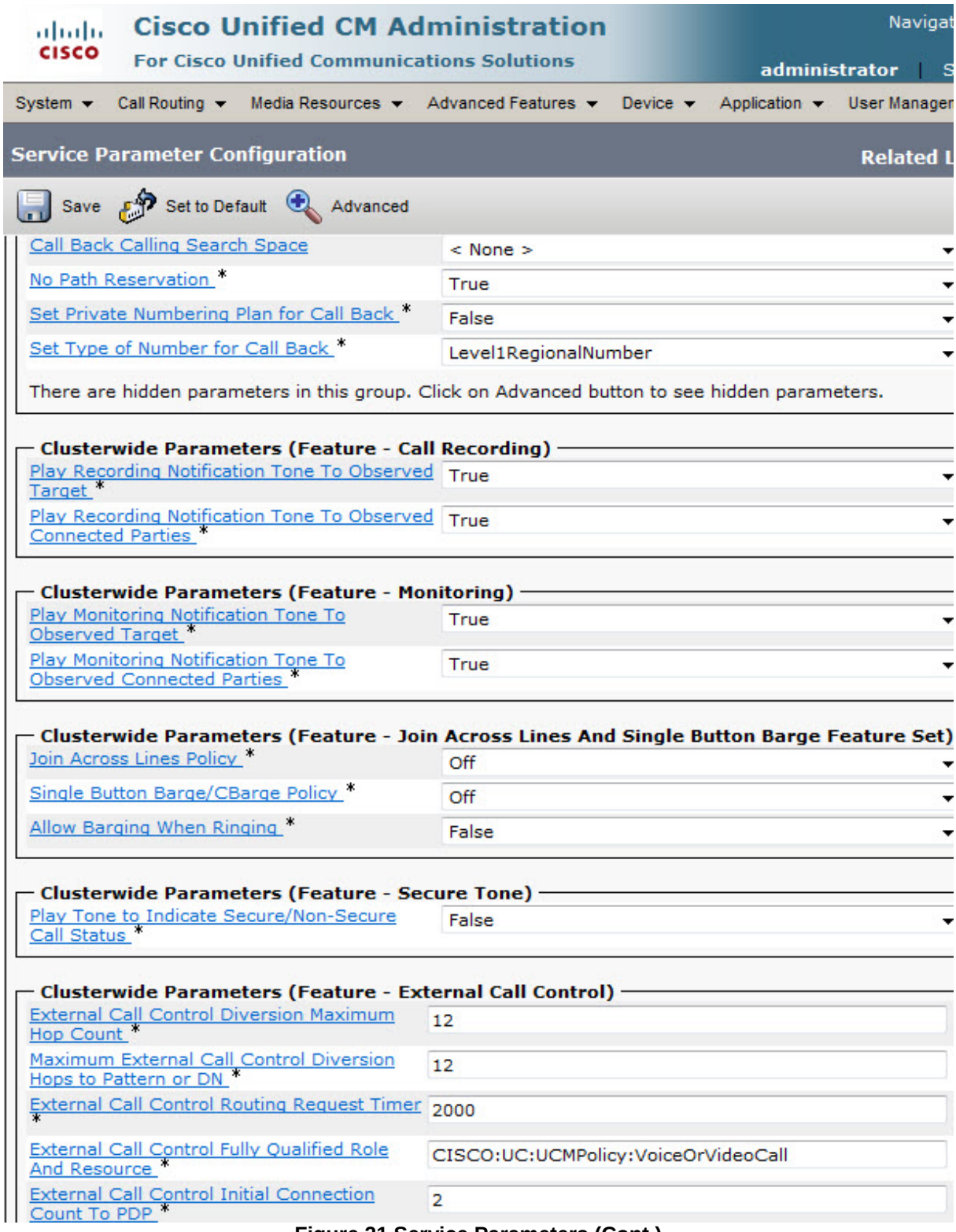


Figure 21 Service Parameters (Cont.)

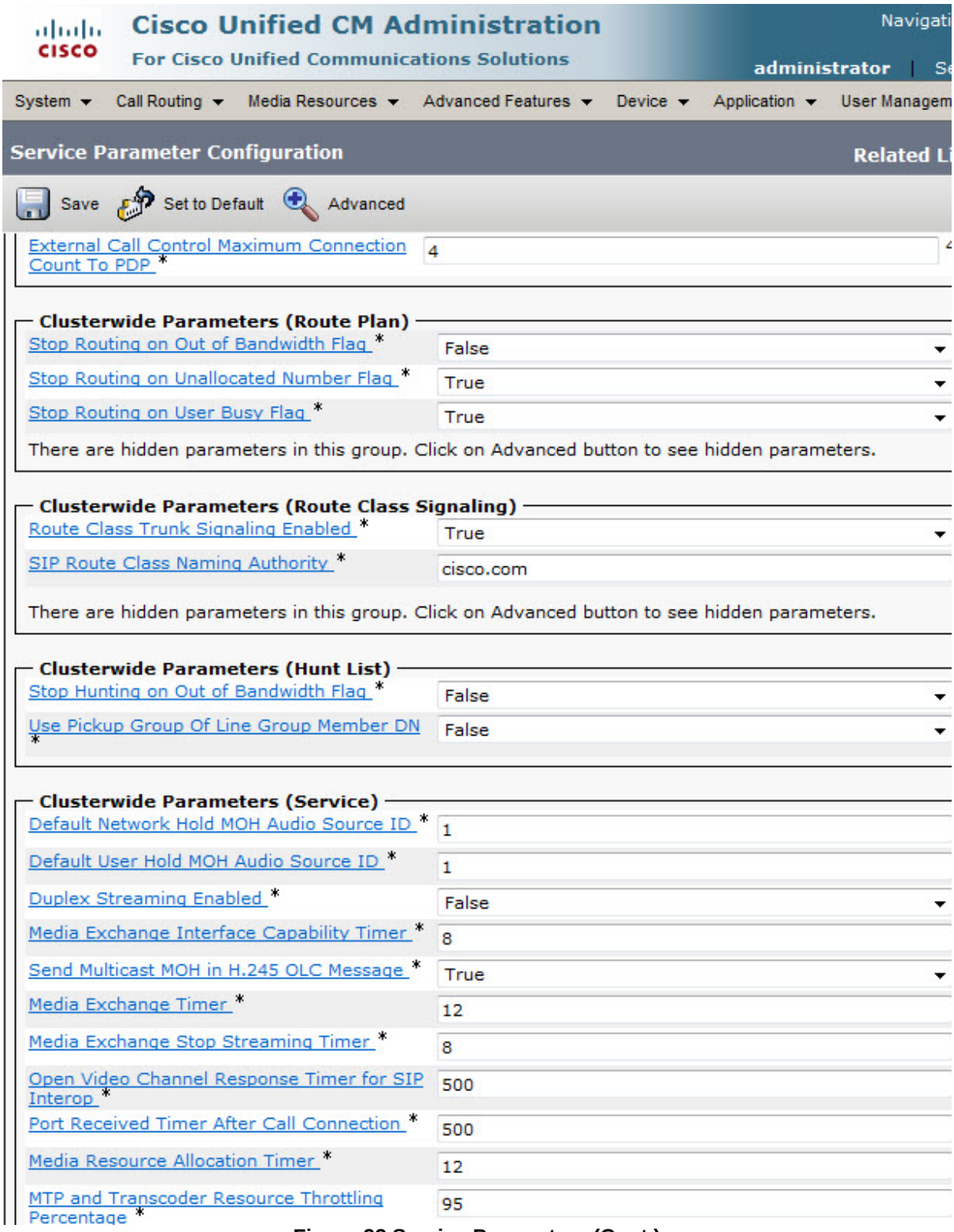


Figure 22 Service Parameters (Cont.)

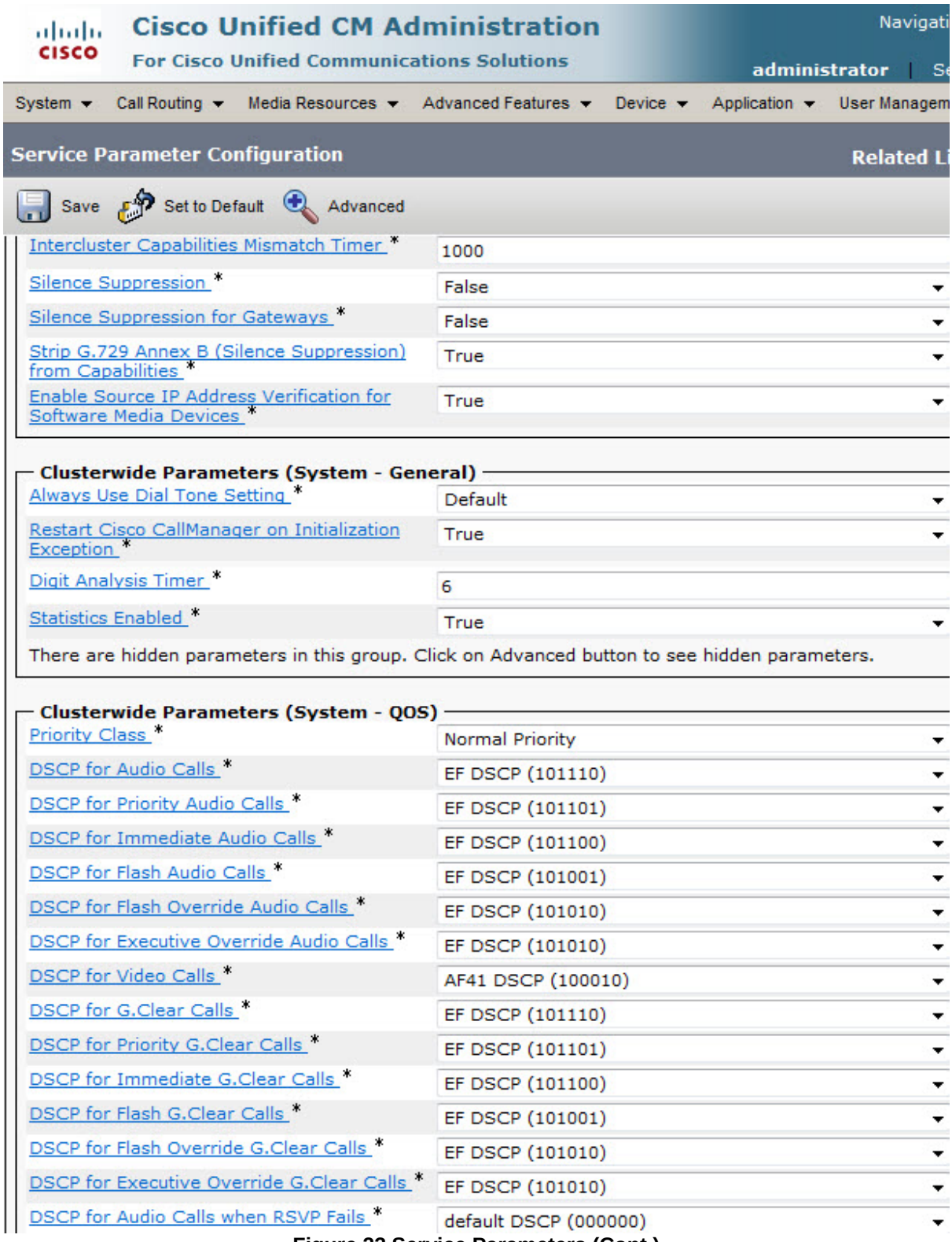


Figure 23 Service Parameters (Cont.)

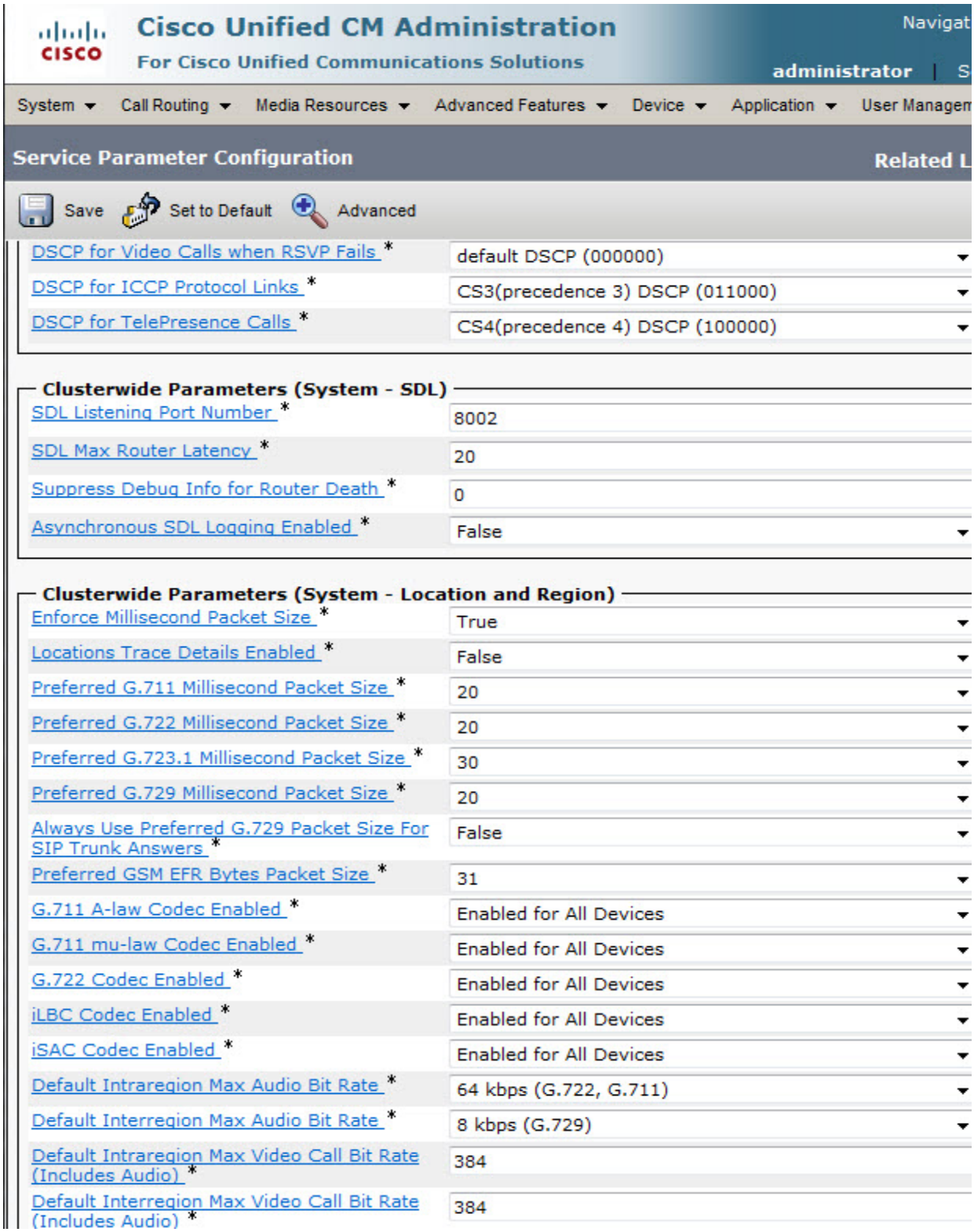


Figure 24 Service Parameters (Cont.)

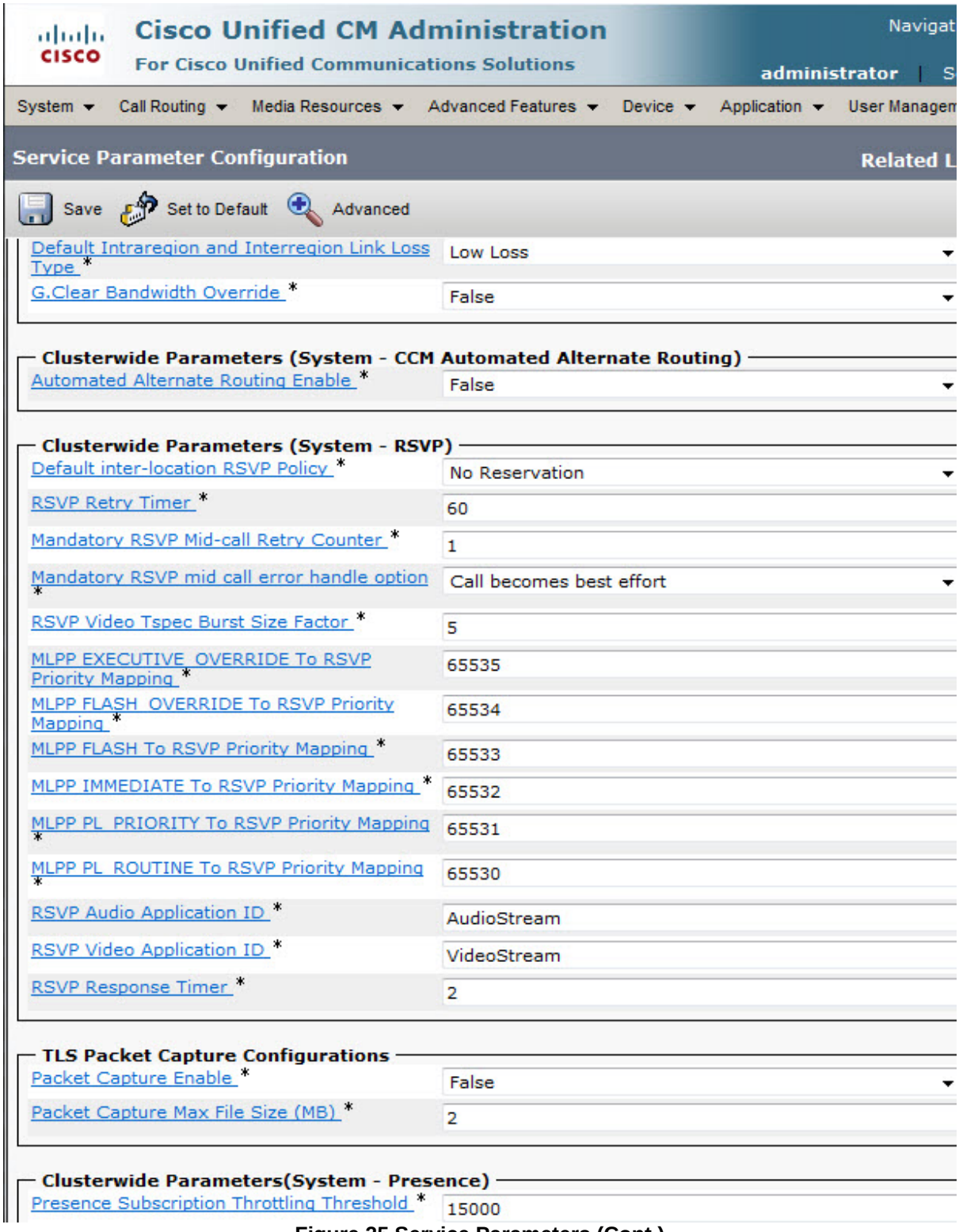


Figure 25 Service Parameters (Cont.)

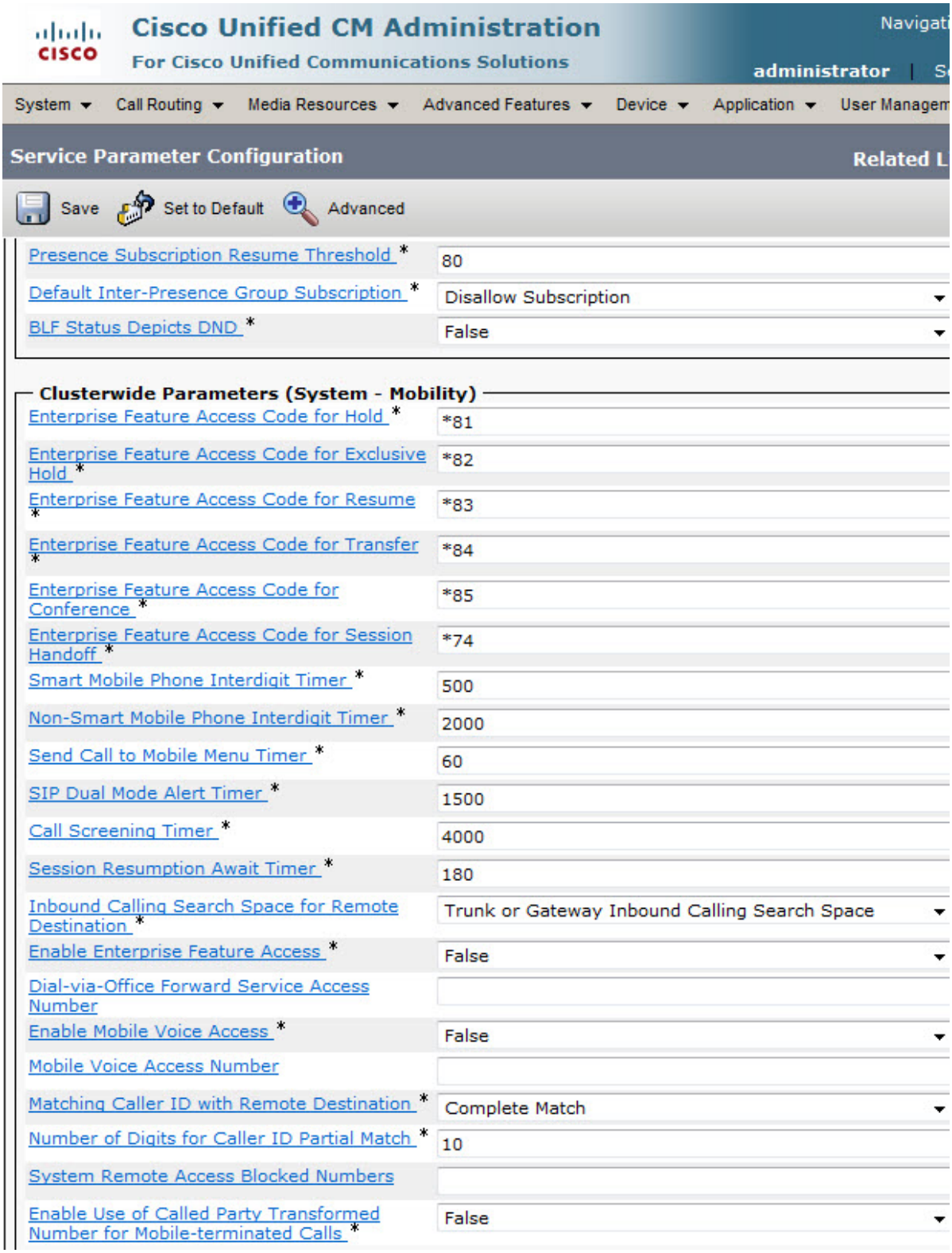


Figure 26 Service Parameters (Cont.)

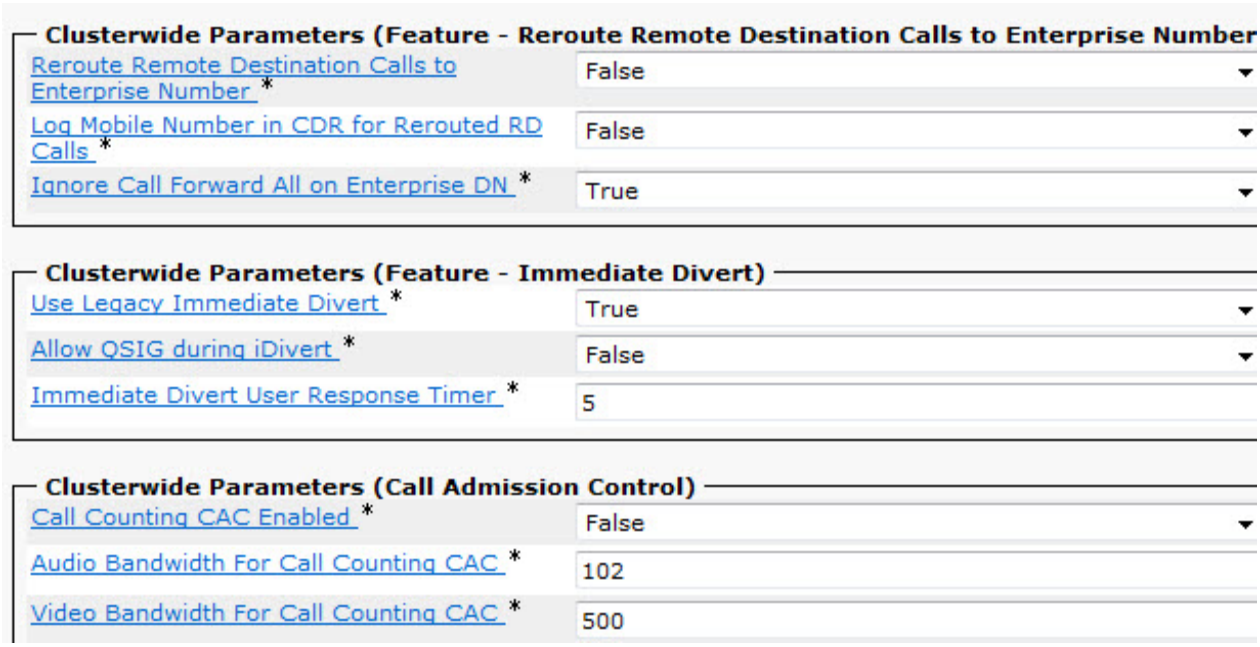


Figure 27 Service Parameters (Cont.)

6.3.6 Device Pools

Device pools are used to define sets of common characteristics for devices.

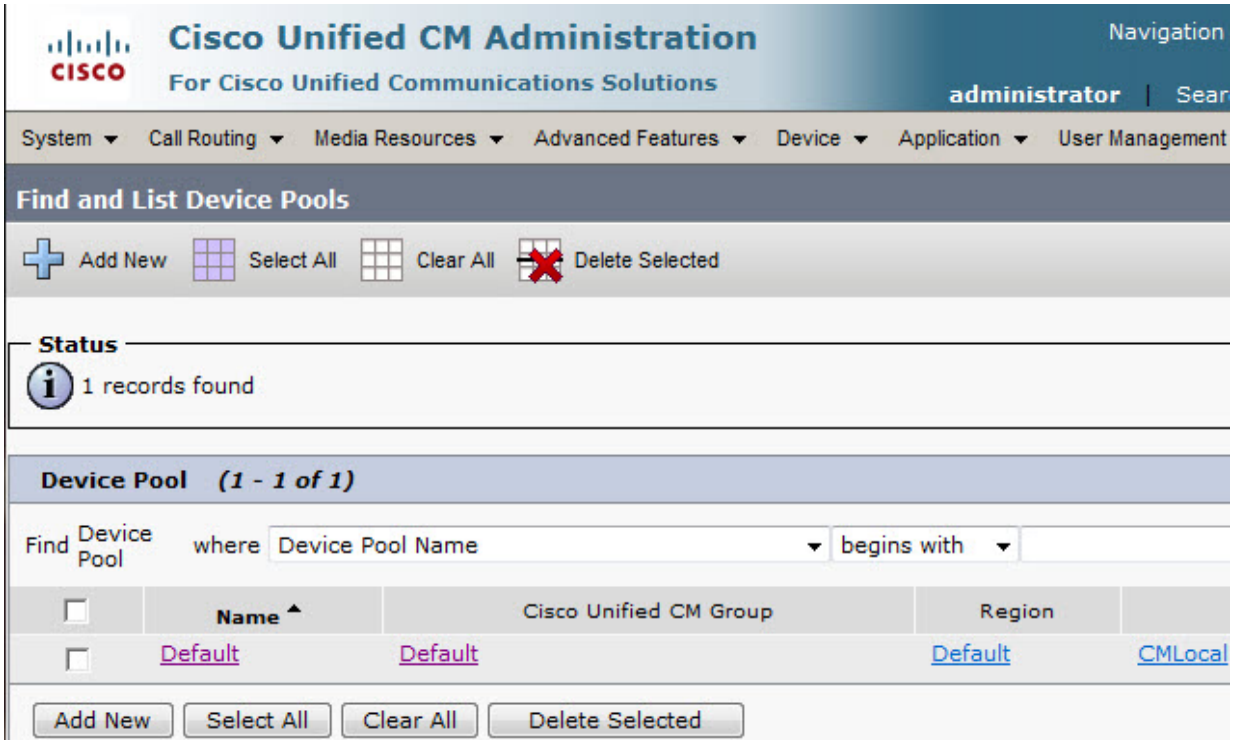


Figure 28 Device Pools

6.3.7 SIP Trunks

SIP Trunk configuration is described below. Figure 29 shows a list of active trunks. This section will describe the procedures for adding a new trunk.

1. Navigate to Device > Trunk > Add New
2. In the Trunk Type menu, select SIP Trunk (Figure 30)
3. The Device Protocol (SIP) and Trunk Service Type None (Default) are default settings.
4. Select Next
5. Assign a Device Name: Cox
6. Set the Device Pool to Default
7. In the Inbound Calls section, Define the Significant Digits: 4
8. In the SIP Information section, assign a Destination Address: 10.70.10.15 (This is the LAN address of the CUBE)
9. In the SIP Trunk Security Profile drop-down menu, select Non Sec UDP SIP Trunk Profile (Refer to Section 6.3.9 for more details)
10. In the SIP Profile drop-down menu, select Cox SIP Profile (SIP Profiles are described in more detail in Section 6.3.8)
11. For the DTMF Signaling Method, select RFC 2833
12. Select Save

NOTE: When route patterns are assigned they will show up in the trunk screen.

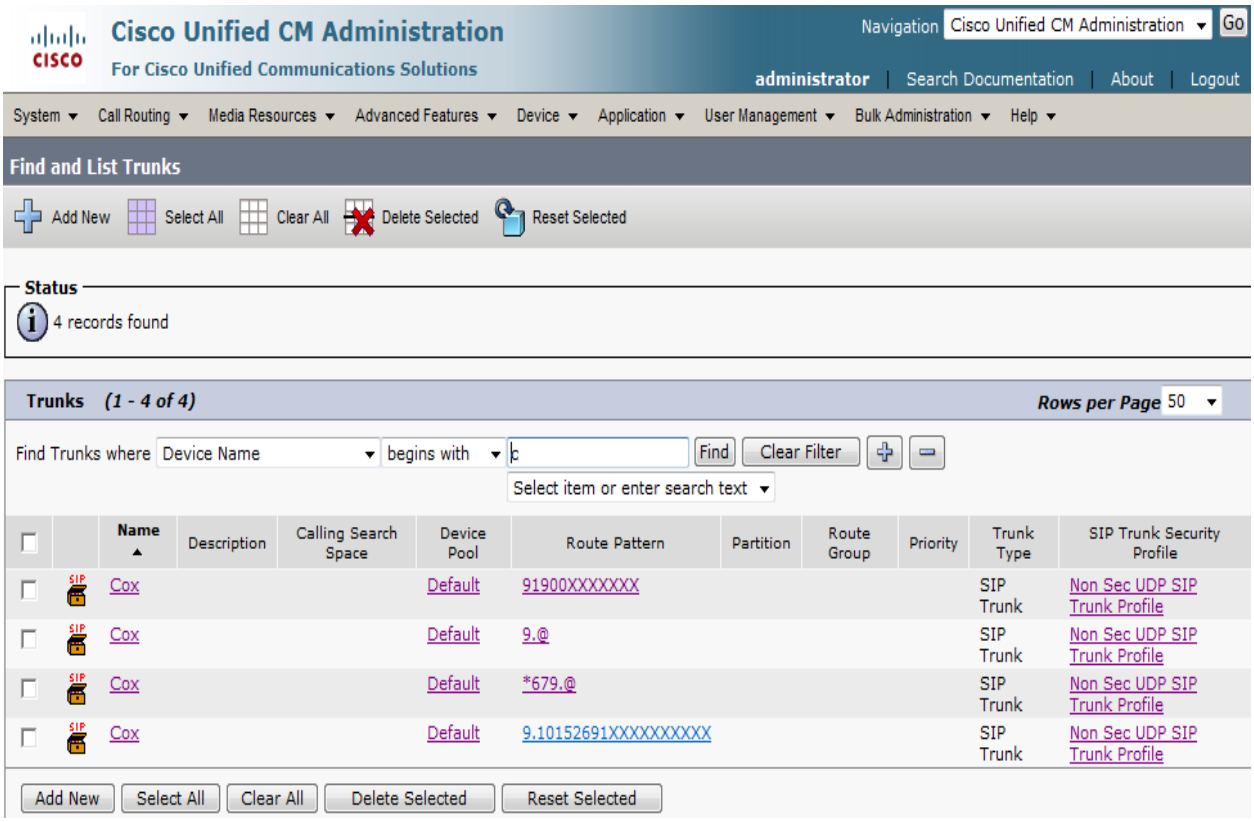


Figure 29 SIP Trunks

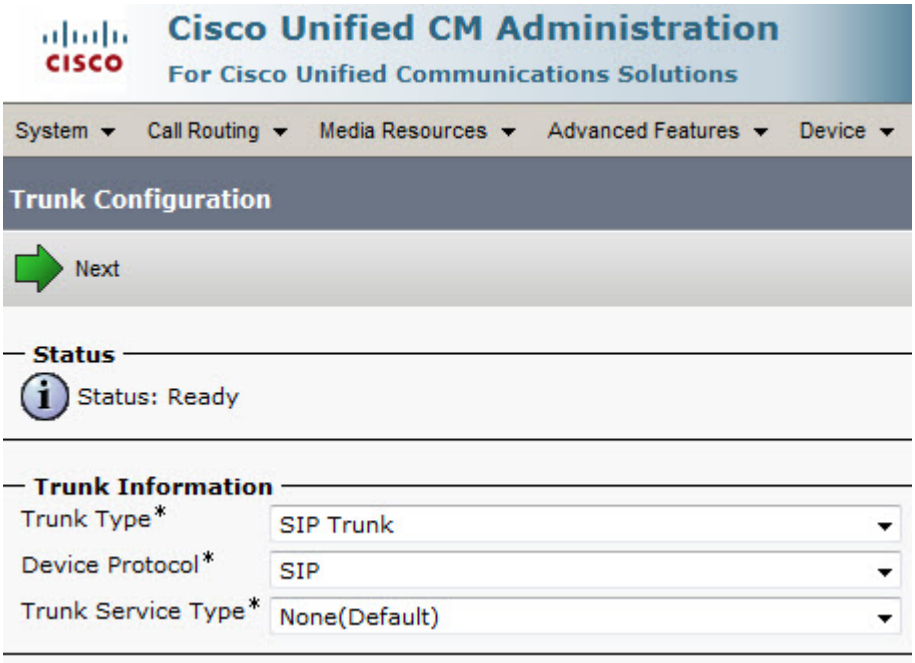


Figure 30 SIP Trunk (Cont.)

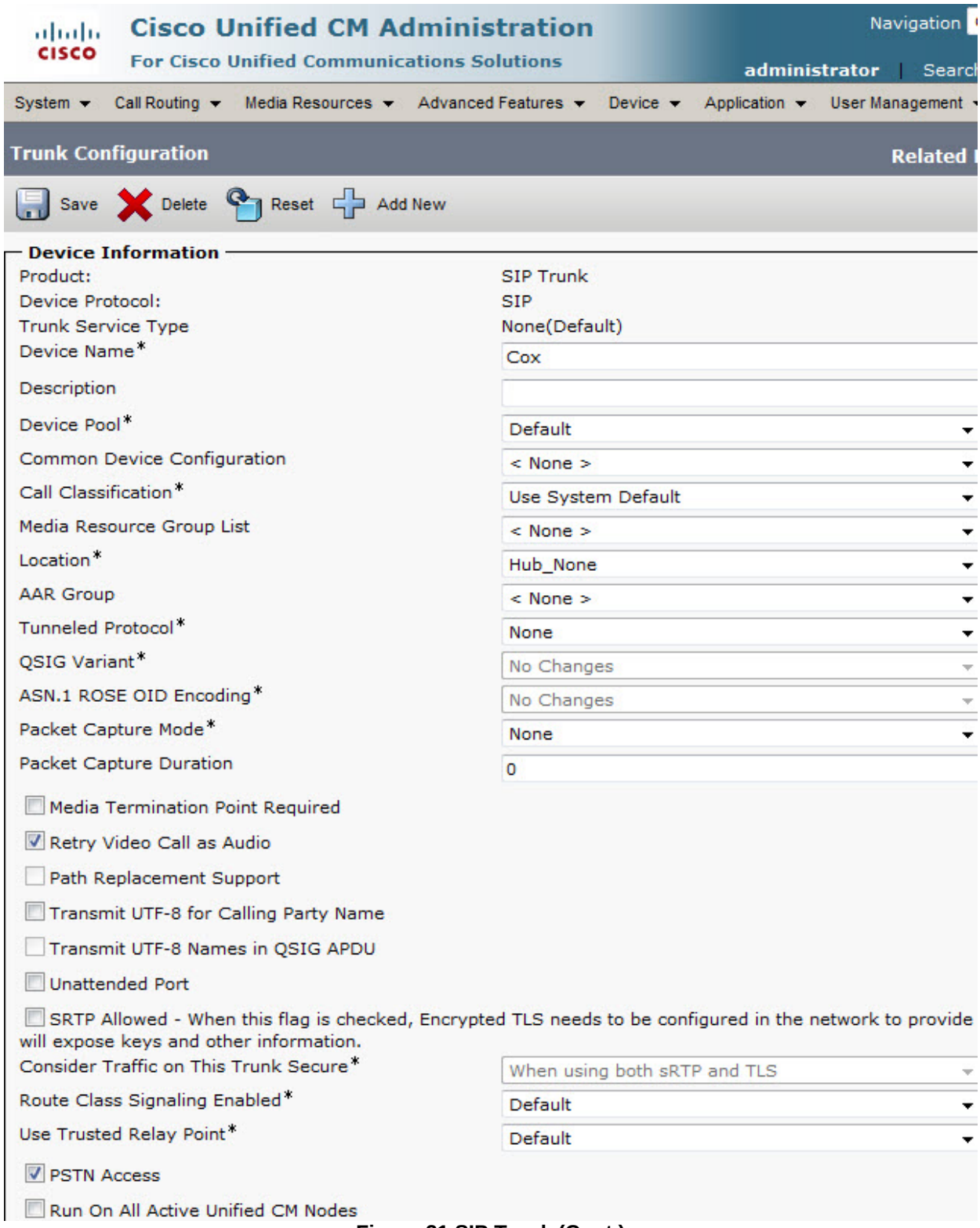


Figure 31 SIP Trunk (Cont.)

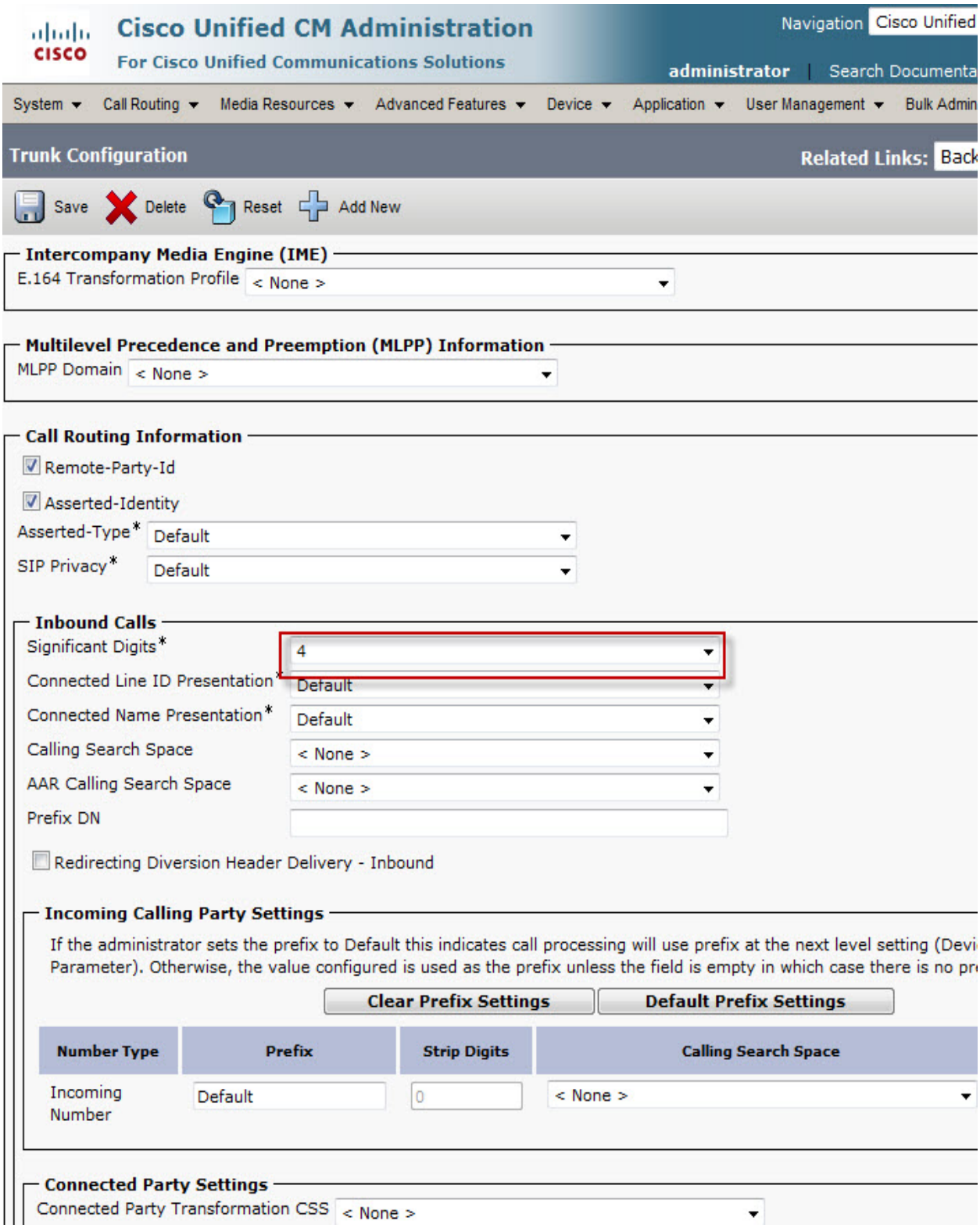


Figure 32 SIP Trunk (Cont.)

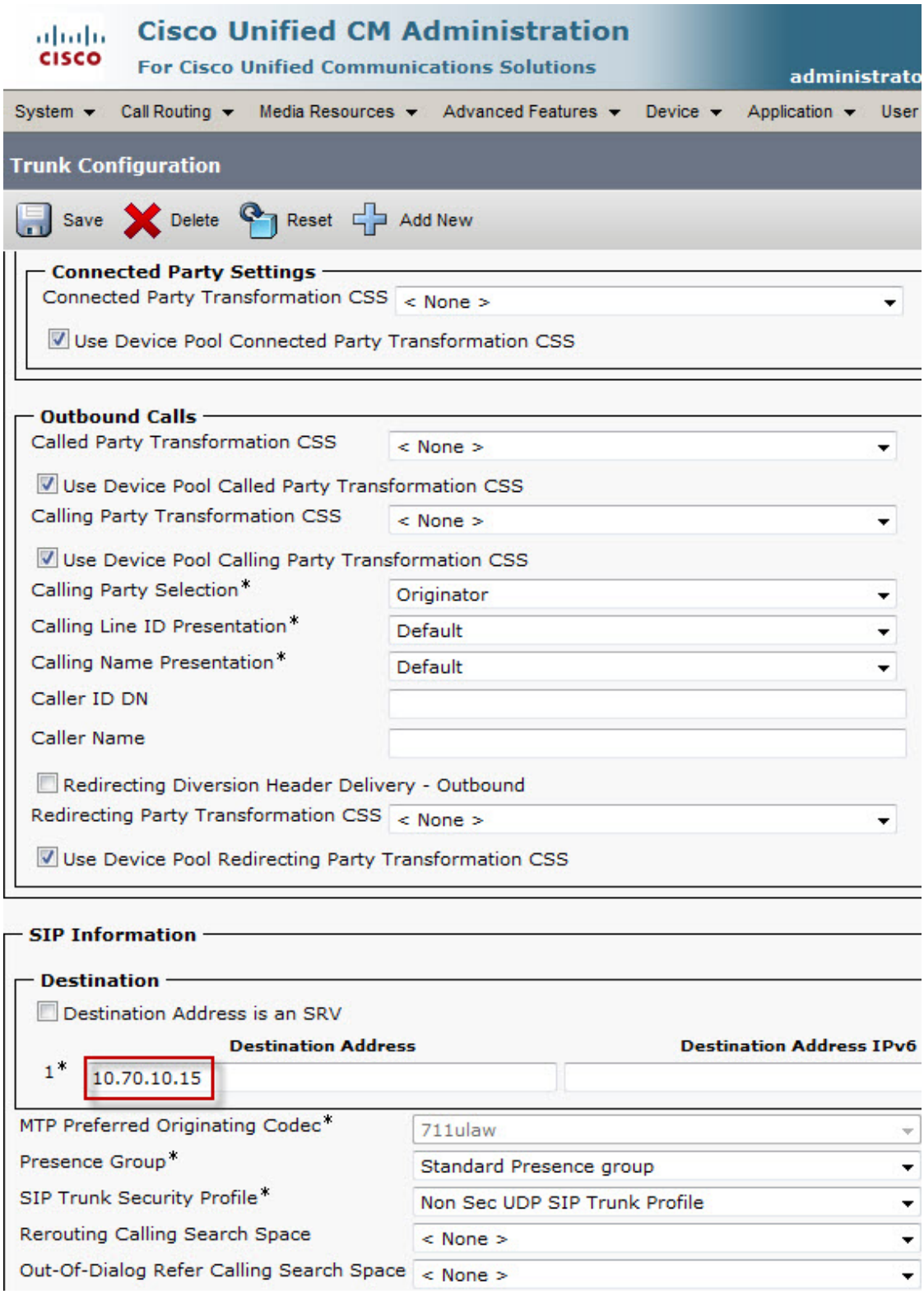


Figure 33 SIP Trunk (Cont.)

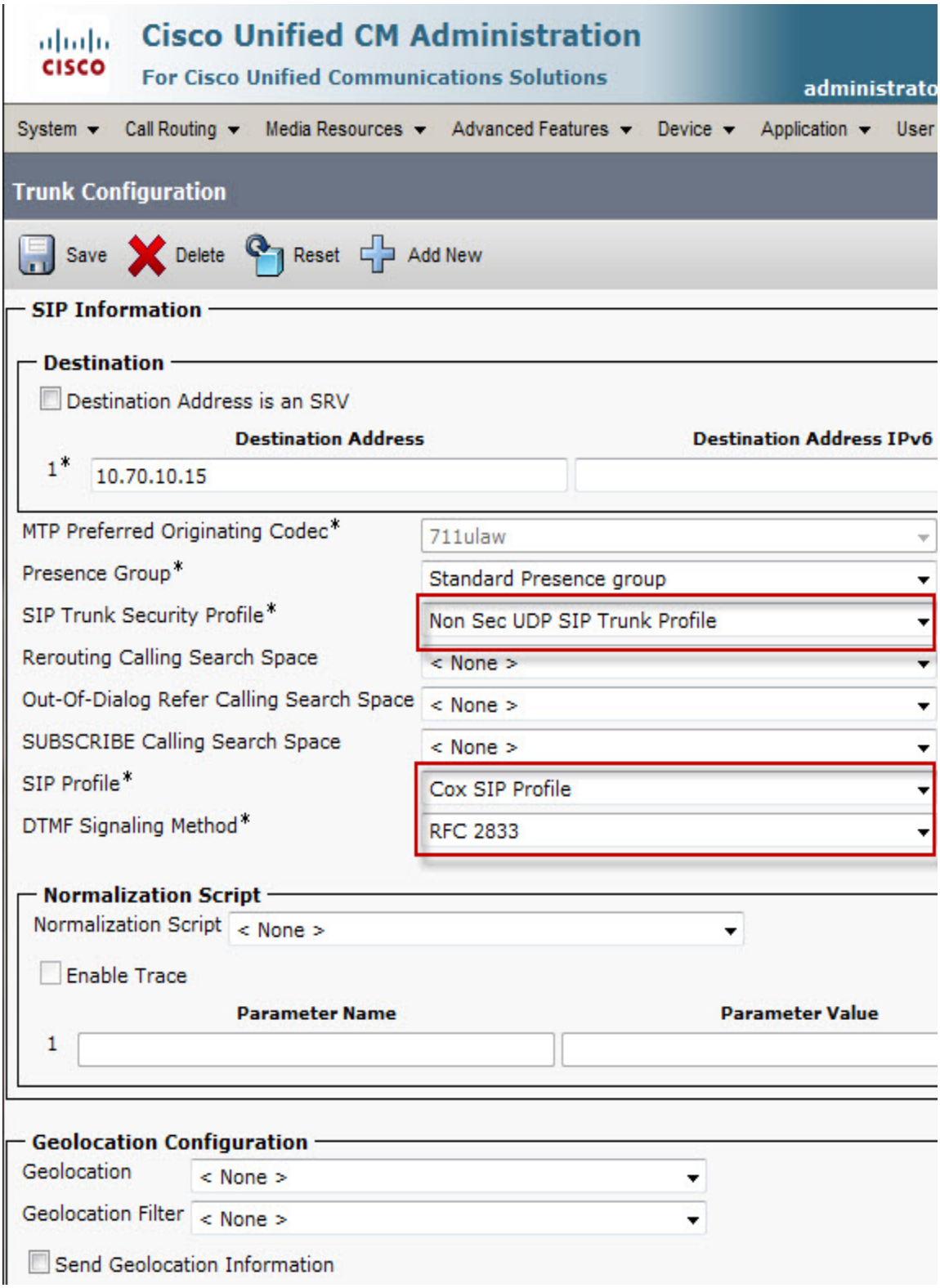


Figure 34 SIP Trunk (Cont.)

6.3.8 SIP Profiles

The Cox SIP Profile is created by modifying the Standard SIP Profile.

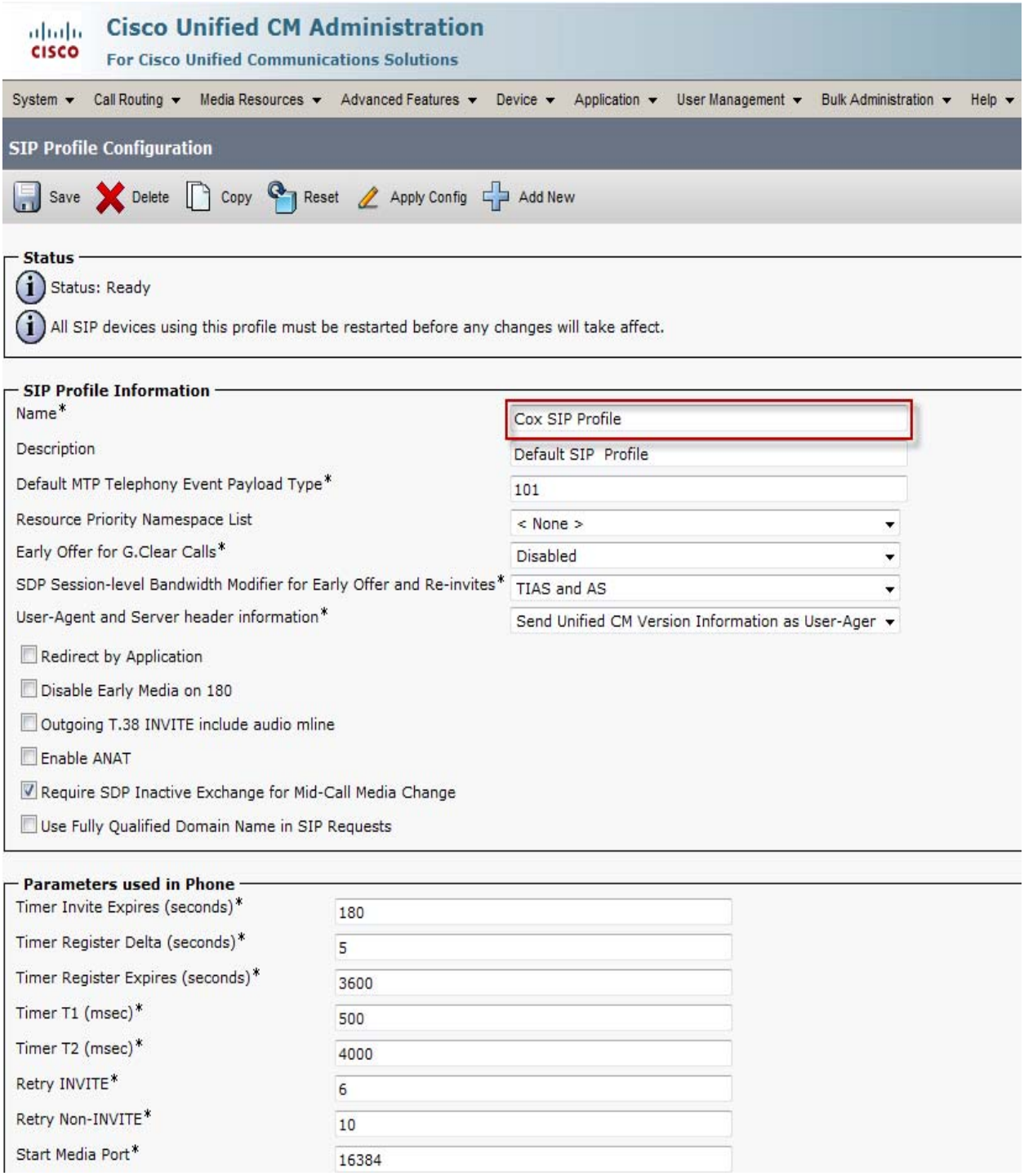


Figure 35 SIP Profile

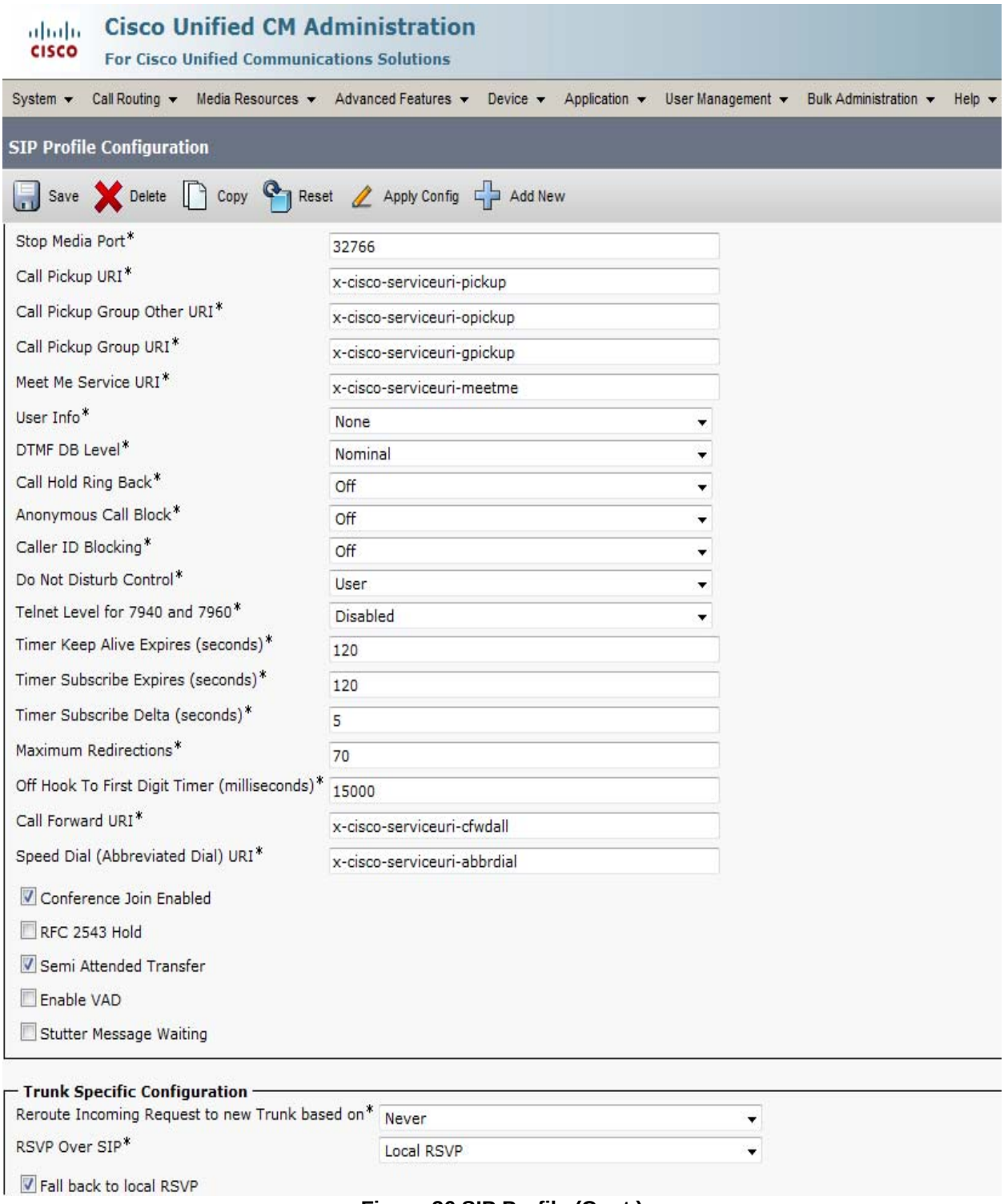


Figure 36 SIP Profile (Cont.)

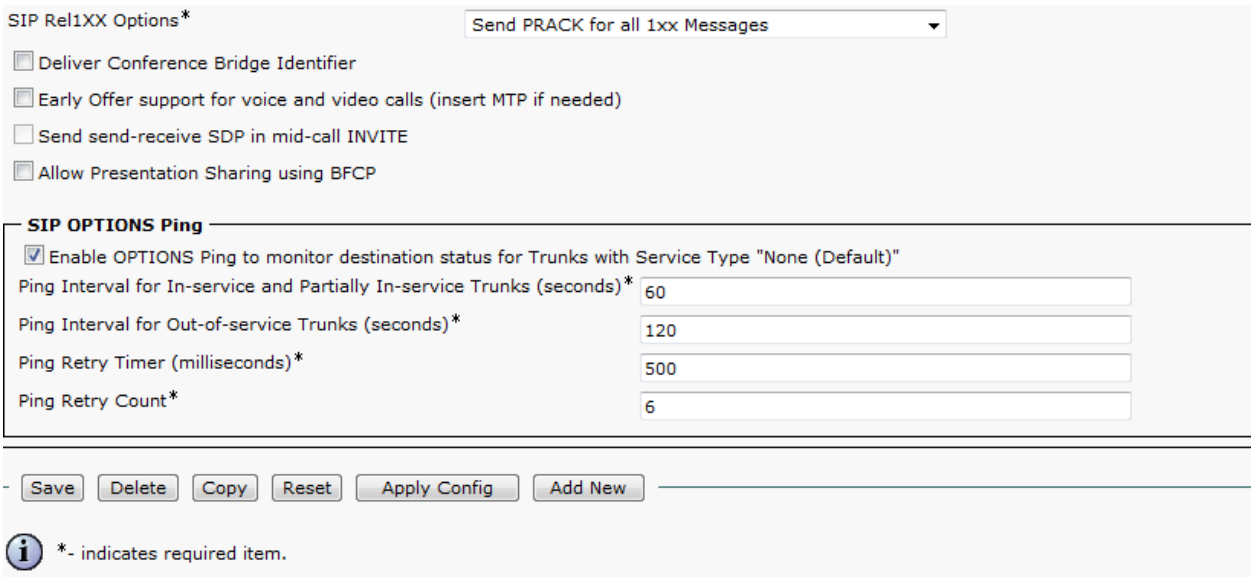


Figure 37 SIP Profile (Cont.)

6.3.9 SIP Trunk Security Profile

1. Navigate to System > Security > SIP Trunk Security Profile
2. Select Add New
3. Assign a SIP Trunk Security Profile name
4. The Incoming Transport Type and Outgoing Transport Type are default settings
5. Select Save

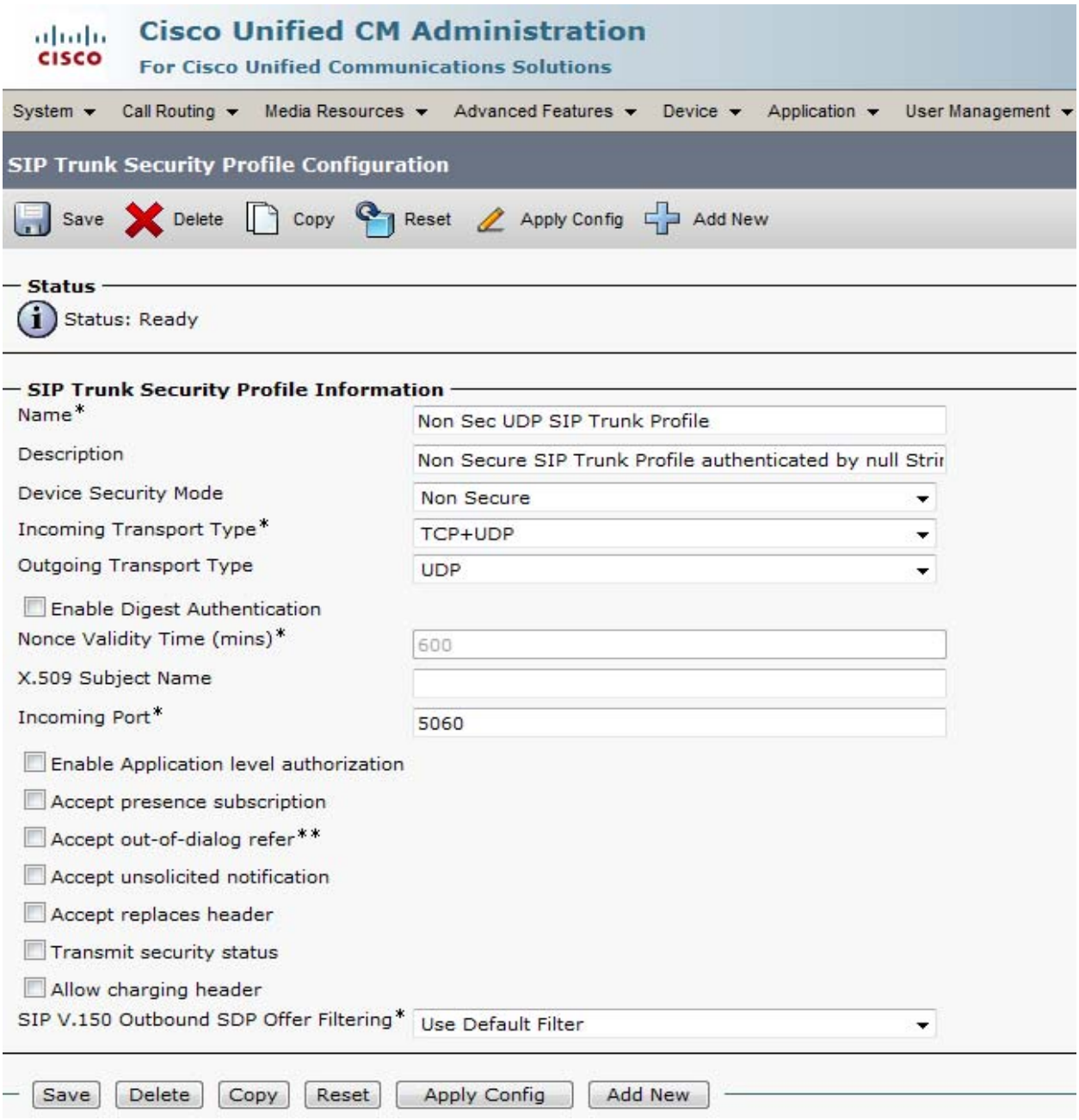


Figure 38 SIP Trunk Security Profile

6.3.10 Route Pattern

The Route Pattern is the essence of the CUCM. It works in conjunction with route filters and route lists to direct calls to specific devices and to include, exclude, or modify specific digit patterns. Figure 34 is the Route Pattern list. The following steps describe adding a new route pattern.

1. Select Add New
2. Define a Route Pattern
3. Select a Gateway/Route List
4. In the Calling Party Transformations section, define the Calling Party Transform Mask. In this case 678238XXXX is used. This setting manipulates the appearance of the calling party's number for outgoing calls (CLID).
5. In the Called Party transformations section, define the Discard Digits. PreDot is used for this example. In this case, the leading 9 will be stripped from the dialed digit string.
6. Select Save

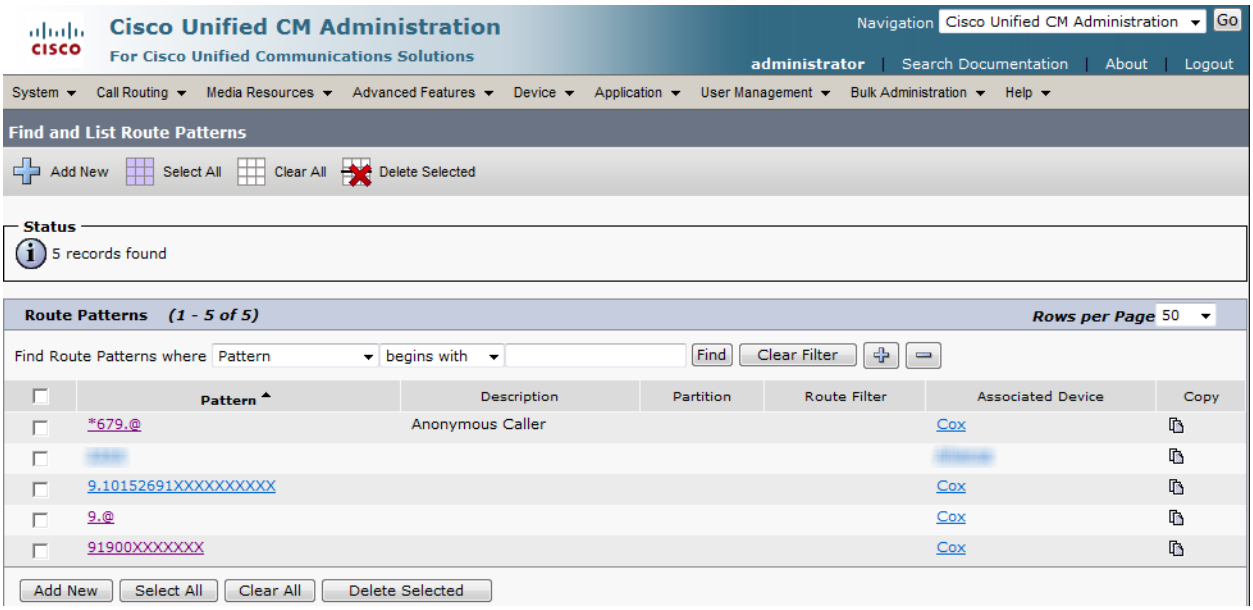


Figure 39 Route Pattern

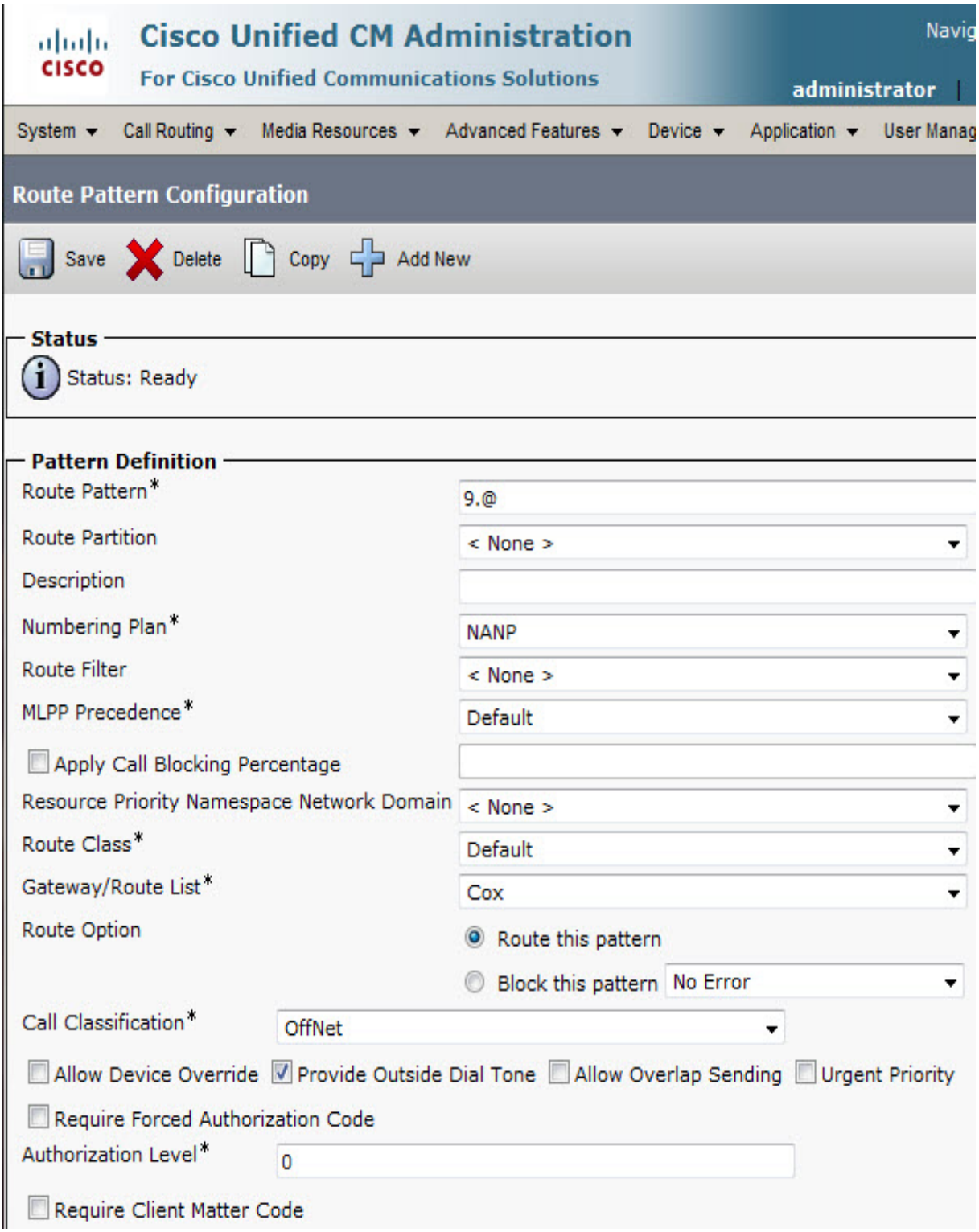


Figure 40 Route Pattern (Cont.)

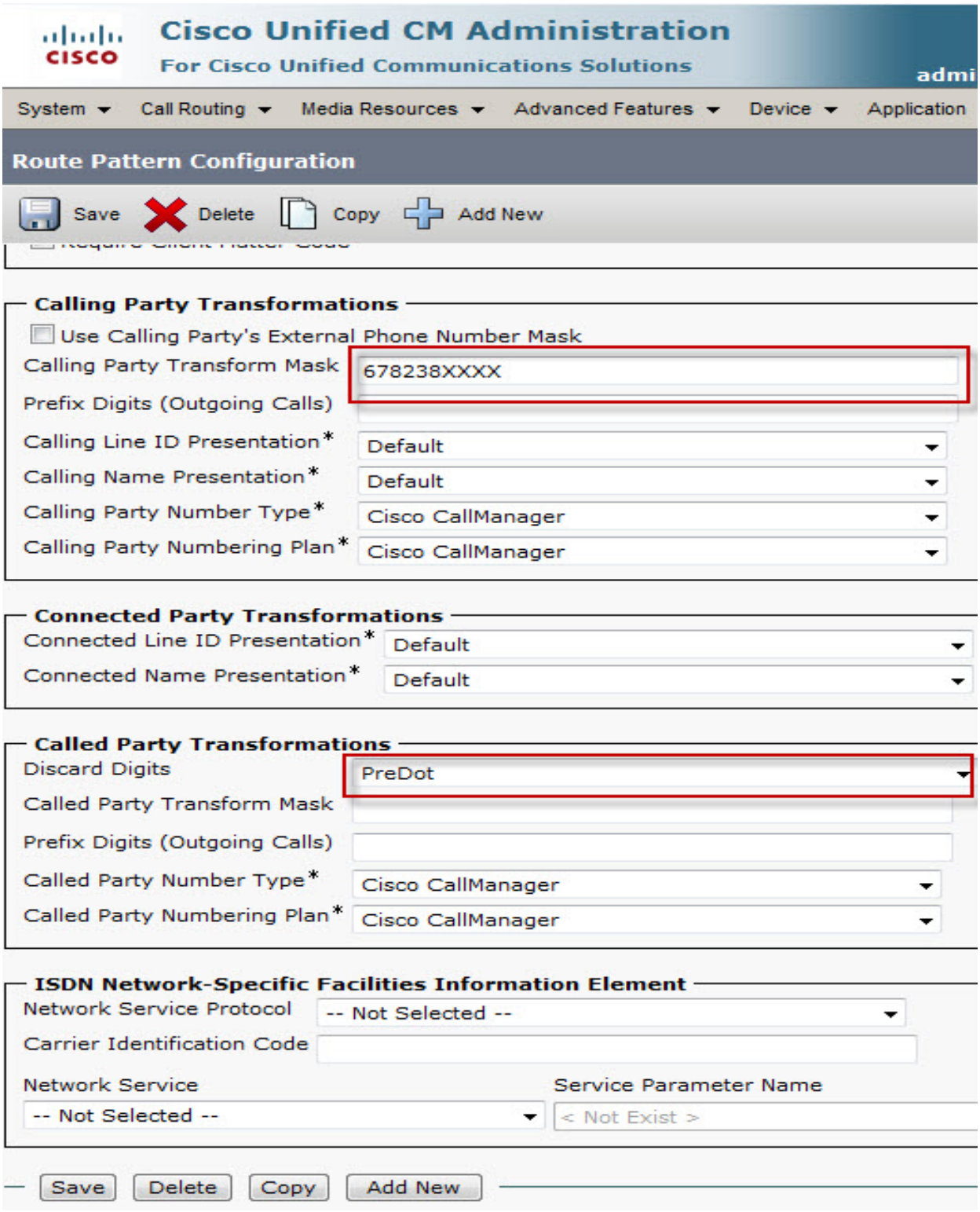


Figure 41 Route Pattern (Cont.)

6.3.11 Line Groups

Line groups are referenced by Hunt Lists

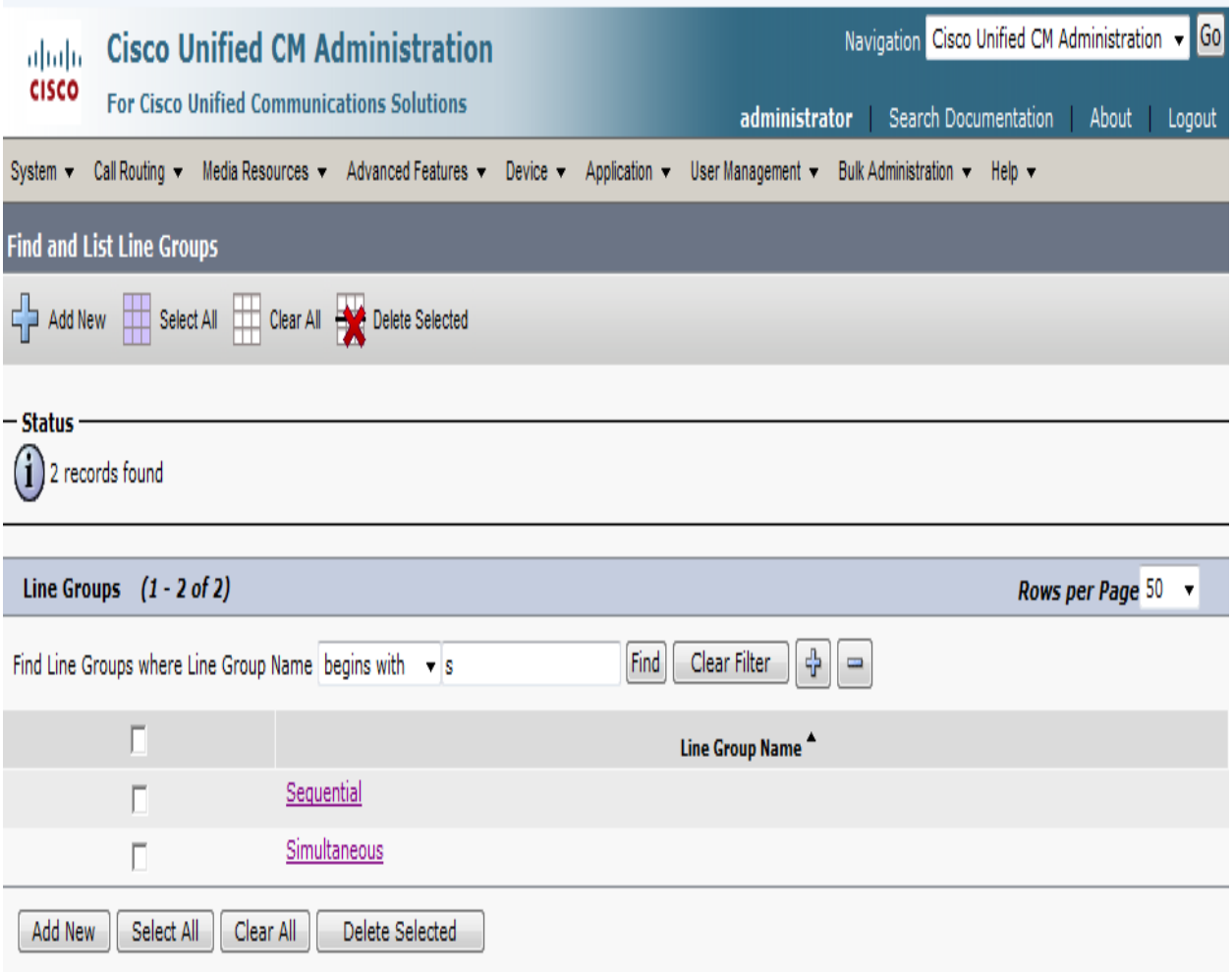


Figure 42 Line Groups

Configure the Distribution Algorithm to determine the order in which Line Group members are selected. For Sequential, select Top Down distribution

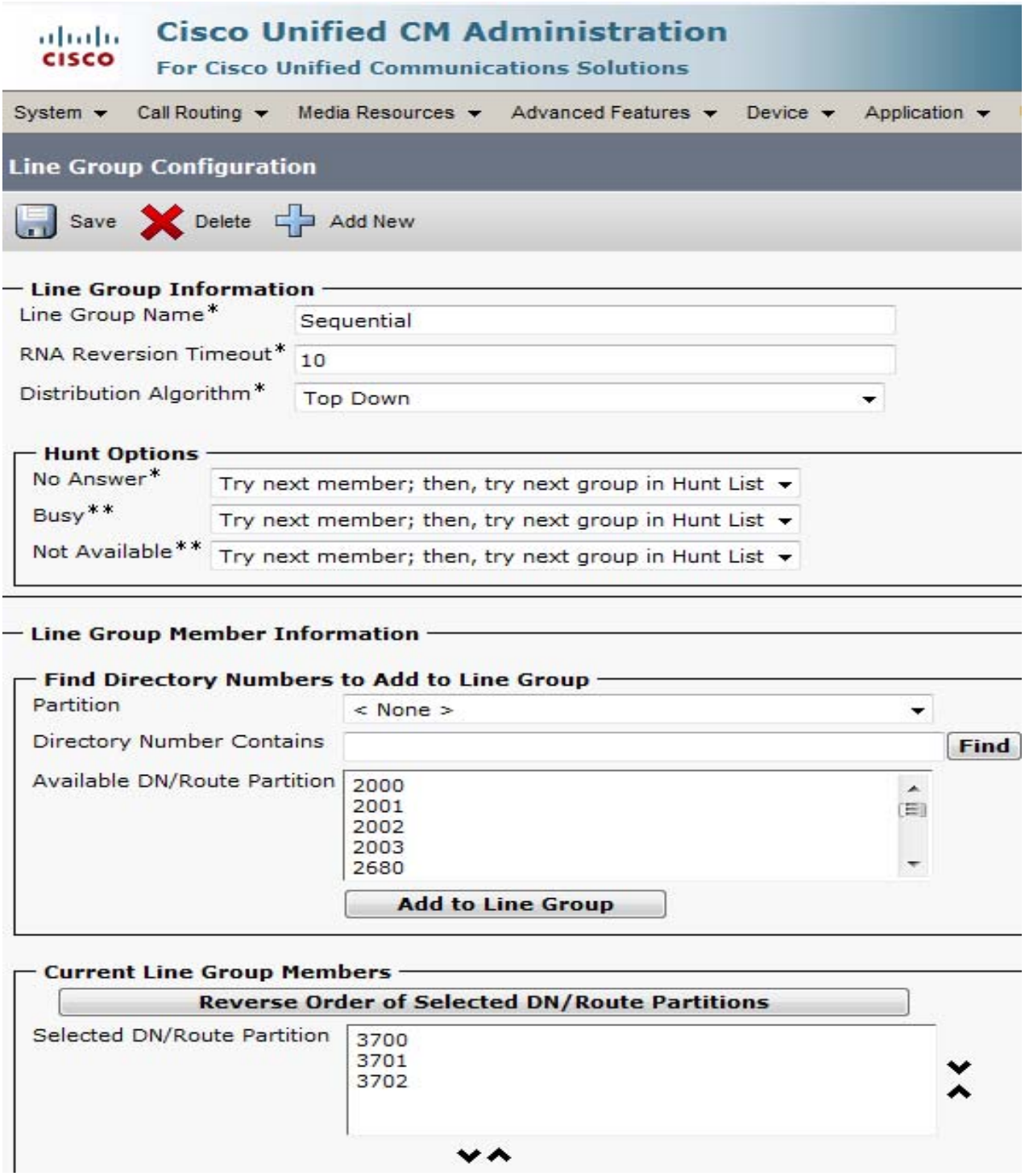


Figure 43 Sequential Line Group

For Simultaneous distribution, select Broadcast as the Distribution Algorithm

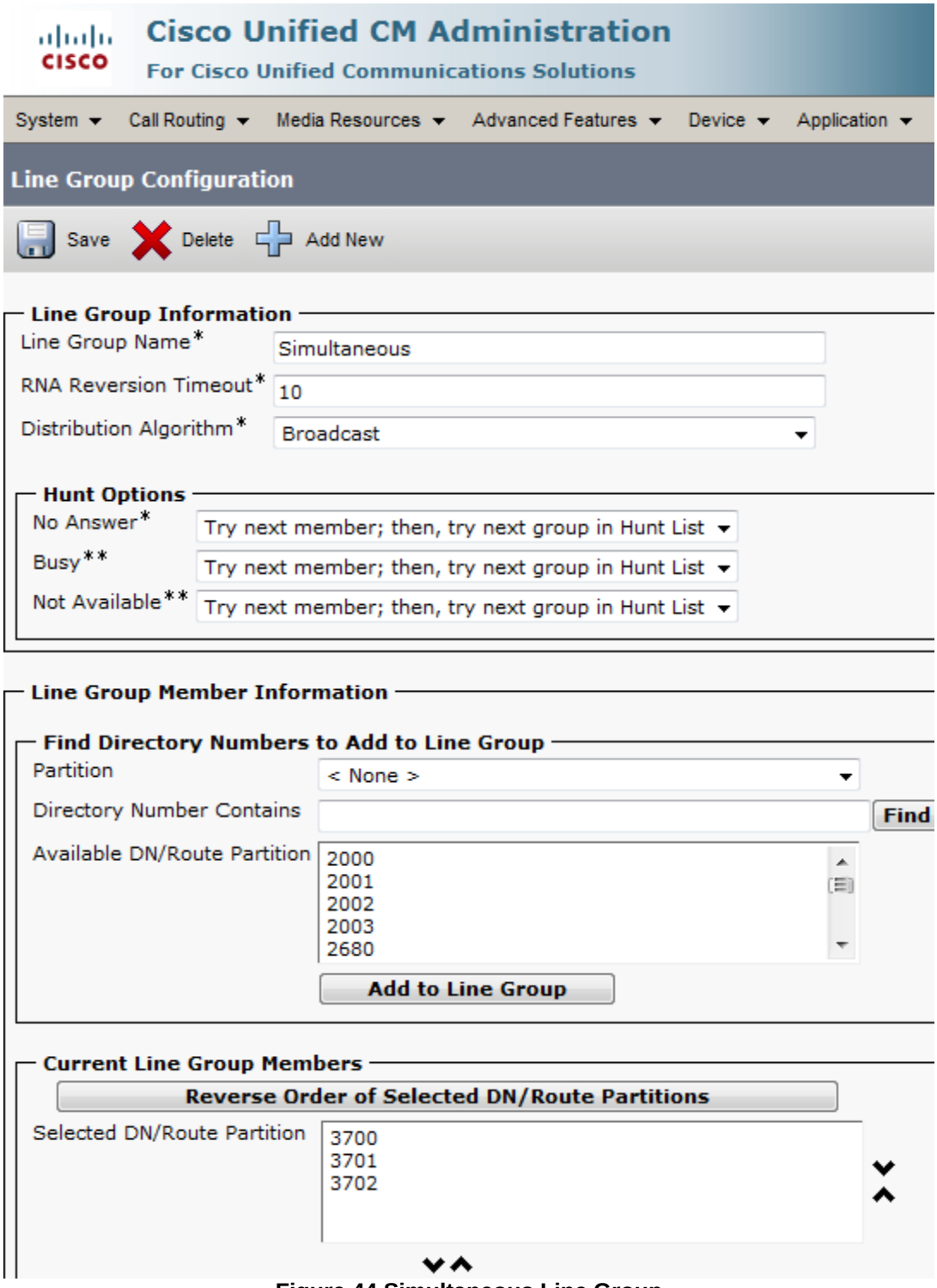


Figure 44 Simultaneous Line Group

6.3.12 Hunt Lists

Hunt Lists are populated with Line Groups. Each Line Group contains Line Group Members.

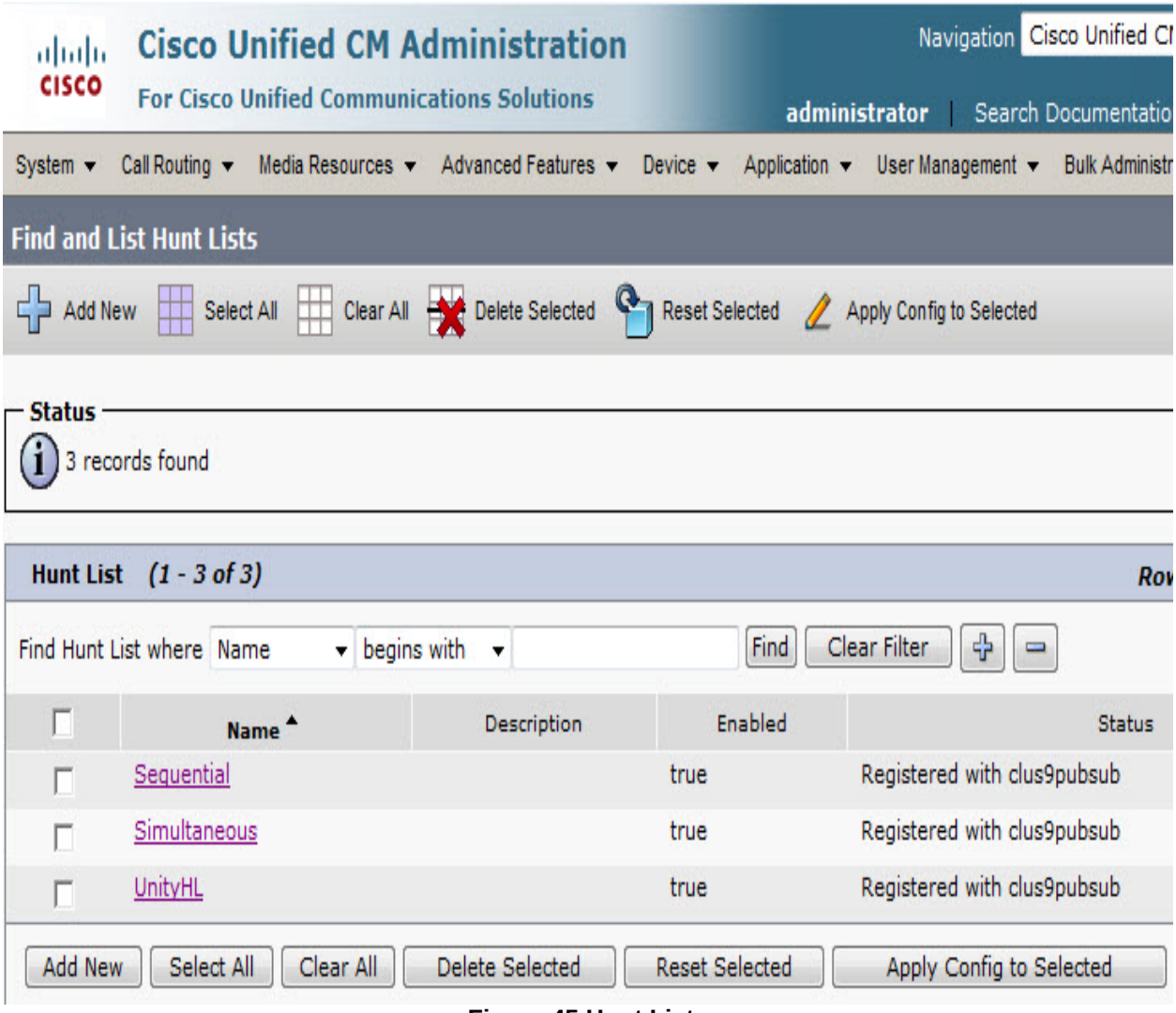


Figure 45 Hunt Lists

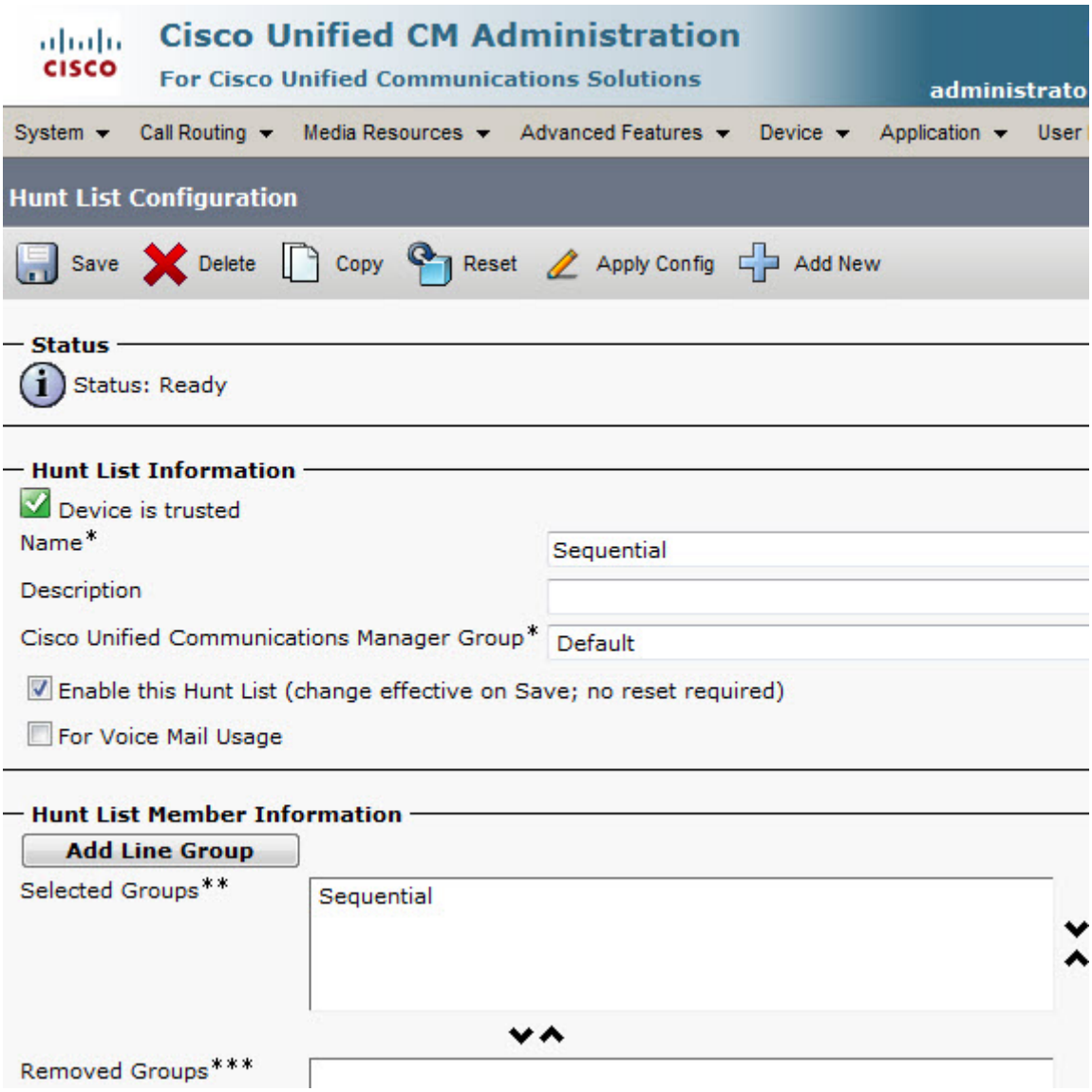


Figure 46 Hunt Lists (Cont.)

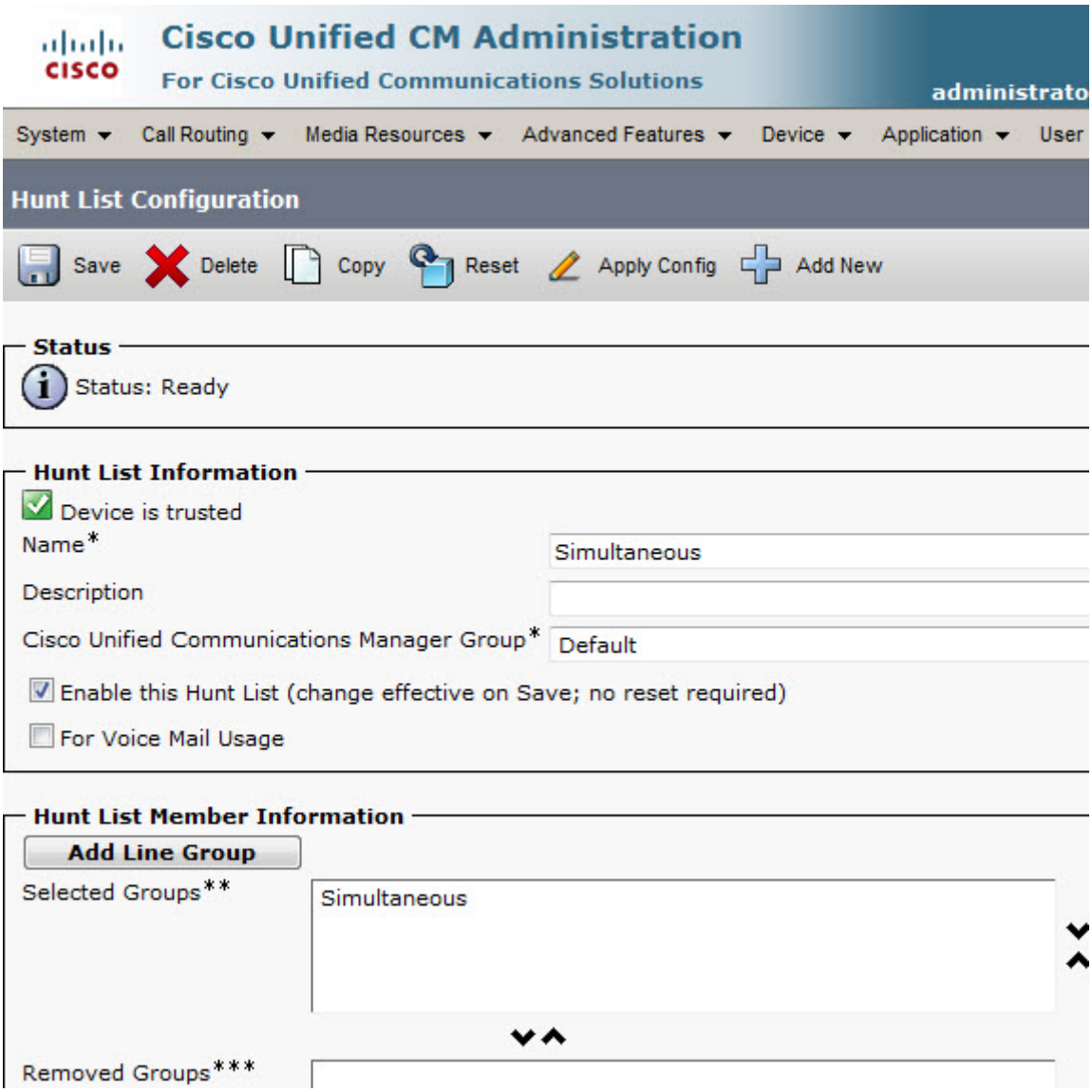


Figure 47 Hunt Lists (Cont.)

6.3.13 Hunt Pilot Numbers

To configure Hunt Pilot Numbers, navigate to Call Routing > Route/Hunt > Hunt Pilot

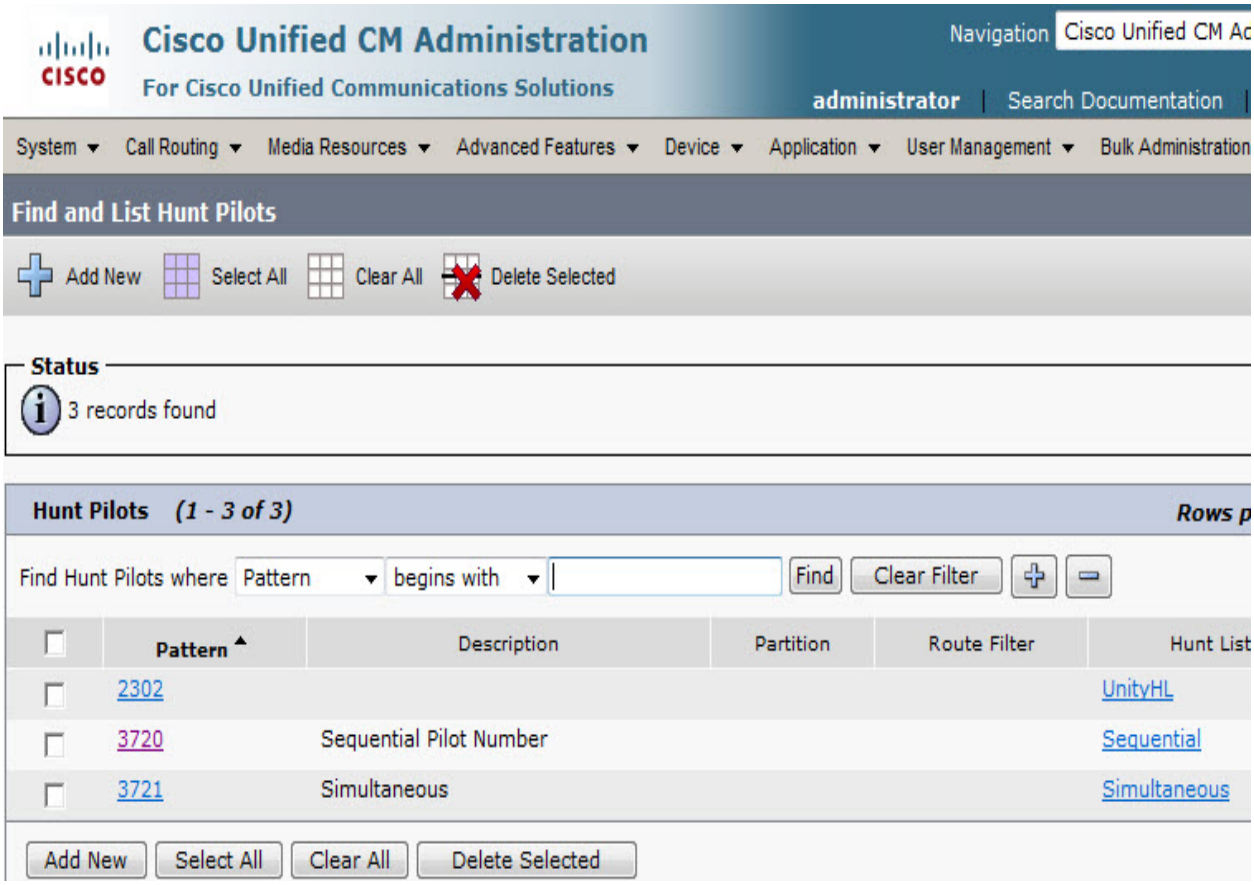


Figure 48 Hunt Pilot Numbers

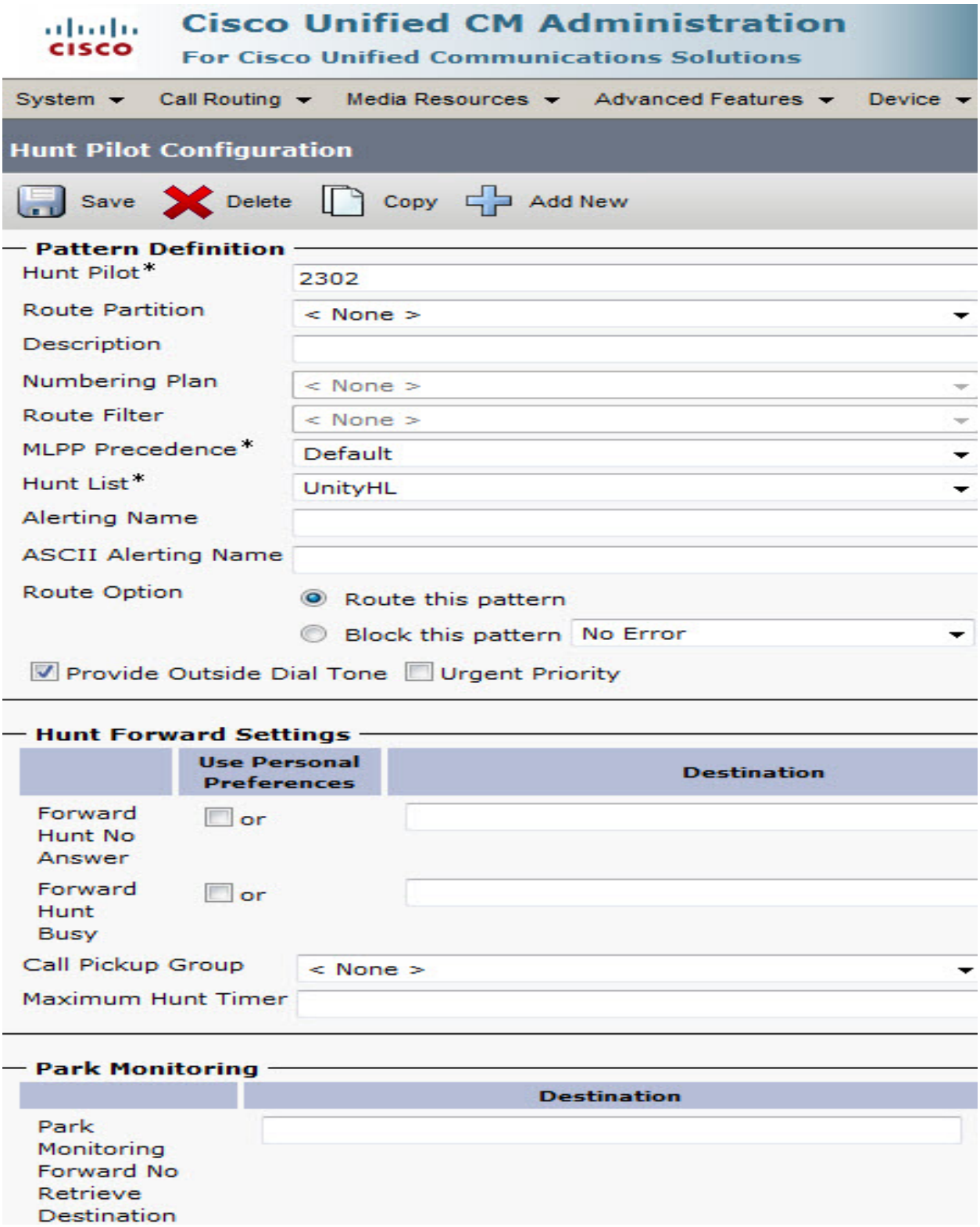


Figure 49 Hunt Pilot Number Unity

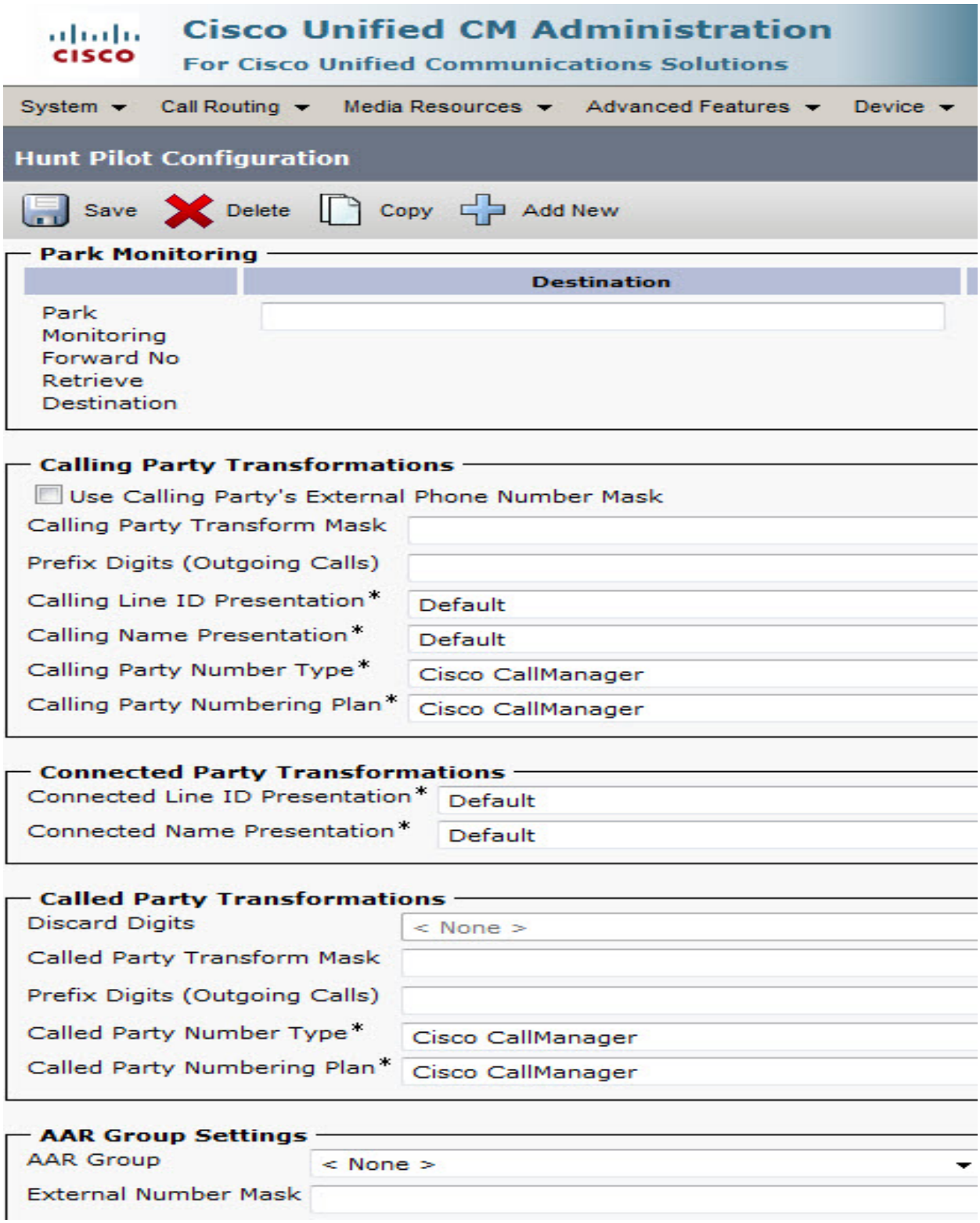


Figure 50 Hunt Pilot Number Unity (Cont.)

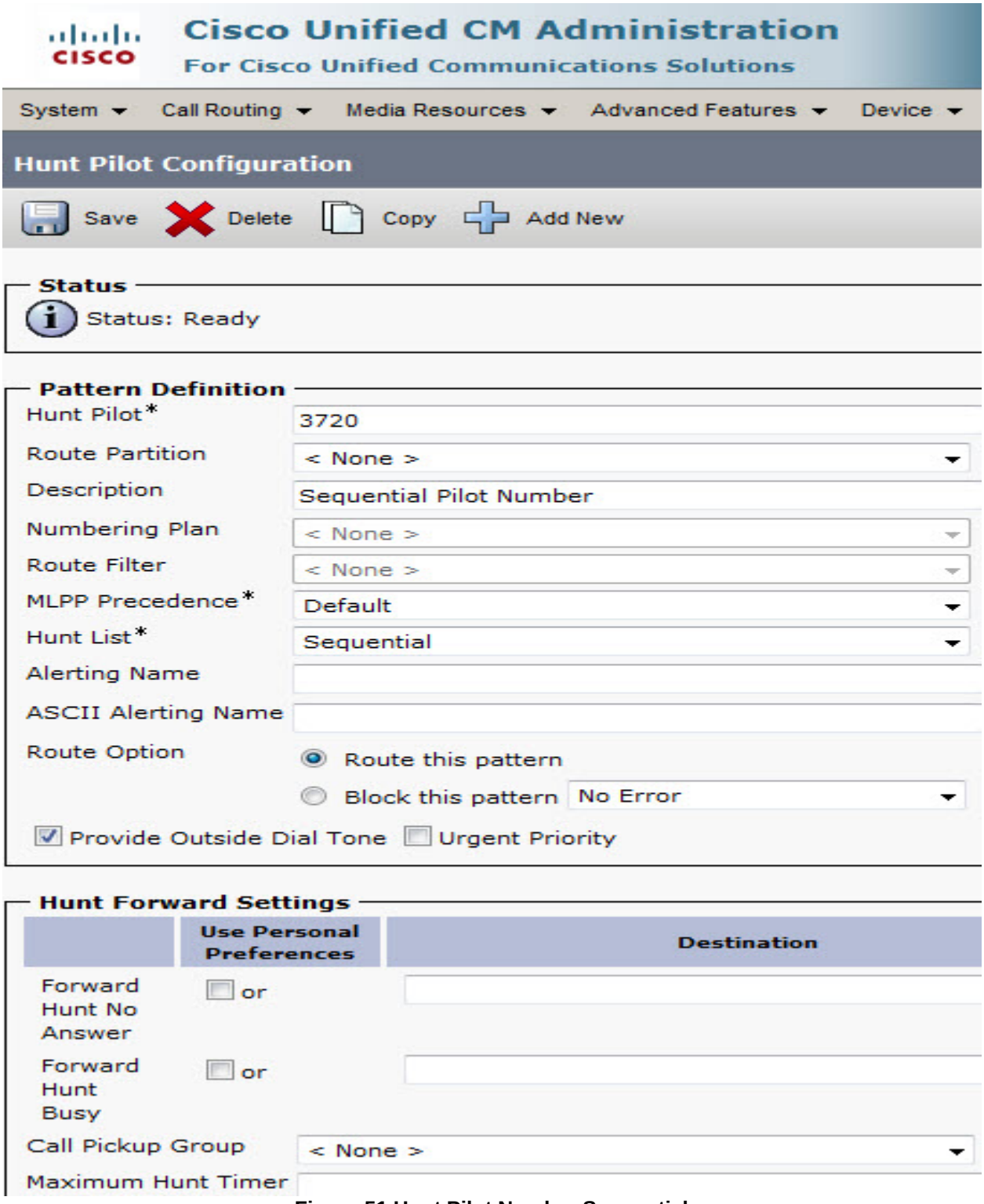


Figure 51 Hunt Pilot Number Sequential

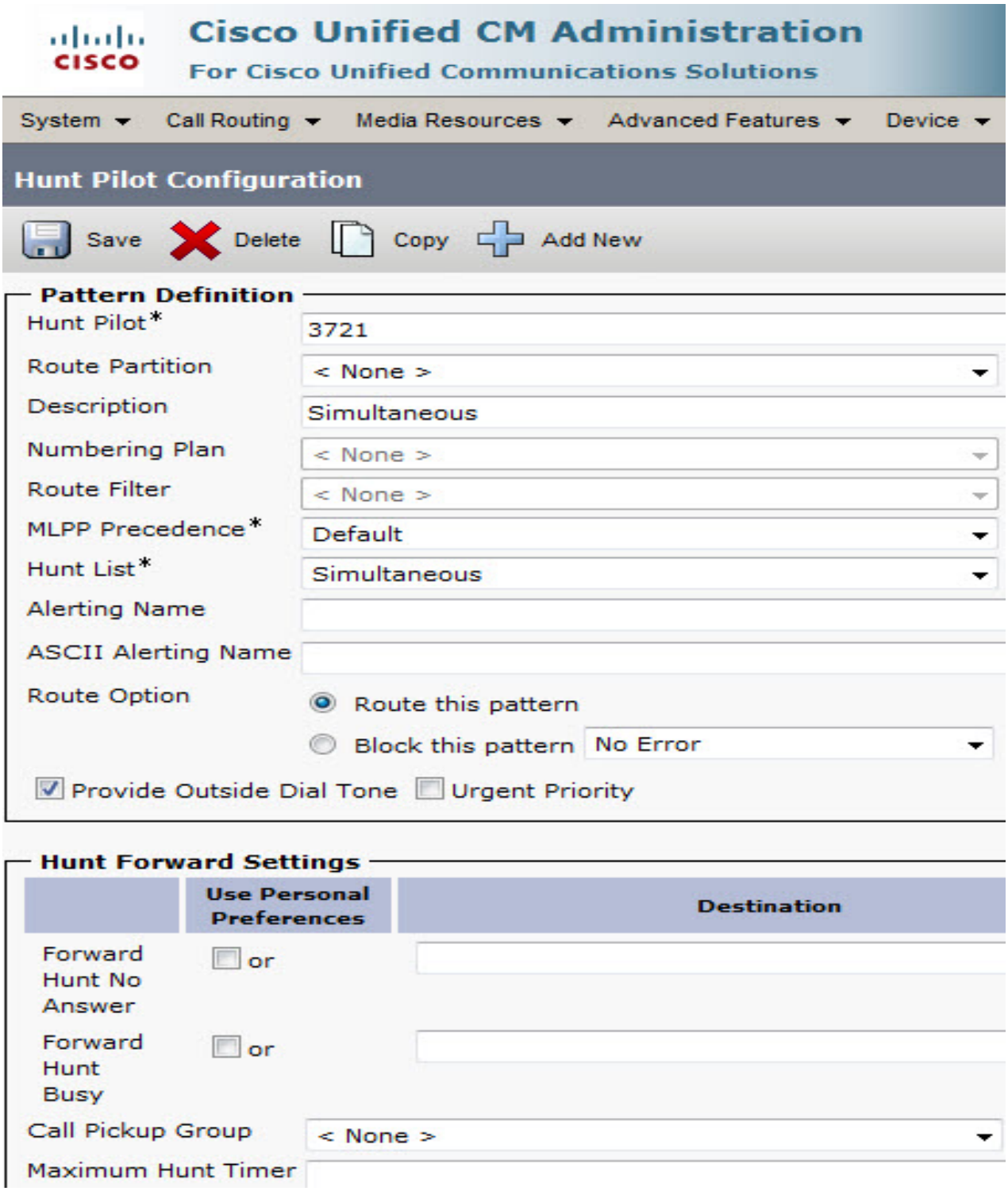


Figure 52 Hunt Pilot Number Simultaneous

6.3.14 Client Matter Codes

Client Matter Codes are configured to track phone calls for billing purposes. To configure Client Matter Codes, navigate to Call Routing > Client Matter Codes > Add New. After a code is created, it must be activated at the route pattern level.

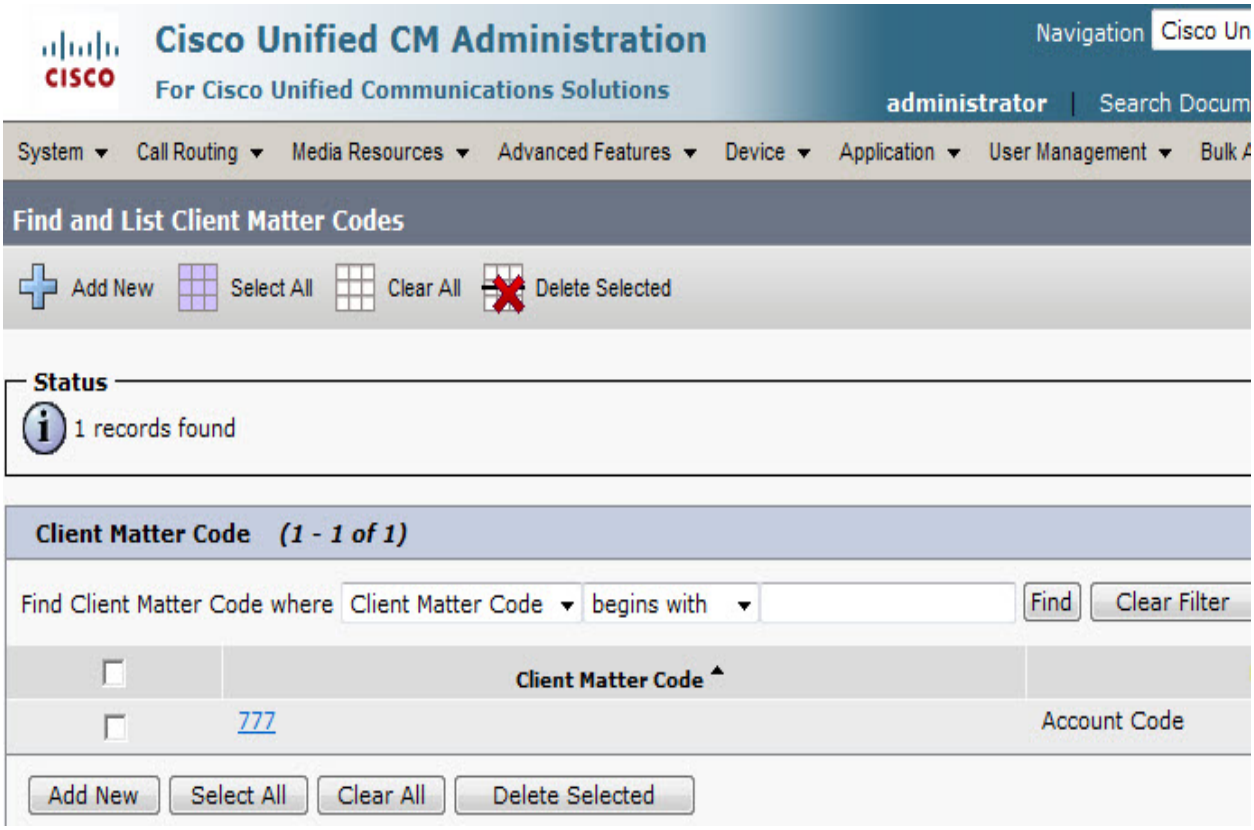


Figure 53 Client Matter Codes

Figure 54 Client Matter Codes (Cont.)

6.3.15 Forced Authorization Codes

Authorization Codes are configured for security reasons. The caller must enter the code in order to complete the call. Navigate to Call Routing > Forced Authorization Codes > Add New. After a code is created, it must be activated at the route pattern level.

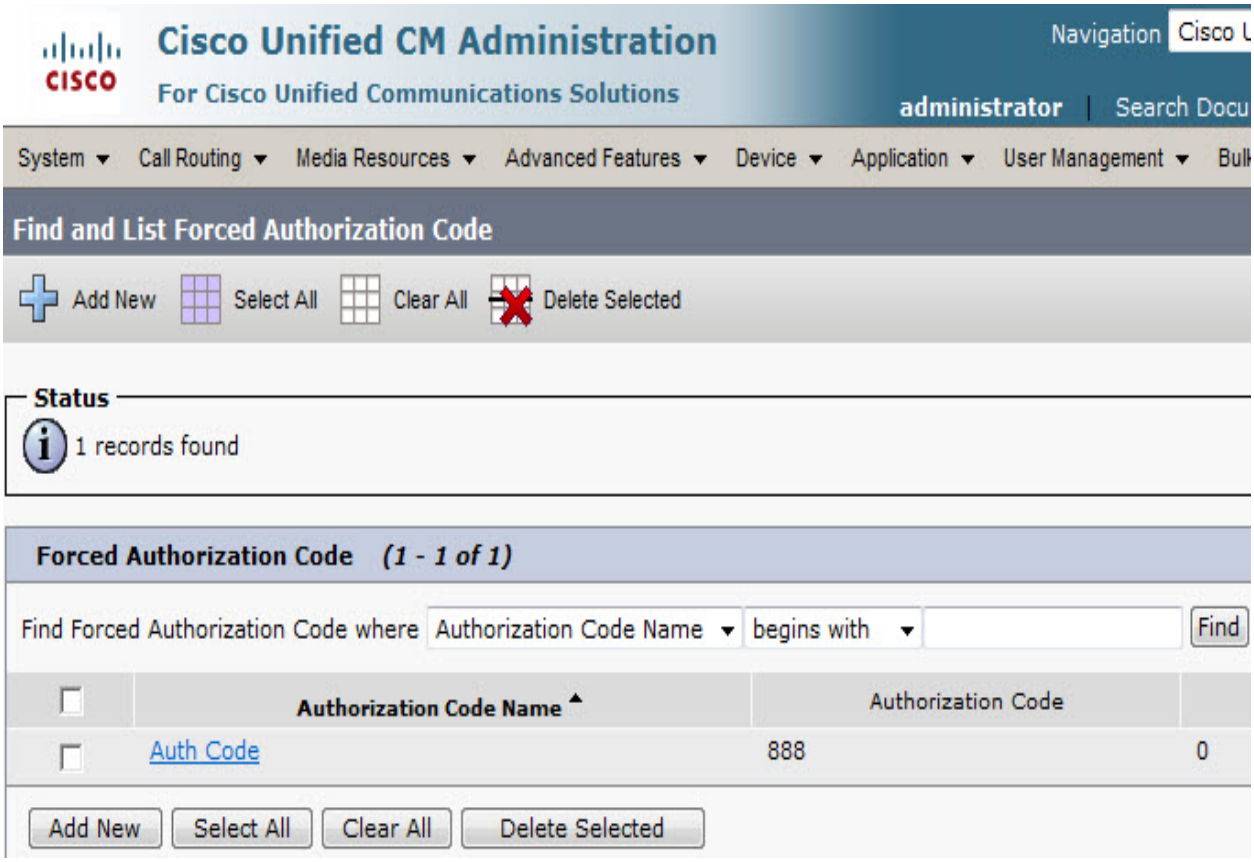


Figure 55 Forced Authorization Codes

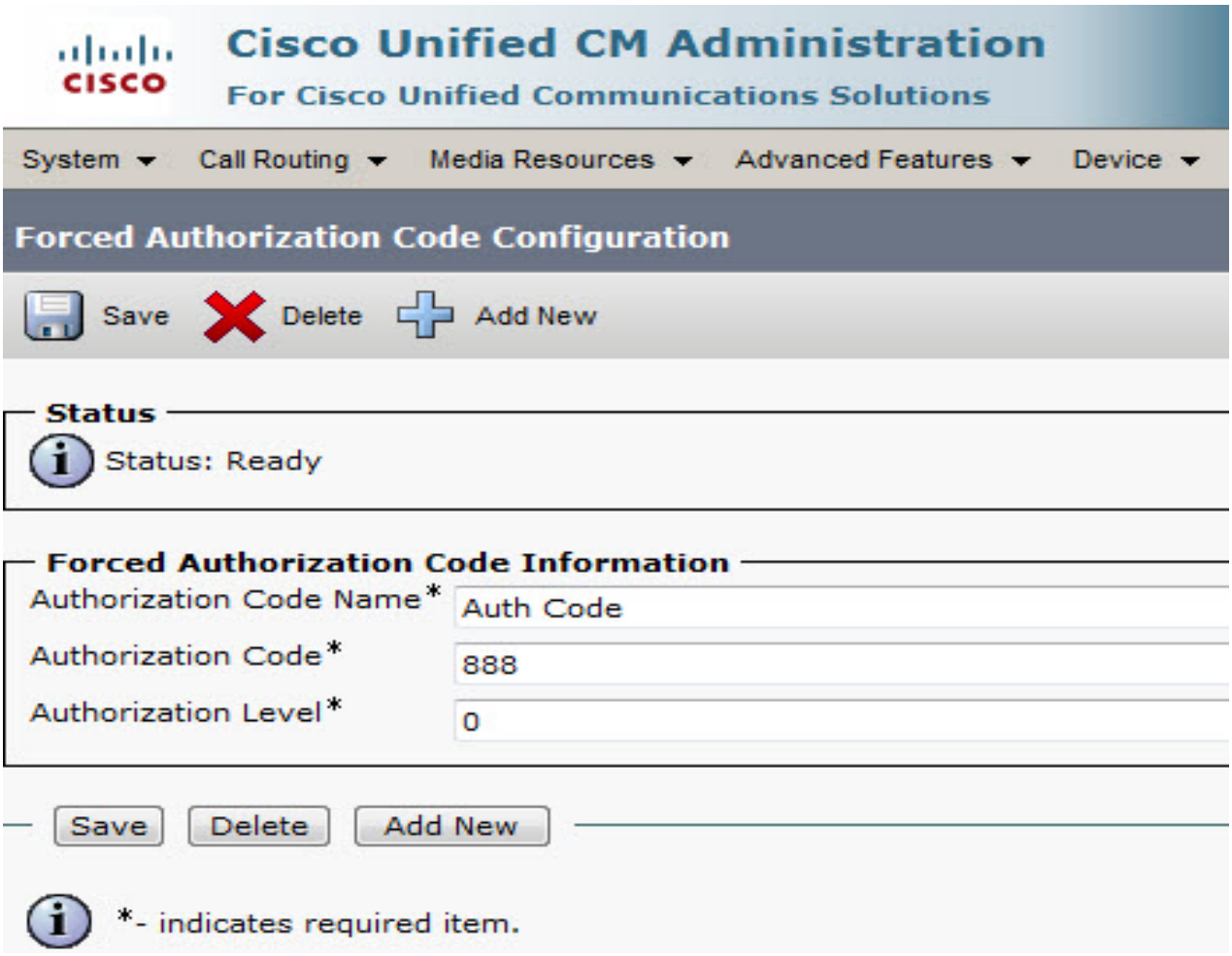


Figure 56 Forced Authorization Codes (Cont.)

6.3.16 Call Park Numbers

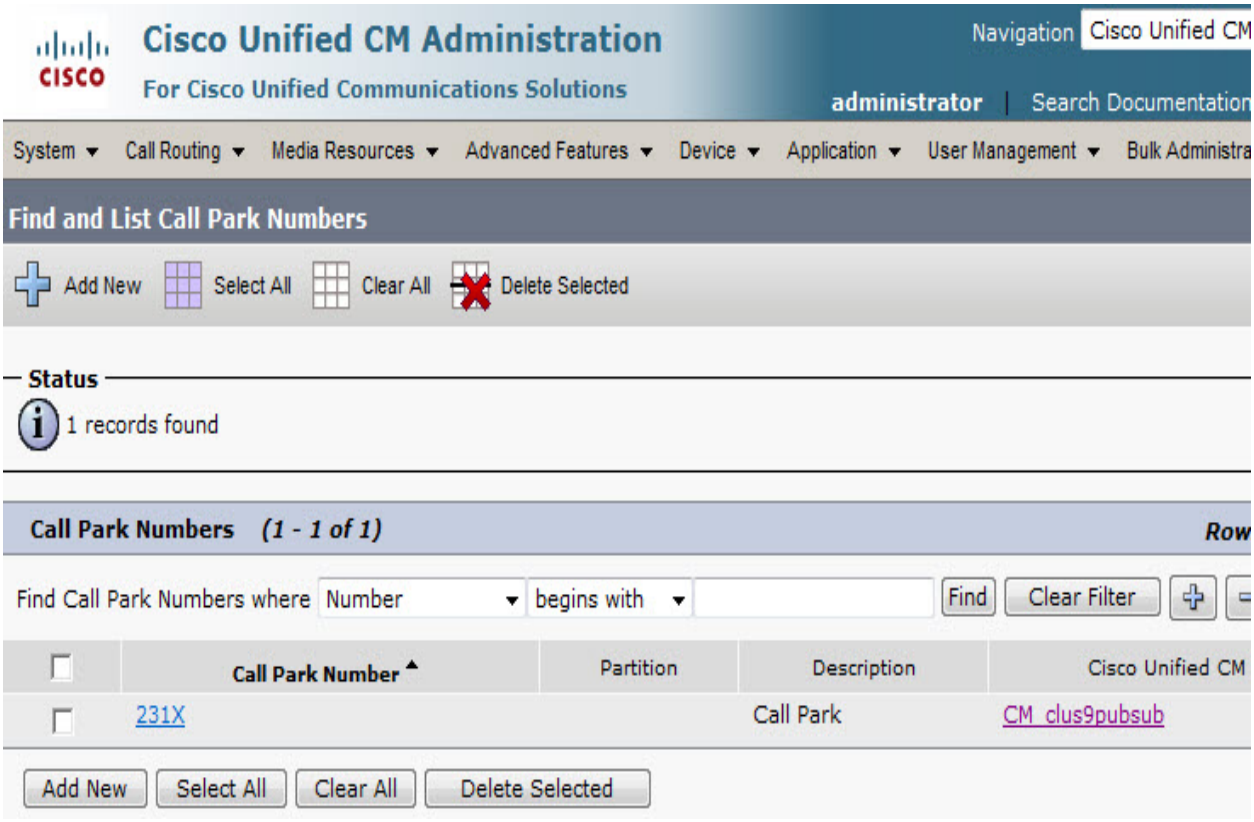


Figure 57 Call Park Numbers

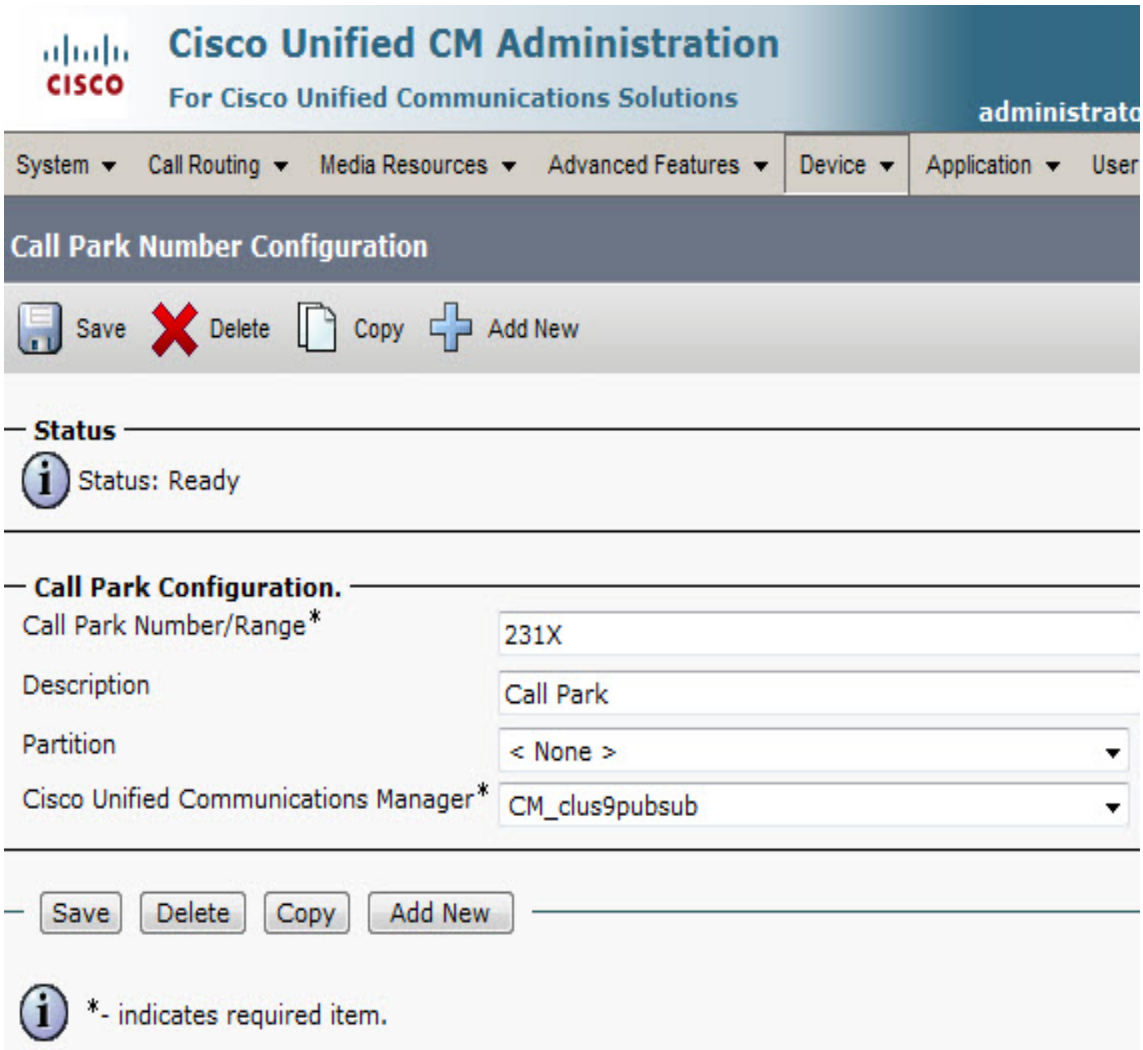


Figure 58 Call Park Numbers (Cont.)

6.3.17 Call Pickup Groups

A phone line can be assigned to only one pickup group. A user can pick up a call by pressing the Pickup softkey if they are in the same pickup group as the ringing phone. If not, the user must press the Group Pickup softkey and enter the pickup group number

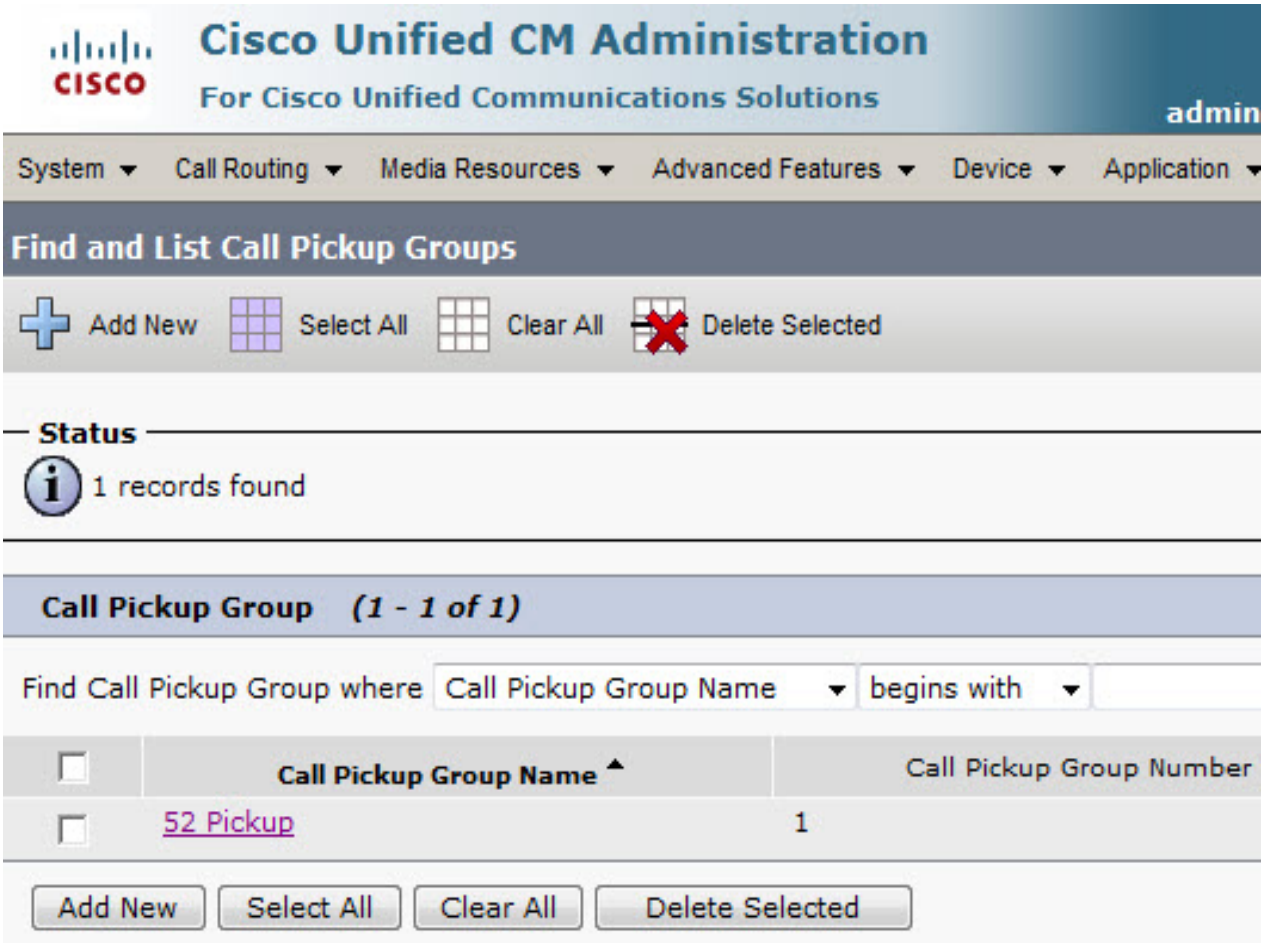


Figure 59 Call Pickup Groups

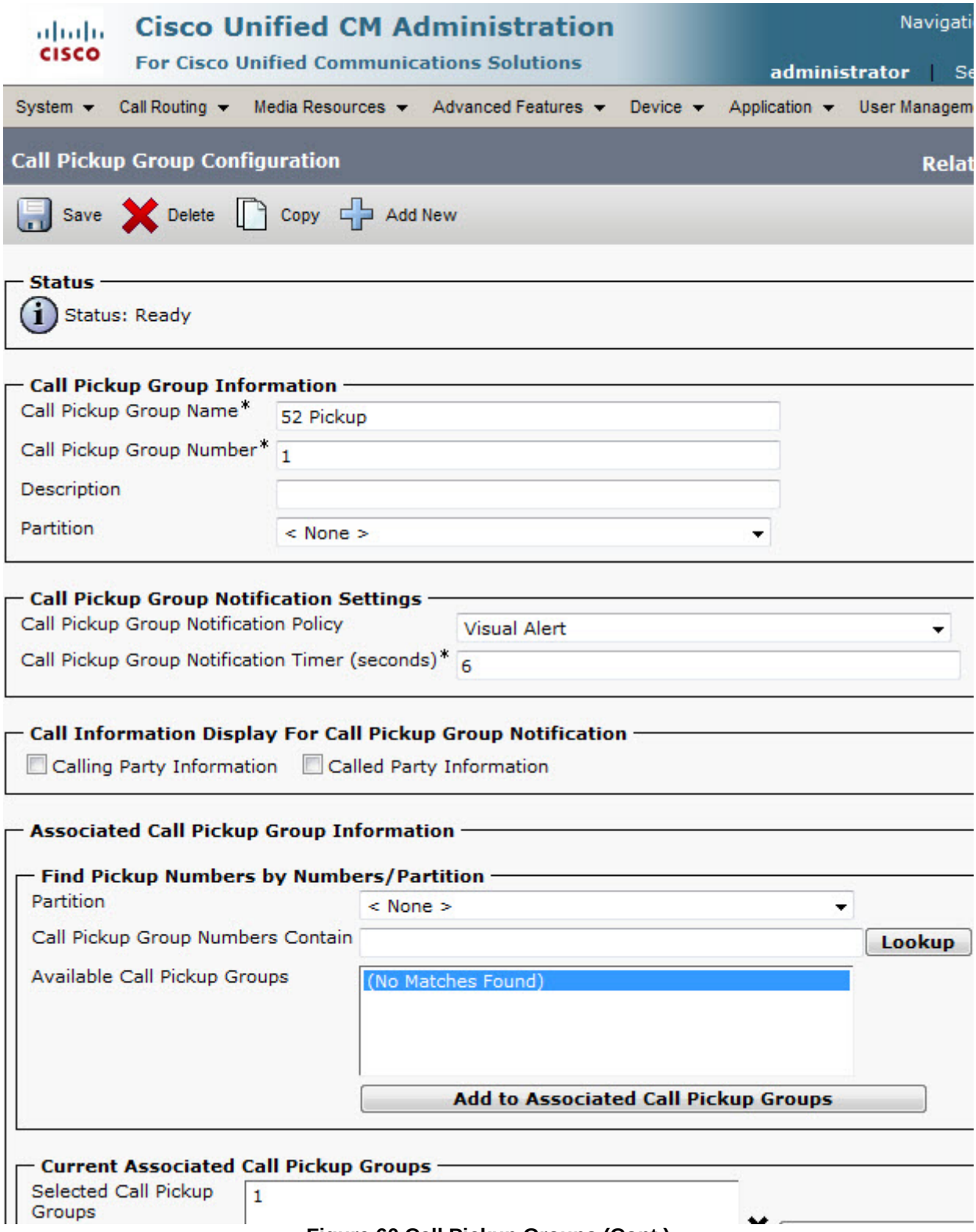


Figure 60 Call Pickup Groups (Cont.)

6.3.18 Route Plan Report

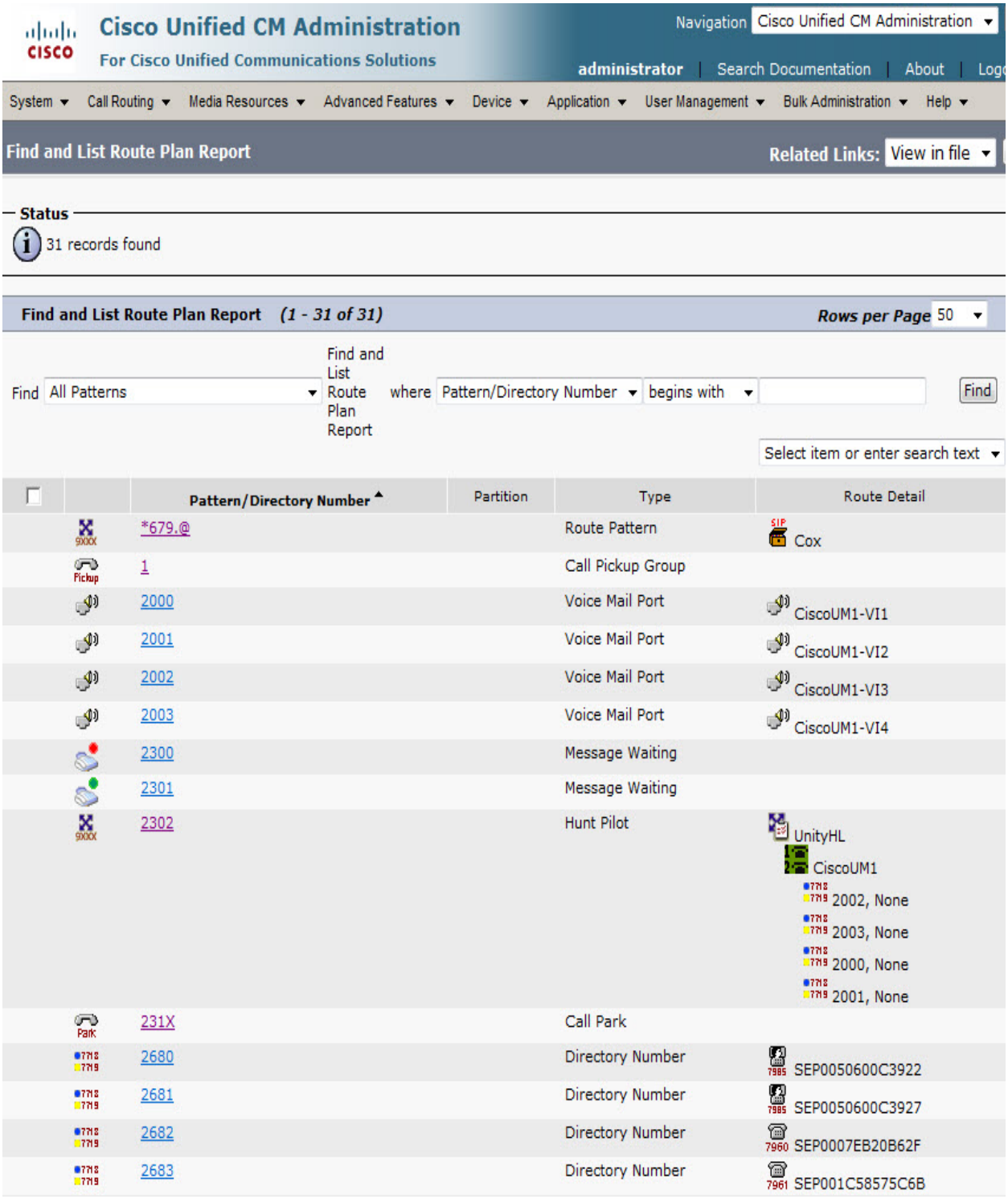


Figure 61 Route Plan Report

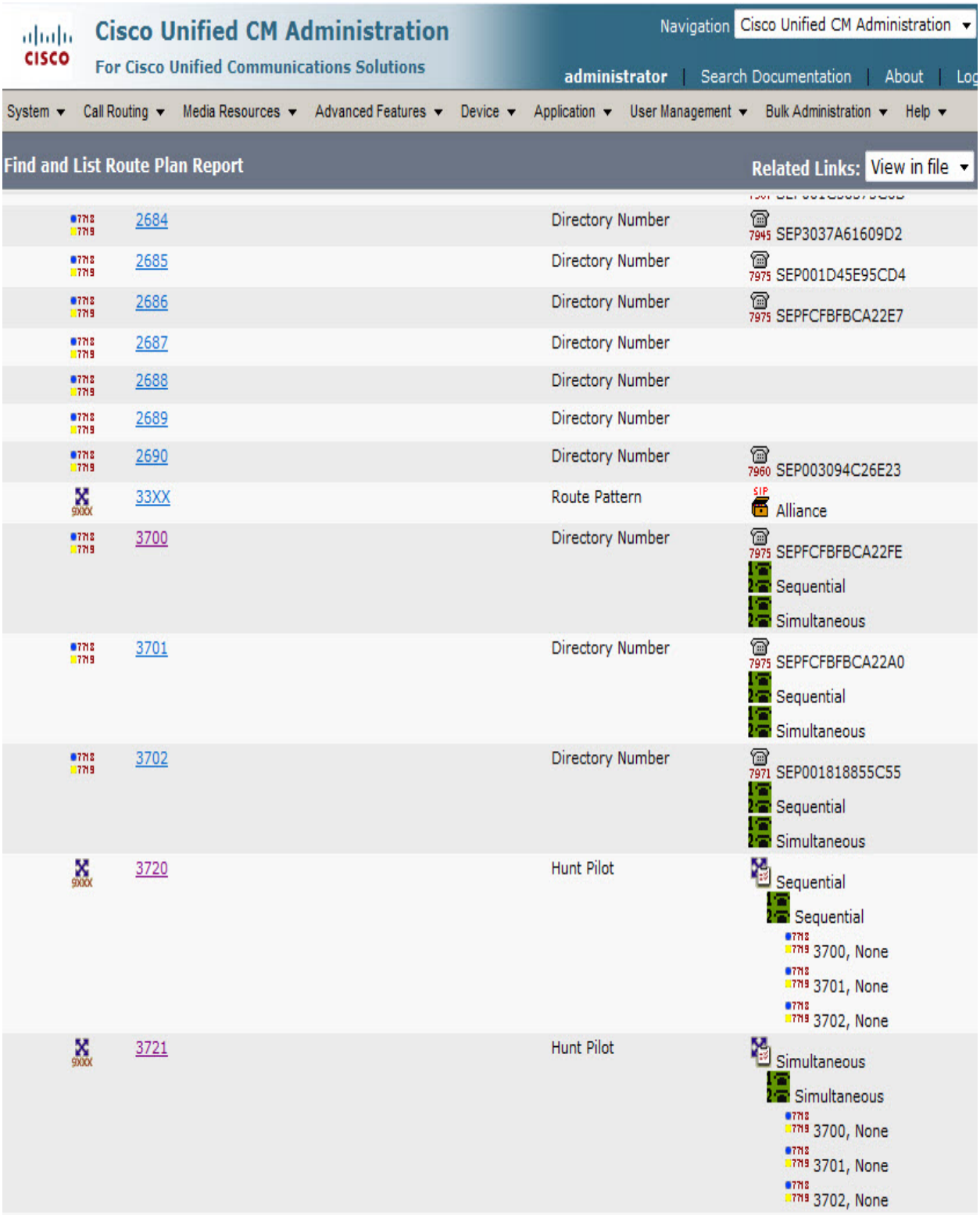


Figure 62 Route Plan Report (Cont.)

9XXX	9.10152691XXXXXXXXXX	Route Pattern	SIP Cox
9XXX	9.@	Route Pattern	SIP Cox
9XXX	91900XXXXXXX	Route Pattern	SIP Cox
DOMAIN	9678xxxxxxx	Domain Routing	

Figure 63 Route Plan Report (Cont.)

6.3.19 Unity Voice Mail Ports

Navigate to Advanced Features > Voice Mail > Cisco Voice Mail Port

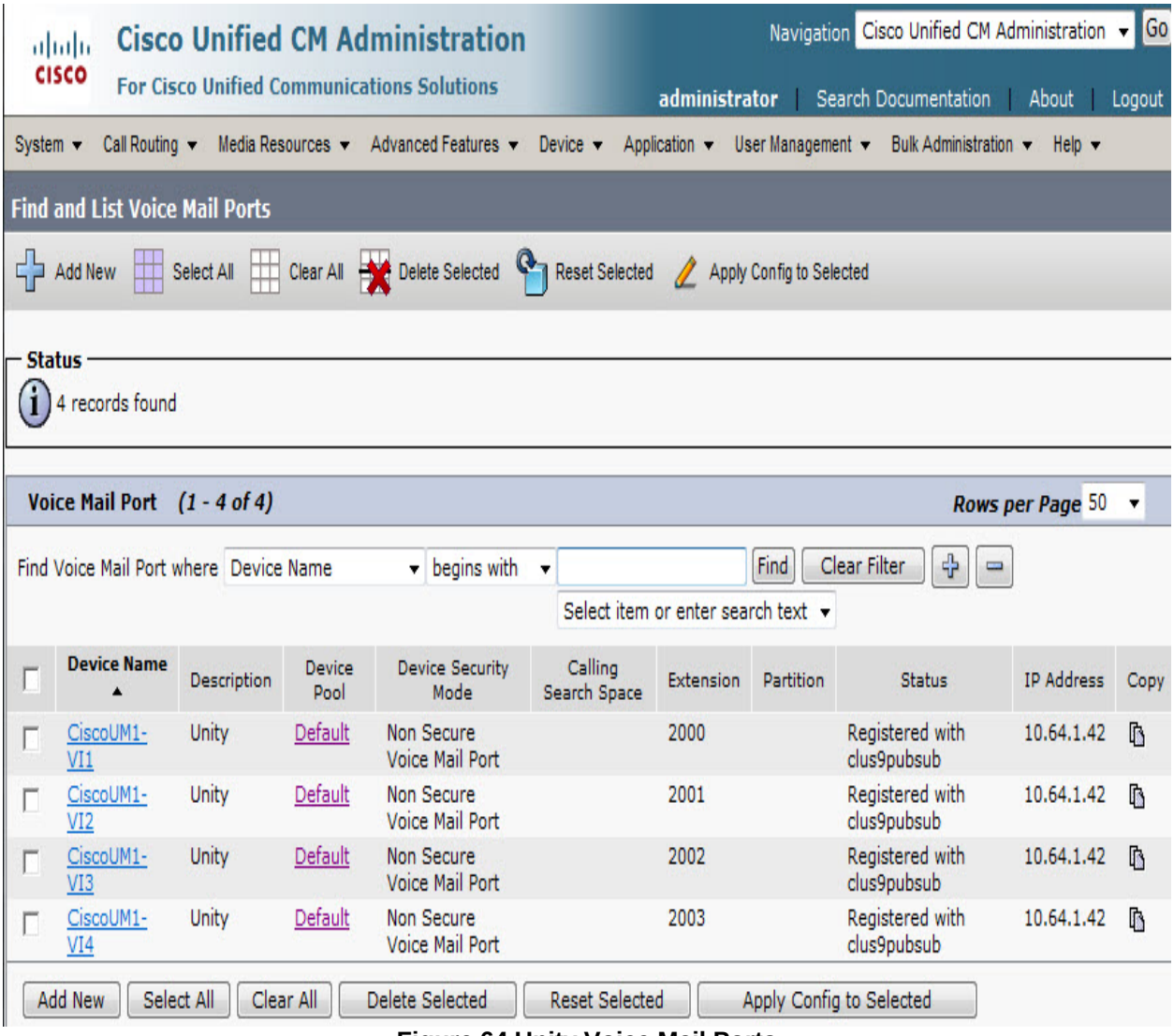


Figure 64 Unity Voice Mail Ports

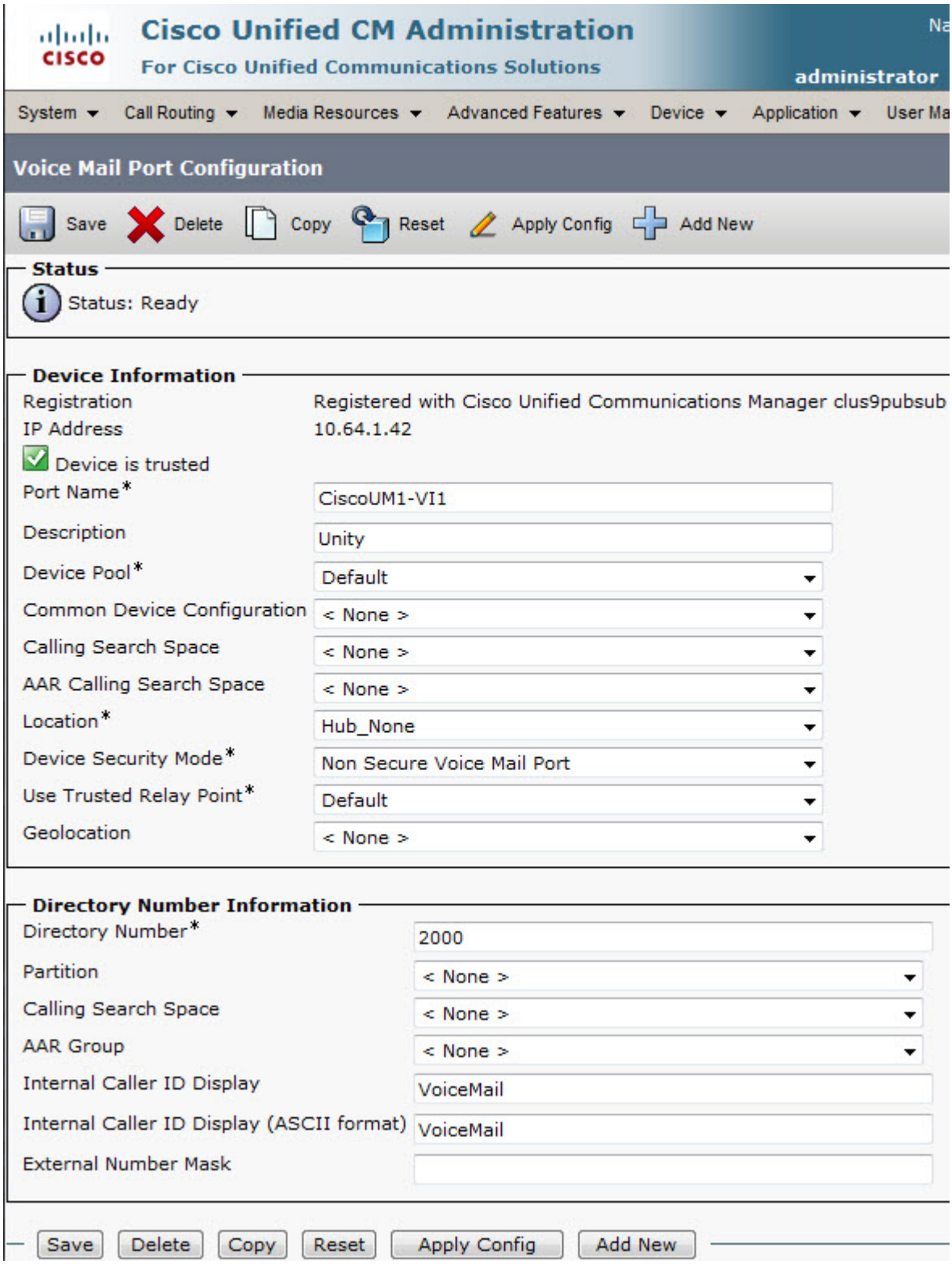


Figure 65 Unity Voice Mail Ports (Cont.)

6.3.20 Message Waiting Indicator

Navigate to Advanced Features > Voice Mail > Message Waiting

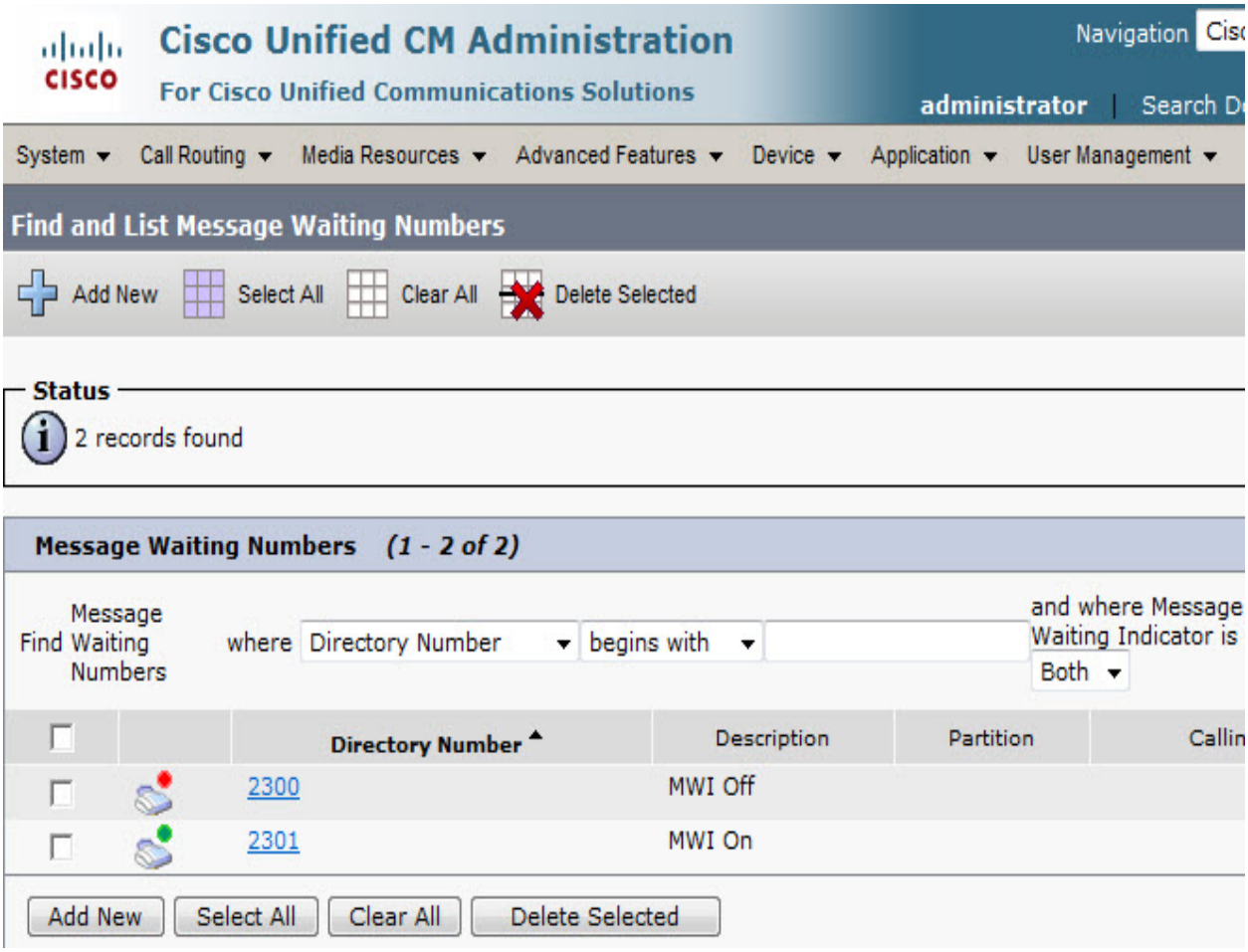


Figure 66 Message Waiting Indicator

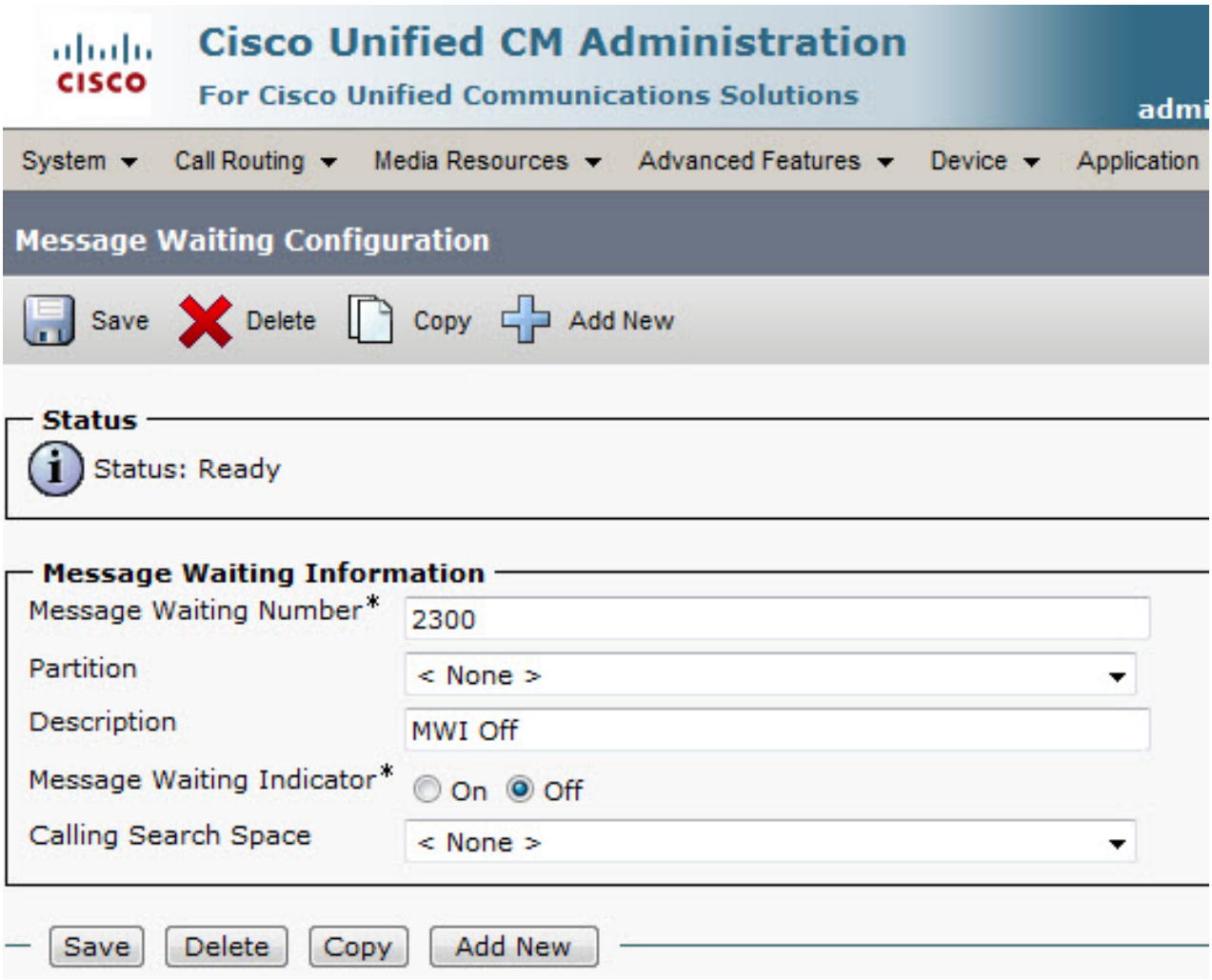


Figure 67 Message Waiting Indicator (Cont.)

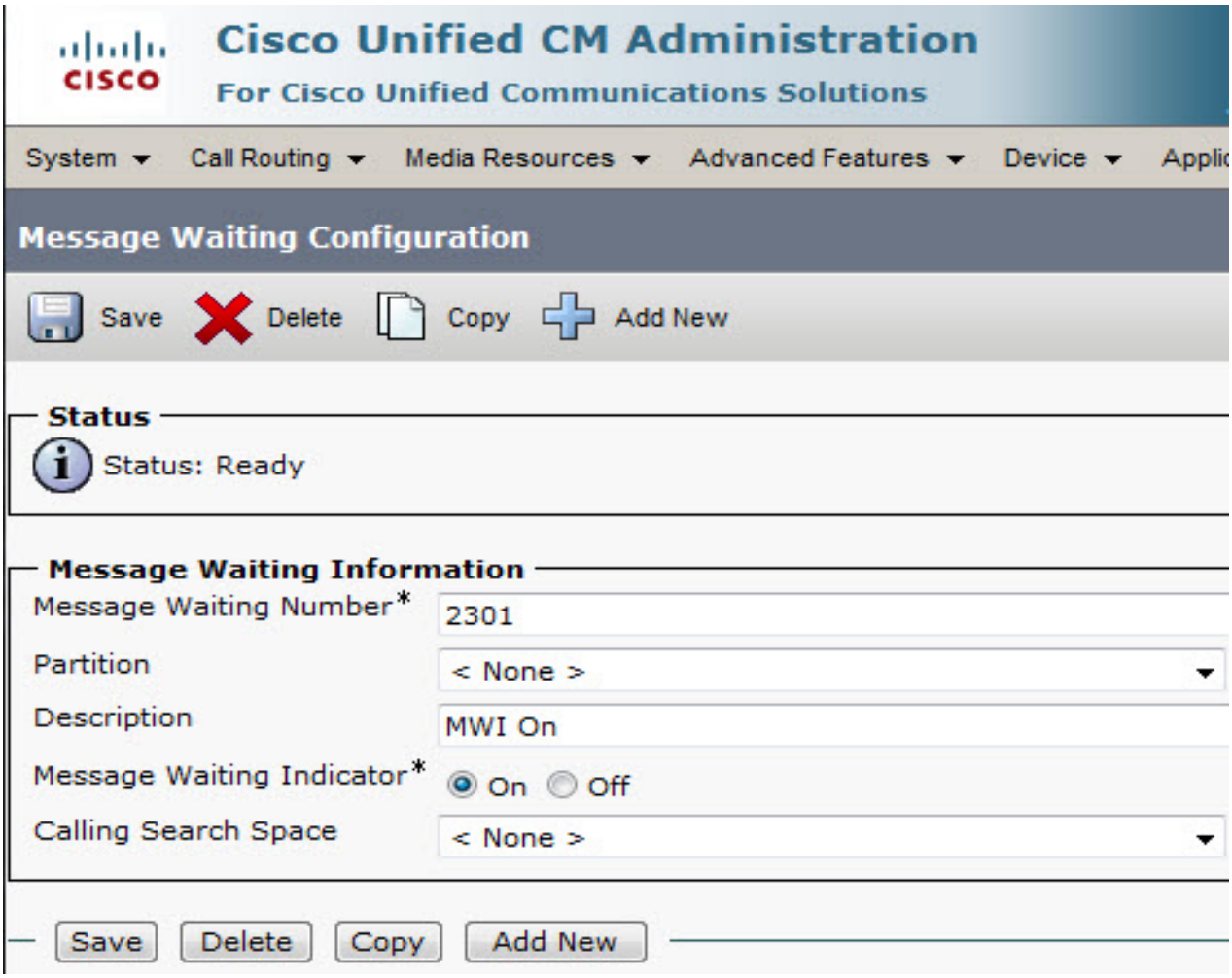


Figure 68 Message Waiting Indicator (Cont.)

6.3.21 Voice Mail Pilot

Navigate to Advanced Features > Voice Mail > Voice Mail Pilot

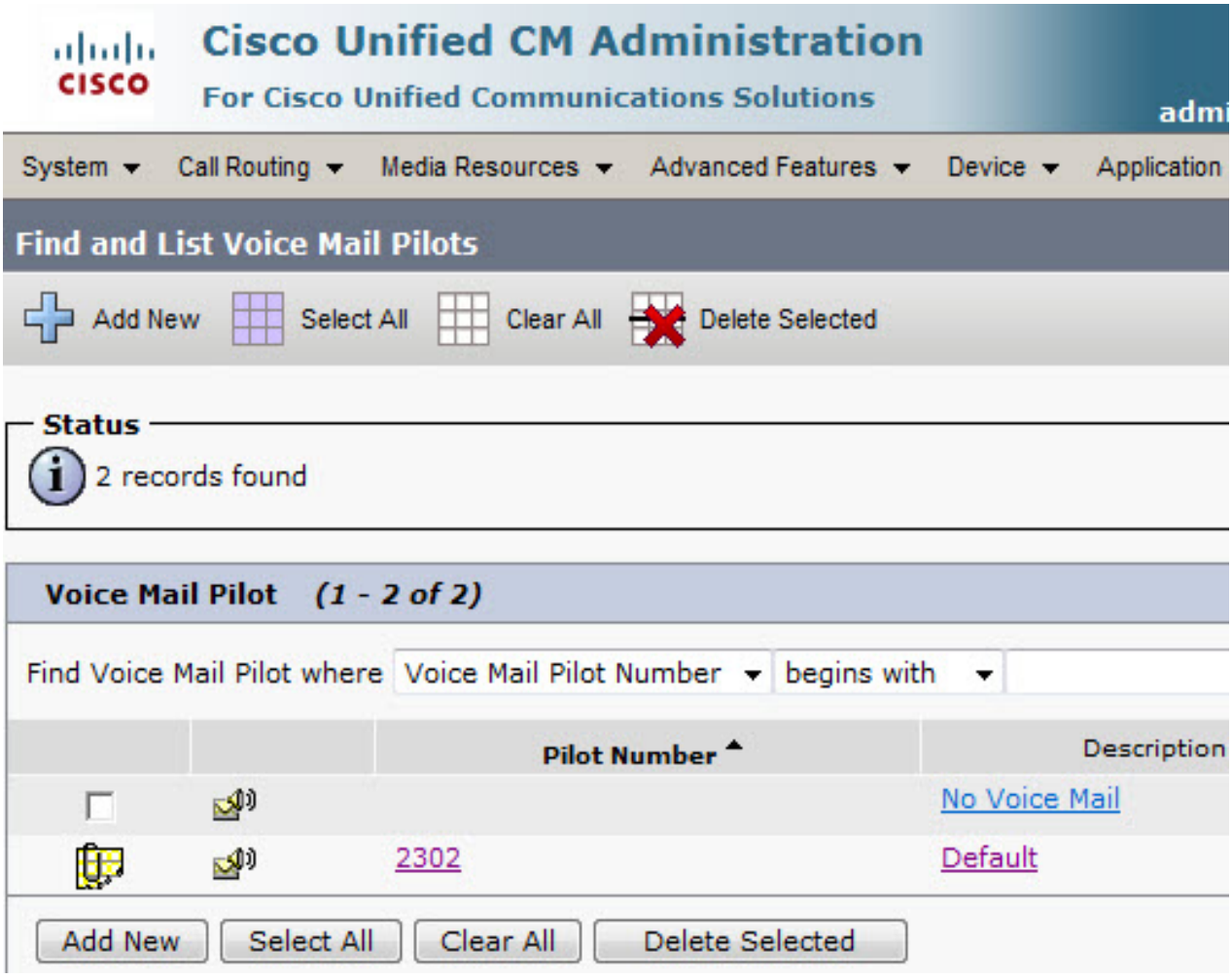


Figure 69 Voice Mail Pilot

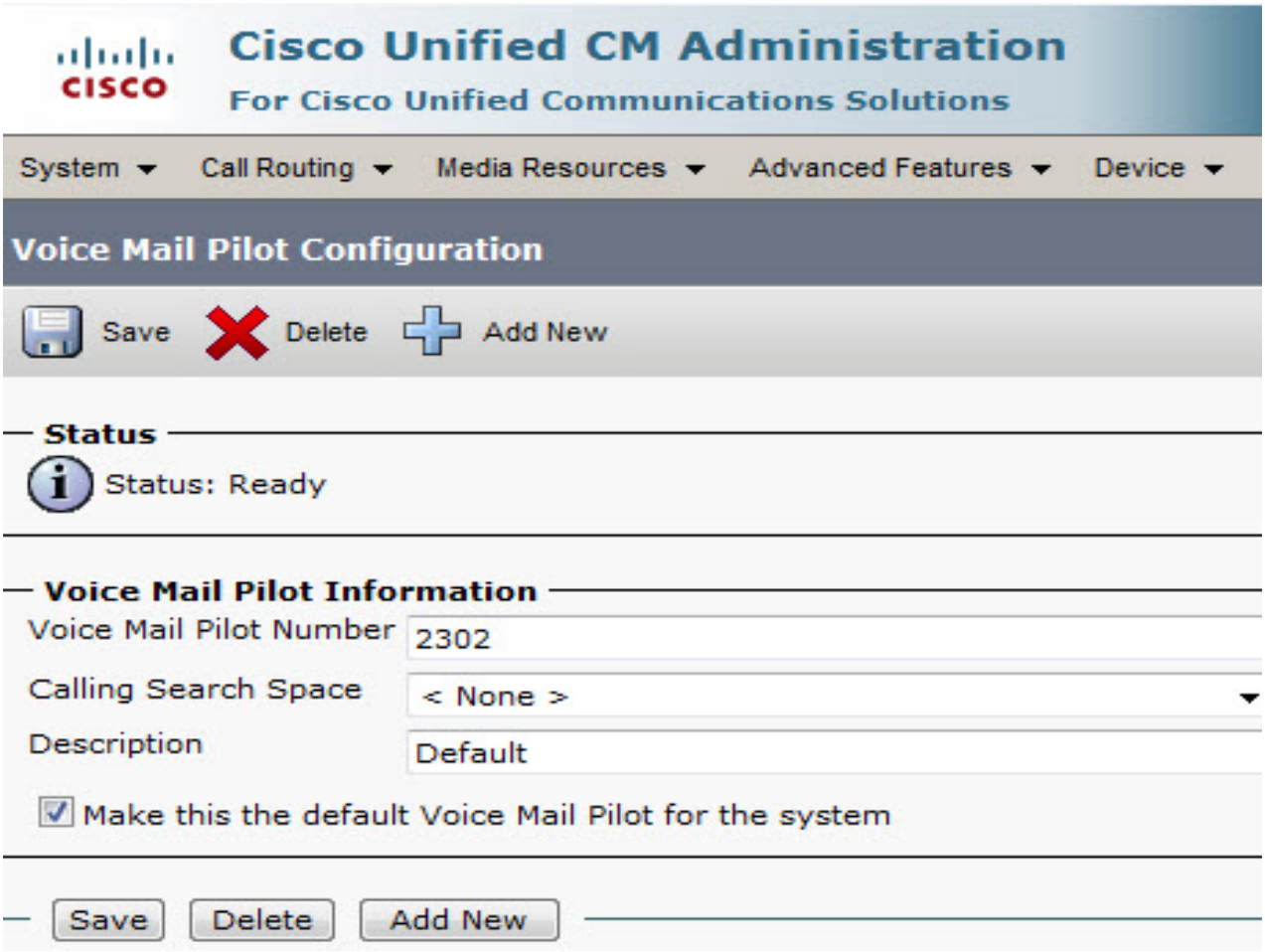


Figure 70 Voice Mail Pilot (Cont.)

6.3.22 Voice Mail Profile

Navigate to Advanced Features > Voice Mail > Voice Mail Profile

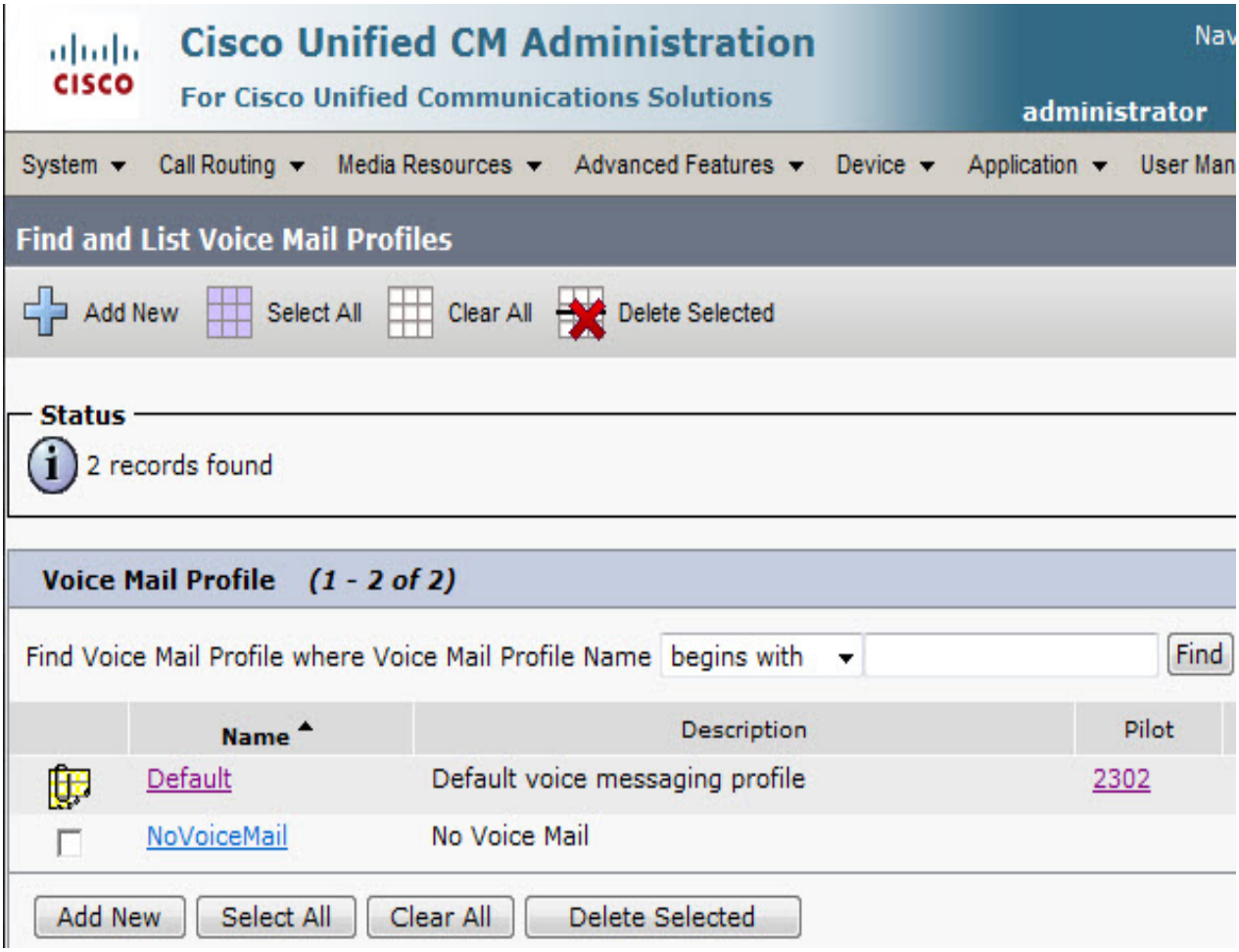


Figure 71 Voice Mail Profile

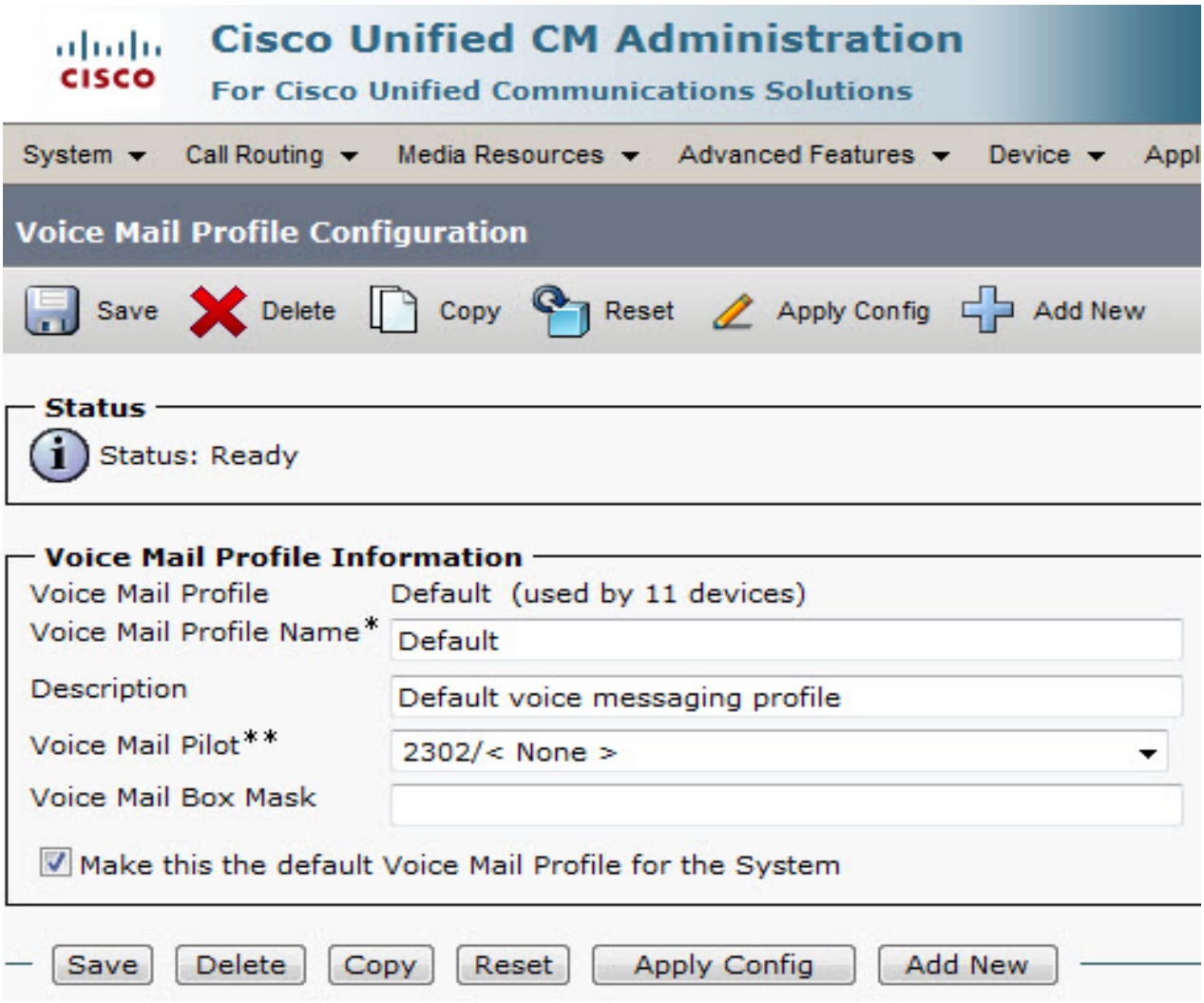


Figure 72 Voice Mail Profile (Cont.)

6.3.23 IP Phone Service

Navigate to Device > Device Settings > Phone Services

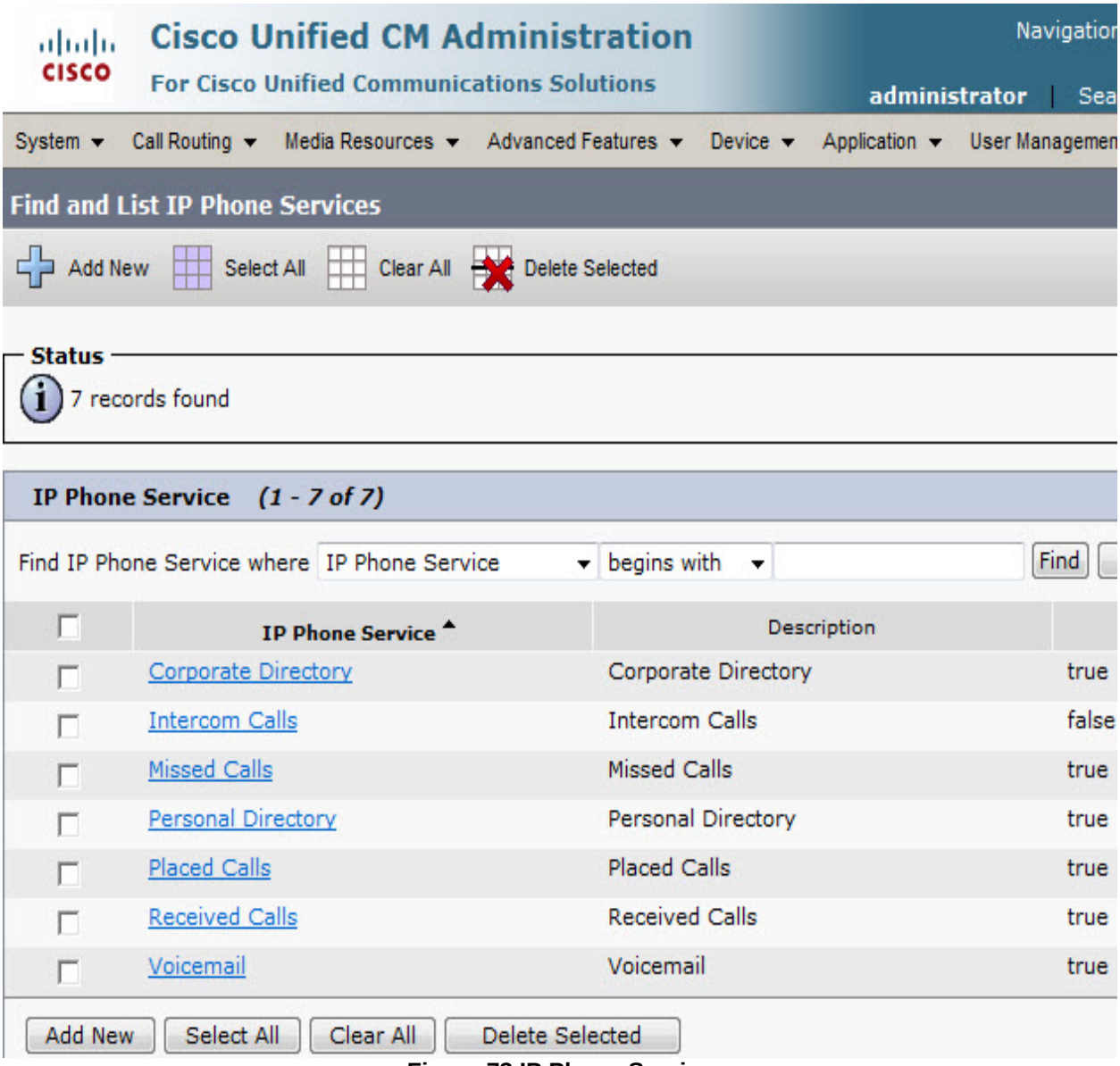


Figure 73 IP Phone Service

6.3.24 Linksys ATA

To set up a fax machine, a Linksys ATA was installed as a third-party SIP device. When the ATA is successfully registered with the CUCM, fax transmission capabilities should be established. The following sections detail these procedures.

1. Connect an Ethernet cable to the Ethernet port on the ATA.
2. After connecting the ATA to the LAN, launch the web browser and direct it to 192.168.0.1/advanced
3. A login screen should appear
4. Log in as user
5. Select Advanced
6. Log in as Administrator
7. Click the WAN Setup tab and assign an address
8. From the Connection Type drop-down menu, select Static IP
9. Assign an IP Address, Subnet Mask, and Default Gateway

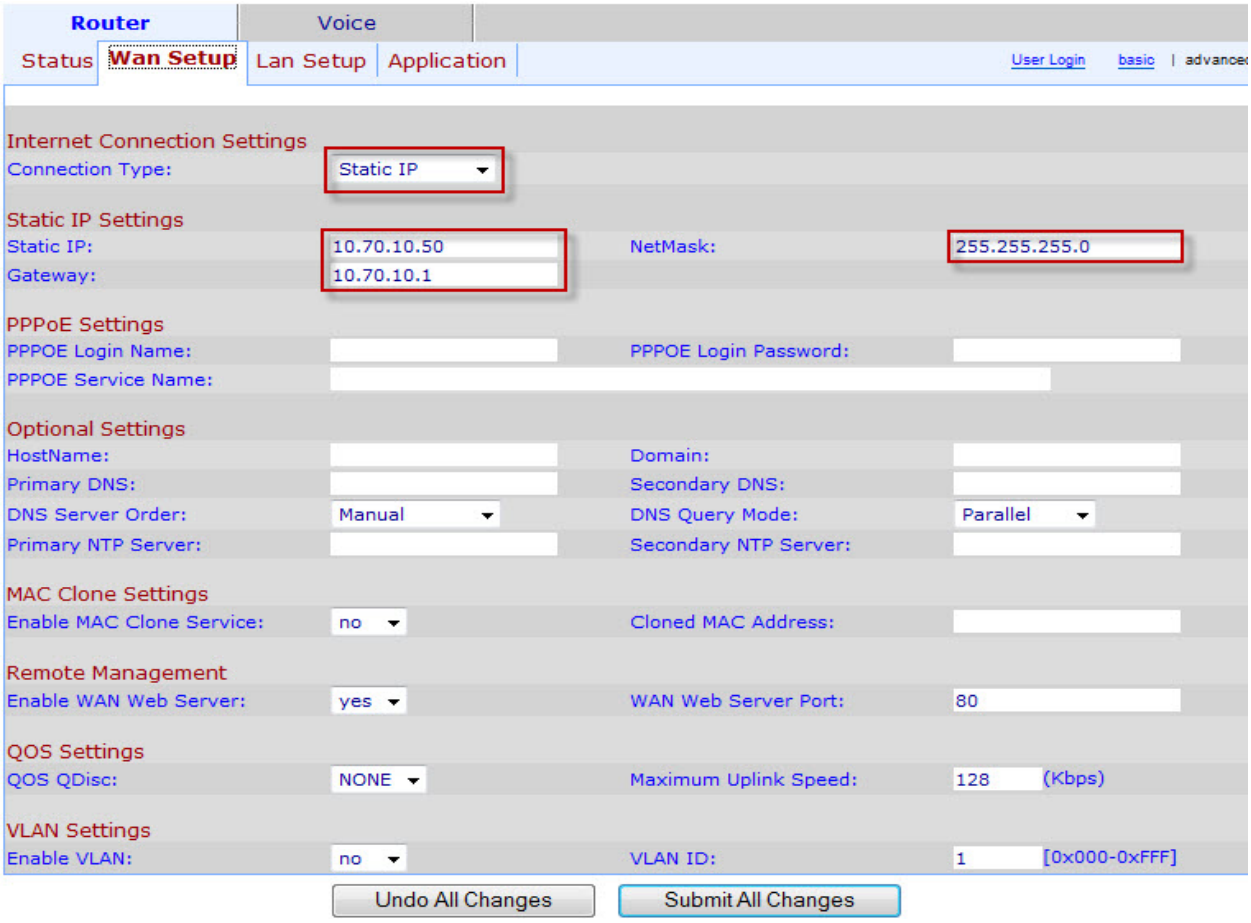


Figure 74 Linksys WAN Setup

10. At the top of the screen, select Voice and Line 1 (or Line 2)
11. In the Subscriber Information section, assign a User ID and Password
12. In the Proxy and Registration section, assign a Proxy address
13. Select the Submit All Changes button
14. The ATA will reboot with the new configuration applied

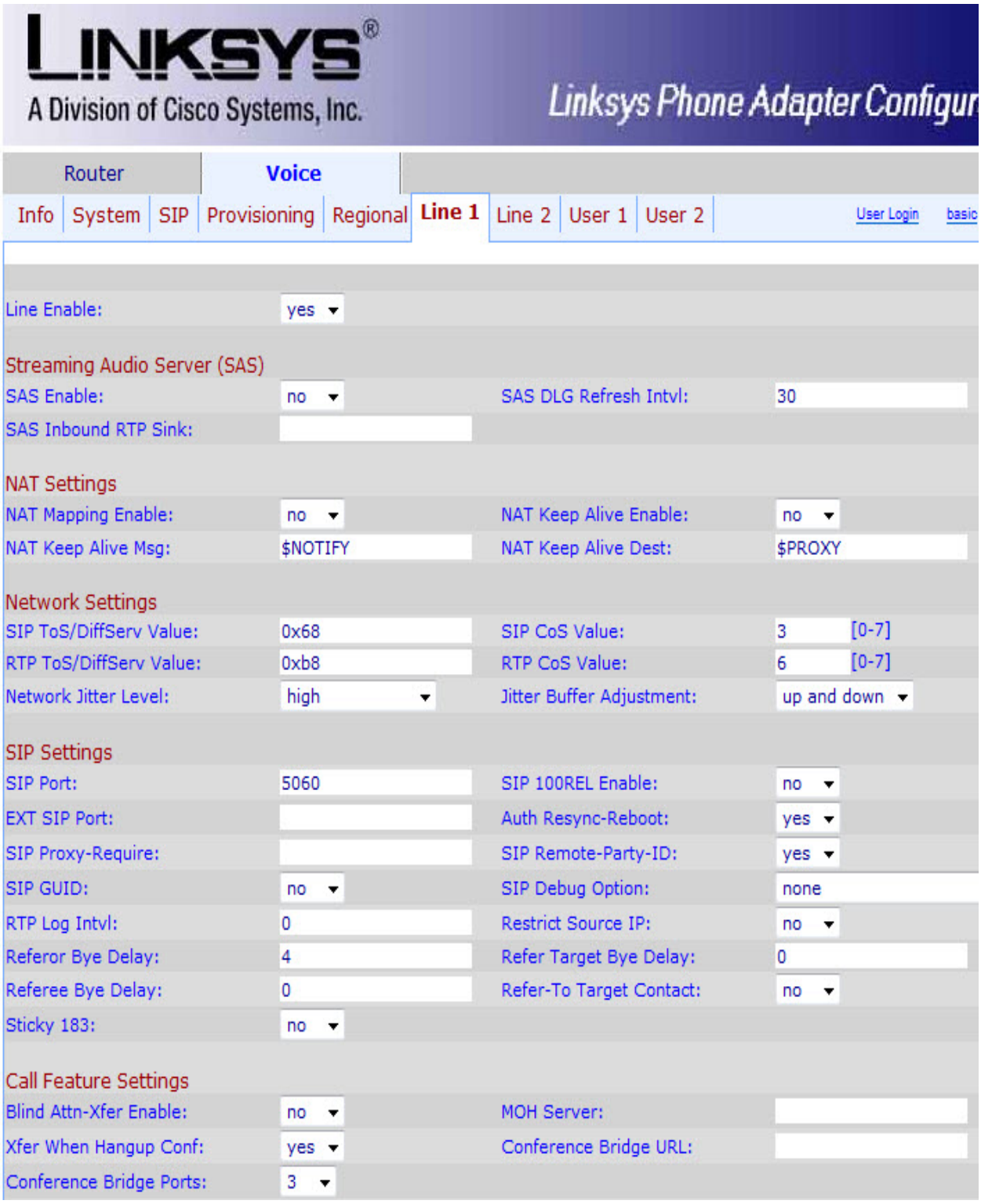
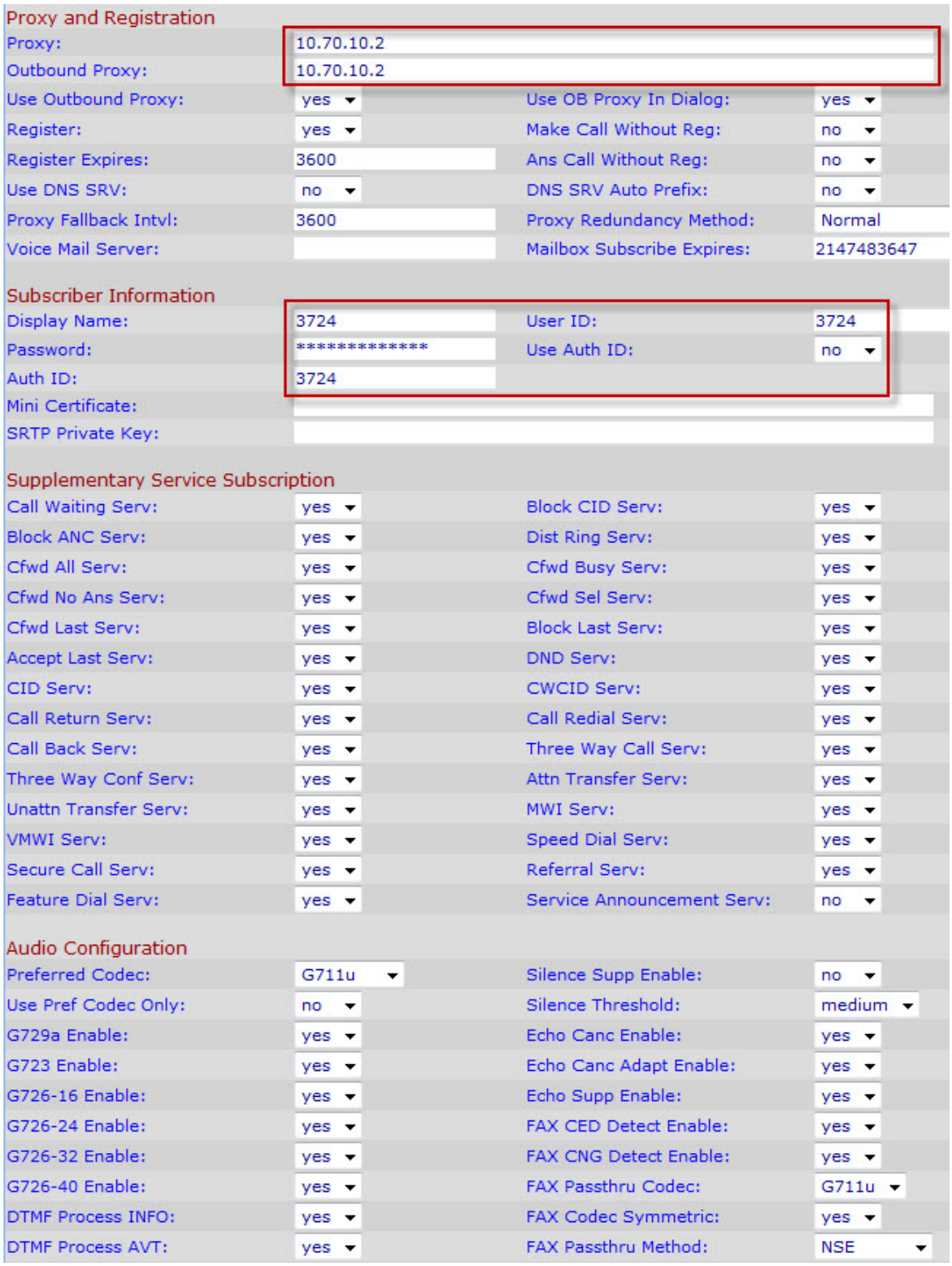


Figure 75 Linksys Line Configuration



Proxy and Registration			
Proxy:	10.70.10.2		
Outbound Proxy:	10.70.10.2		
Use Outbound Proxy:	yes	Use OB Proxy In Dialog:	yes
Register:	yes	Make Call Without Reg:	no
Register Expires:	3600	Ans Call Without Reg:	no
Use DNS SRV:	no	DNS SRV Auto Prefix:	no
Proxy Fallback Intvl:	3600	Proxy Redundancy Method:	Normal
Voice Mail Server:		Mailbox Subscribe Expires:	2147483647
Subscriber Information			
Display Name:	3724	User ID:	3724
Password:	**********	Use Auth ID:	no
Auth ID:	3724		
Mini Certificate:			
SRTP Private Key:			
Supplementary Service Subscription			
Call Waiting Serv:	yes	Block CID Serv:	yes
Block ANC Serv:	yes	Dist Ring Serv:	yes
Cfwd All Serv:	yes	Cfwd Busy Serv:	yes
Cfwd No Ans Serv:	yes	Cfwd Sel Serv:	yes
Cfwd Last Serv:	yes	Block Last Serv:	yes
Accept Last Serv:	yes	DND Serv:	yes
CID Serv:	yes	CWCID Serv:	yes
Call Return Serv:	yes	Call Redial Serv:	yes
Call Back Serv:	yes	Three Way Call Serv:	yes
Three Way Conf Serv:	yes	Attn Transfer Serv:	yes
Unattn Transfer Serv:	yes	MWI Serv:	yes
VMWI Serv:	yes	Speed Dial Serv:	yes
Secure Call Serv:	yes	Referral Serv:	yes
Feature Dial Serv:	yes	Service Announcement Serv:	no
Audio Configuration			
Preferred Codec:	G711u	Silence Supp Enable:	no
Use Pref Codec Only:	no	Silence Threshold:	medium
G729a Enable:	yes	Echo Canc Enable:	yes
G723 Enable:	yes	Echo Canc Adapt Enable:	yes
G726-16 Enable:	yes	Echo Supp Enable:	yes
G726-24 Enable:	yes	FAX CED Detect Enable:	yes
G726-32 Enable:	yes	FAX CNG Detect Enable:	yes
G726-40 Enable:	yes	FAX Passthru Codec:	G711u
DTMF Process INFO:	yes	FAX Codec Symmetric:	yes
DTMF Process AVT:	yes	FAX Passthru Method:	NSE

Figure 76 Linksys Line Configuration (Cont.)

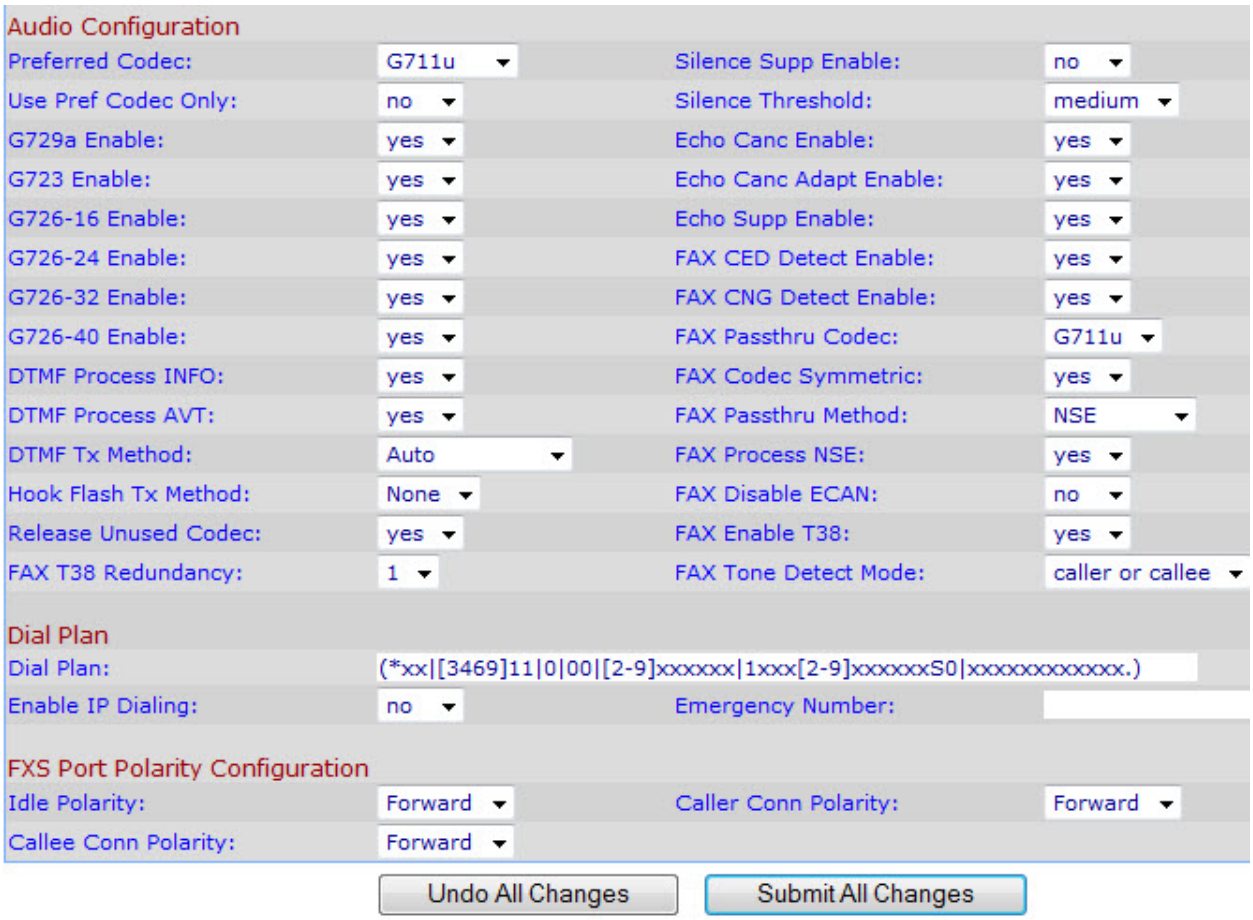


Figure 77 Linksys Line Configuration (Cont.)

When the Linksys configuration is complete, it must register with CUCM. The steps for registering the ATA with CUCM are detailed below.

A device must be added. The ATA is added as a phone.
1. Navigate to Device > Phone > Add New
2. From the Phone Type drop-down menu, select Third Party SIP Device (Advanced)
3. Select Next
4. Populate the MAC Address field with the mac address of the ATA
5. Assign a Device Pool: Default
6. From the Phone Button Template drop-down menu, select Third-party SIP Device (Advanced)
7. From the Common Phone Profile drop-down menu, select Standard Common Phone Profile
8. In the Protocol Specific Information section, choose Third-party SIP Device Advanced-Standard SIP Non-Secure Profile in the Device Security Profile drop-down menu
9. Choose a SIP Profile: Cox SIP Profile
9. All other settings are default values
10. Select Save
11. Click on the Apply Config button and select OK
12. The screen will be updated with Association Information on the upper left side
13. On the left side of the Phone Configuration screen (Figure 78) under Modify Button Items, click on Line 1
14. Assign a directory number (Figure 80)
15. Select save

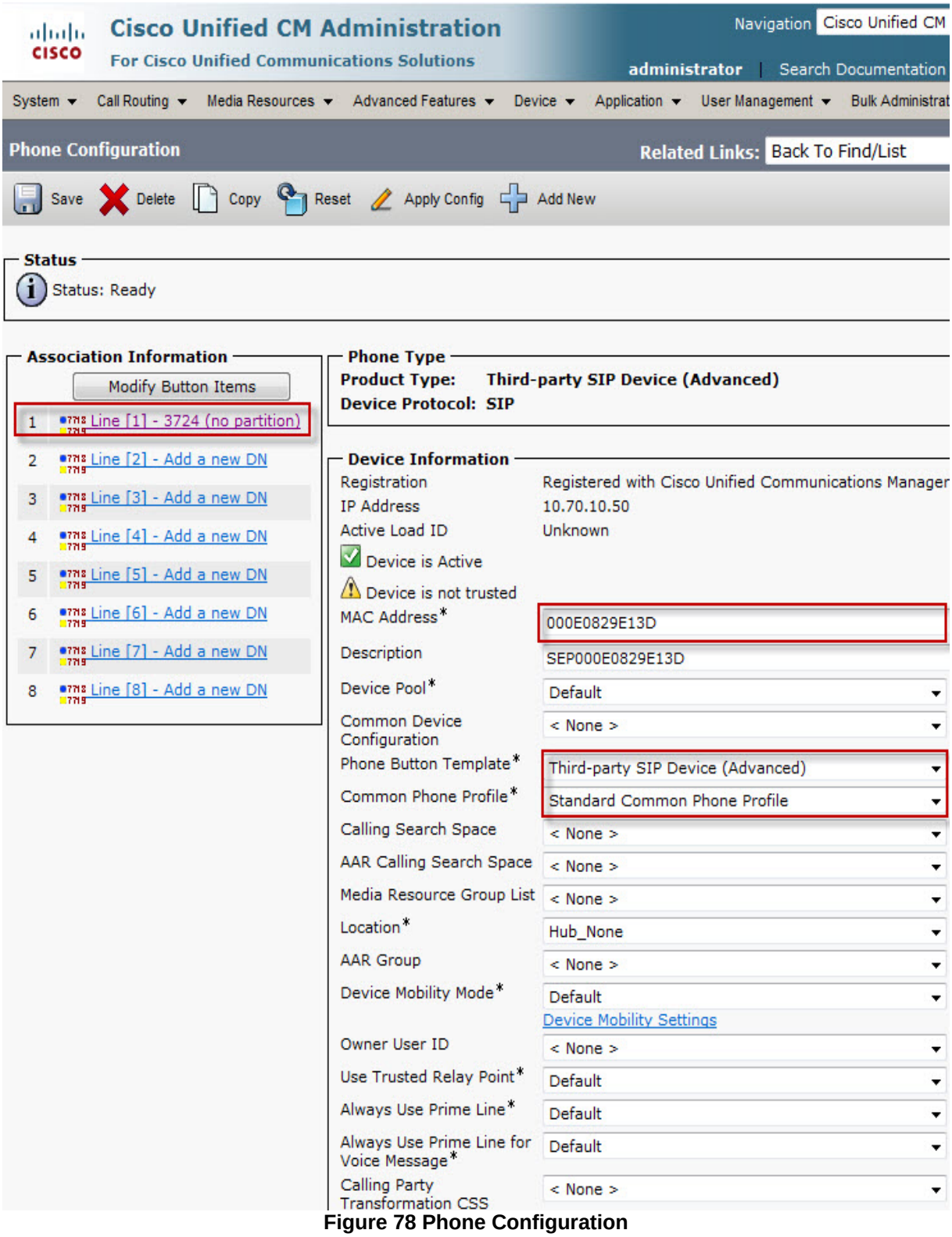


Figure 78 Phone Configuration

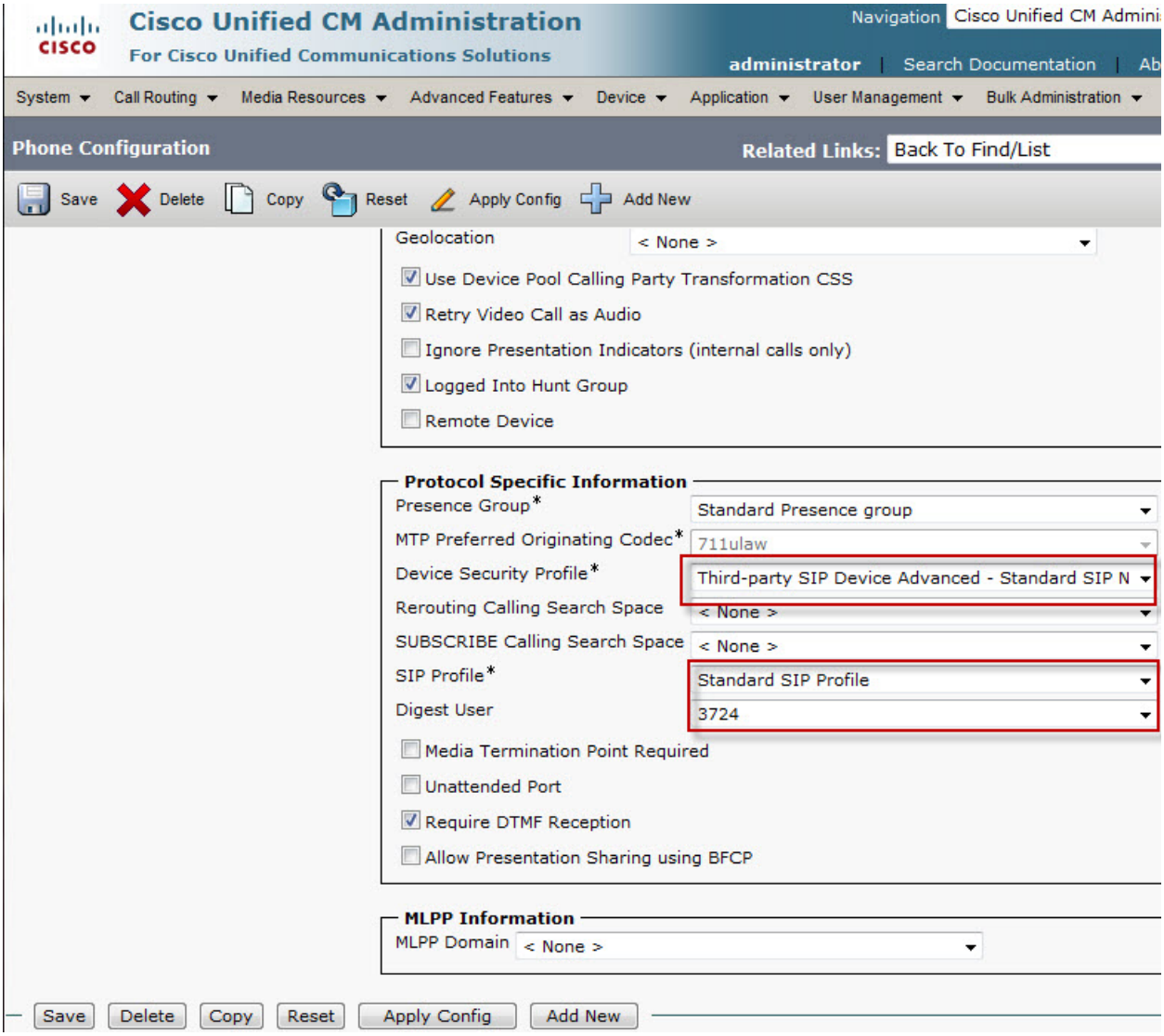


Figure 79 Phone Configuration (Cont.)

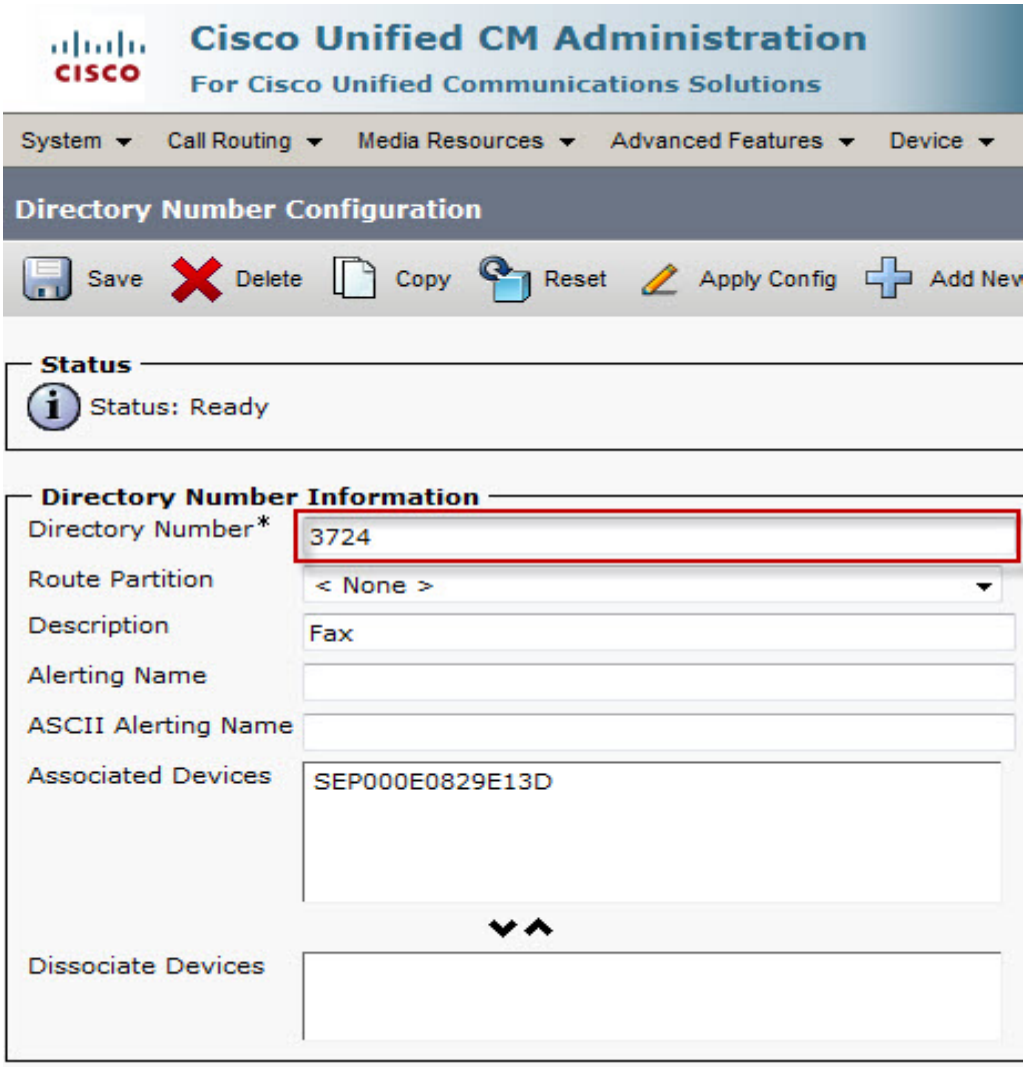


Figure 80 Directory Number

An End User must be created

1. Navigate to User Management > End User > Add New
2. Assign a User ID and Last Name. All other values are default setings
3. In the Device Information section, click Device Association. The device name should appear
4. Select Save

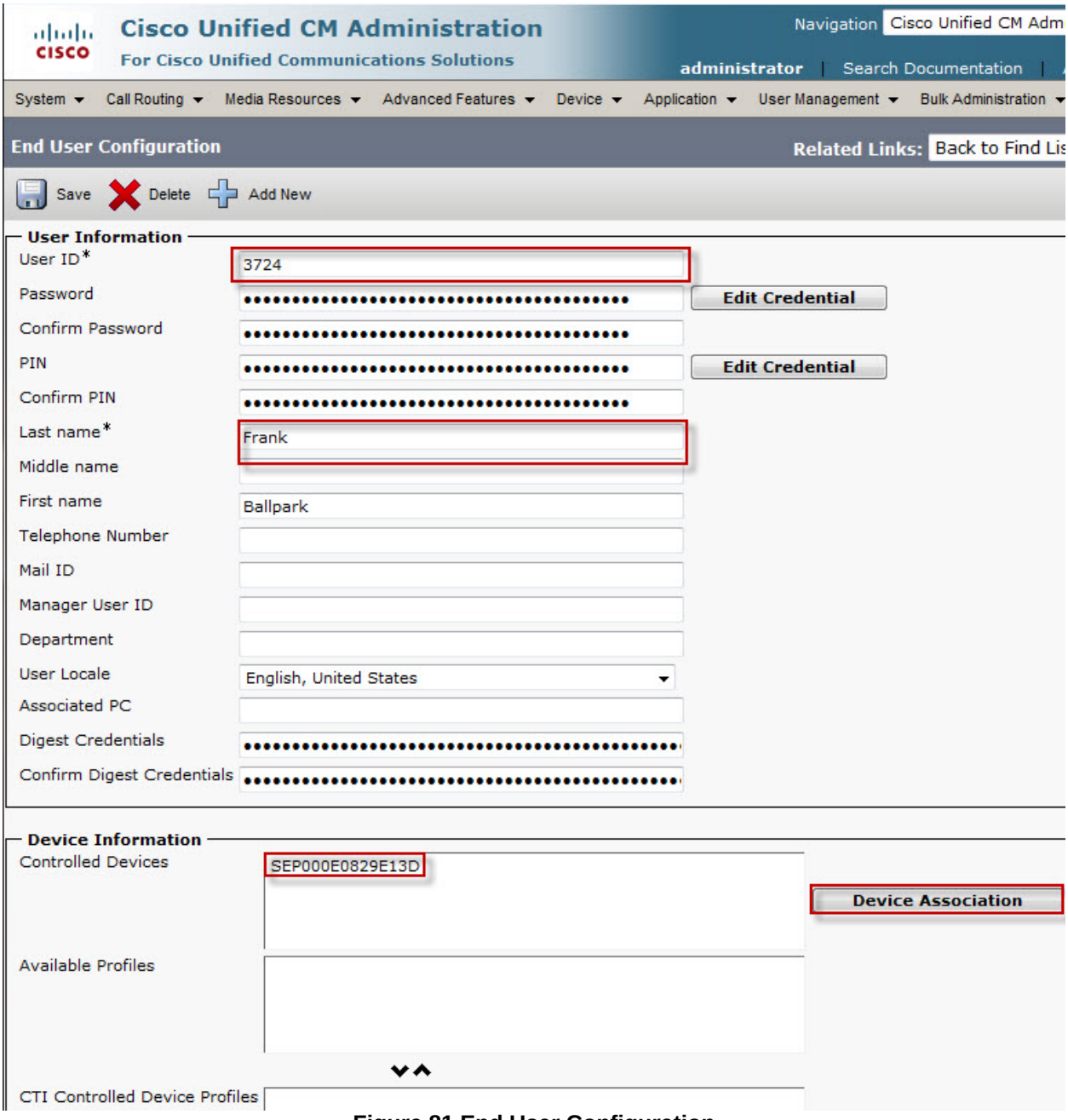


Figure 81 End User Configuration

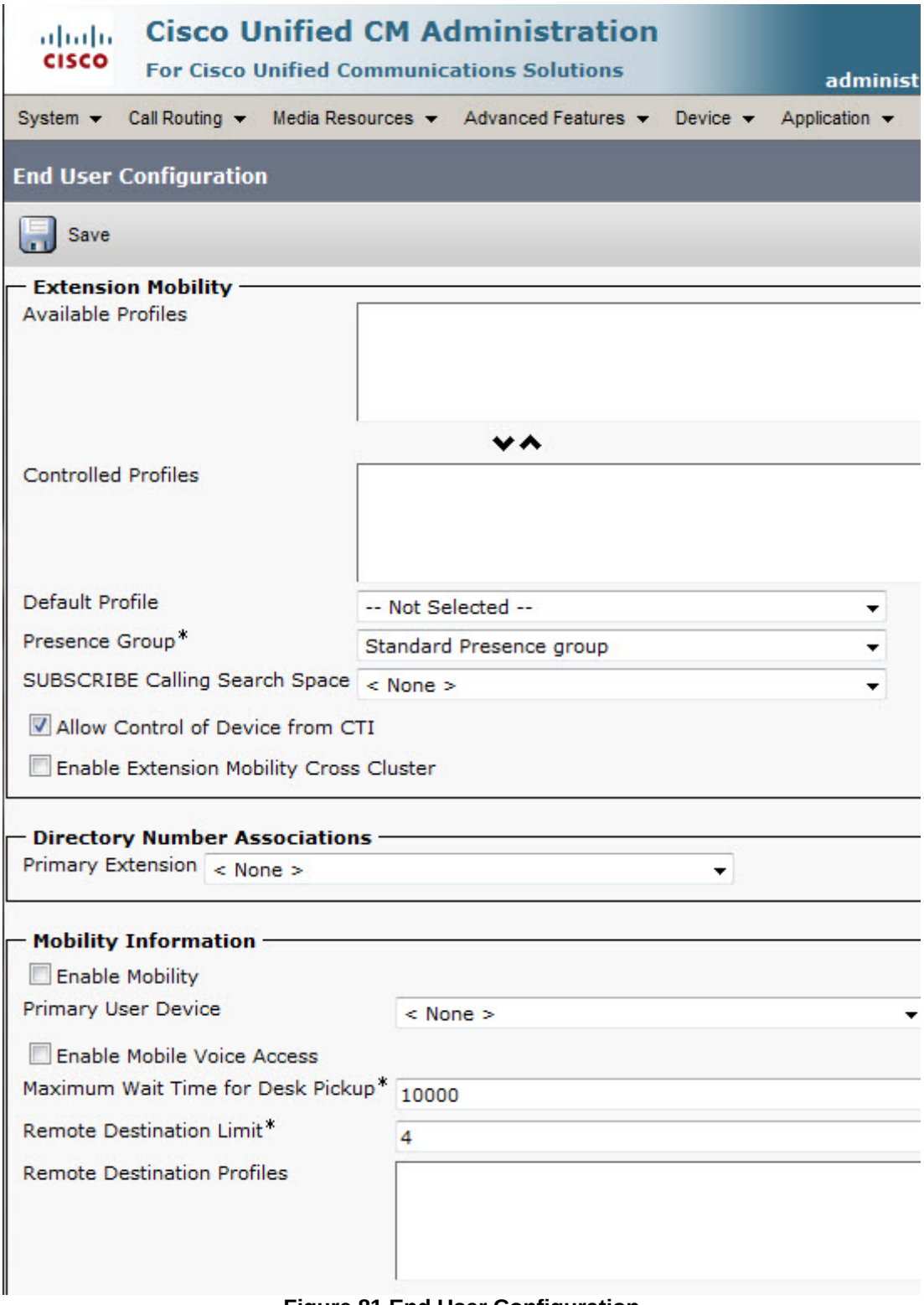


Figure 81 End User Configuration