Application Note

Mitel 3300 ICP Release 7.1 Using SIP to Cisco Unified Communications Manager 6.1

June 18, 2008 - Initial Version

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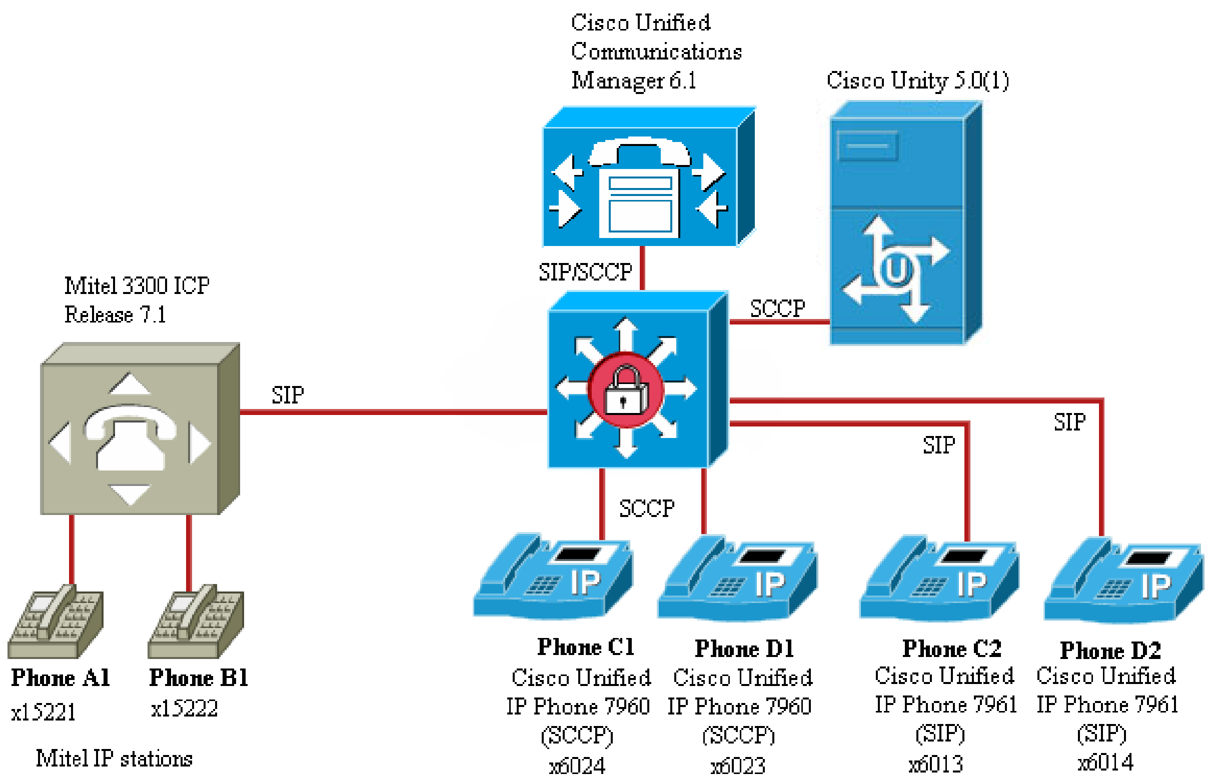
Introduction

- This application note provides interoperability information and documented configurations for a SIP trunk connection between a Mitel 3300 ICP Release 7.1 and Cisco Unified Communications Manager 6.1. End-to-end calls were made to verify basic calls and other features. Figure 1 shows the integration topology.
- The following basic call and supplementary services features were verified: proper establishing and disconnecting of calls; calling name and number presentation and restriction; alerting name; call transfer (consultation and early-attended); call forwarding (all, busy, and no reply); call hold; call conference; and voicemail access with MWI activation and deactivation. Please note that this document does not address performance and scalability, which are part of a broader criteria for a deployment-ready solution.

Network Topology

Figure 1. Basic Call Setup



Limitations

These are the known limitations, caveats, or integration issues with the specified configuration:

- The Mitel 3300 ICP SIP Session Timer value must equal or exceed the Cisco Unified Communications Manager 6.1 SIP Min-SE Value service parameter. The Min-SE Value can be found under the Cisco Unified Communications Manager Service Parameters form. The Mitel SIP Session Timer can be found on the SIP Peer Profile form. Screenshots of both of these forms are provided on pages 11and 126, respectively.
- The Cisco Unified IP Phones and Mitel IP phones depicted in this integration do not support a true blind transfer, but rather, an ”early attended” transfer. The difference is that a party initiating a blind transfer drops out before the final destination starts ringing. To do a blind transfer, the IP phones must have a specific button or soft key. An early attended transfer initiates the transfer, gets ringback, and then completes the transfer before the final destination answers. Some Cisco Unified IP Phones (for example, 7940 and 7960 with SIP loads) do support blind transfer.
- The Mitel 3300 ICP did not have a MoH source, so MoH was not verified in this configuration. Cisco Unified Communications Manager did have a MoH source, and it worked properly with both Cisco and Mitel extensions.
- A call from a Mitel extension to a Cisco Unified IP Phone that is forwarded to voice mail hits the ”open greeting” and not the proper voice mailbox. This is due to the diversion header difference between Cisco Unified Communications Manager and the Mitel 3300 ICP. Therefore, a full voice mail integration using the Mitel 3300 ICP voice mail system was not possible.
- A full voice mail integration using Cisco Unity was possible, and Message Waiting Indication (MWI) works properly. However, the Mitel 5220 IP handset has a feature where a ”message” soft key appears, and the user can use it to go directly to voice mail. This soft key does not appear for voice mail and/or MWI from Unity. The Mitel user has to dial the VM pilot number to access voicemail. This could easily be programmed into a button on the Mitel handset.
- Differences between the SIP implementations of Calling / Connected Line Identification on the Mitel 3300 ICP Release 7.1 and Cisco Unfied Communications Manager 6.1 affect a variety of interoperability test cases. Cisco Unfied Communications Manager 6.1 uses Remote Party ID for calling party information, and the Mitel 3300 ICP does not. Diversion headers are structured differently between the two systems. The following points summarize the differences and the limitations that arise from these differences.
    - The Mitel 3300 ICP does not use P-Asserted Identity (PAI) or Remote-Party-ID (RPID) for Caller ID information. The calling party information (both name and number) is in the ”From” and ”Contact” fields in the INVITE message. The called party information is in the ”To” (number only) and ”Contact” (both name and number) fields in the 180 Ringing message. The connected party information is in the ”To” (number only) and ”Contact” (both name and number) fields in the 200 OK message.
    - Cisco Unified Communications Manager 6.1 uses Remote-Party-ID (RPID) in addition to the the ”From” and ”Contact” fields for Caller ID information. The calling party information (both name and number) is in the ”From” field and RPID in the INVITE message. The calling number is also in the ”Contact” field. The called party information is in the ”To” (number only) and ”Contact” fields (number only) and in the RPID (both name and number) in the 180 Ringing and in 183 Session Progress messages. The connected party information is in the ”To” (number only) and ”Contact” fields (number only) and in the RPID (both name and number) in the 200 OK message.
    - The Mitel 3300 ICP Release 7.1 and Cisco Unified Communications Manager 6.1 both send forwarding info in a Diversion header. However the Mitel diversion header has the structure: **Diversion:<sip:15221@172.20.220.254>;reason=unconditional;counter=1**, while the Cisco Unified Communications Manager diversion header has this structure: **Diversion: “Alert-C2” <sip:6013 @172.20.214.254>;reason=user-busy;privacy=off,screen=yes**. This issue impacts Calling Name/Number ID on forwarded calls and voice mail scenarios where a Mitel extension calls a Cisco Unified IP Phone that forwards to Mitel voice mail. This causes the originating Mitel extension to hit the ”open greeting” in Mitel voice mail, instead of the proper voice mailbox for the Cisco Unified IP Phone.
    - Connected Name and Number are not supported between the two systems.
    - Calling Line ID is not supported on Network/External Transfers.

- Original calling number is not displayed on the final destination on calls originating from a Cisco Unified IP Phone to a Mitel extension and forwarded locally to another Mitel extension (for example, C1 calls A1 and A1 forwards to B1.) The originator's name is displayed, however, after answer. This is not an issue for the reverse call flow (for example, A1 calls C1, and C1 forwards to D1). The Mitel IP phones display the originator information after a forwarded call is answered, even if all parties are local.
- Neither forwarding name nor number is displayed on the final destination on calls originating from a Mitel extension to a Cisco Unified IP Phone and forwarded externally to another Mitel extension (for example, A1 calls C1, and C1 forwards to B1). This is not an issue for the reverse call flow (for example, C1 calls A1, and A1 forwards to D1). In those cases, forwarding number is displayed. This is likely due to the different structures in the diversion header.
- For Local Call Conferences originated from a network/external call (for example, C1 calls A1, and A1 conferences in B1; A1 calls C1, and C1 conferences in D1), the Caller ID was not updated properly. The conferencing phone and the third phone properly displayed "conference" during a three-way conference call, but the originating phone displayed the dialed number. After the conferencing phone dropped out, the third phone displayed the originator's name (and number, if the third phone was a Cisco Unified IP Phone), but the originator still displayed dialed number.
- For Network/External Call Conferences originating from a local call (for example, C1 calls A1, and A1 calls D1; A1 calls C1, and C1 conferences in B1), the Caller ID was not properly updated. The conferencing phone properly displayed "conference" during a three-way conference call, but the originating phone displayed the dialed number, and the third phone displayed the conferencing party's name (and number, if the third phone was a Cisco Unified IP Phone). These disaplays did not change when the conferening phone dropped out.

System Components

Hardware Requirements

The following hardware is required:

- Two Cisco Media Convergence Servers 7800
- Catalyst 3560 PoE switch
- Two Cisco Unified IP Phones 7961
- Two Cisco Unified IP Phones 7960
- Mitel 3300 MX Controller
- Two Mitel 5220 IP phones

Software Requirements

The following software is required:

- Cisco Unified Communications Manager Release 6.1
- Cisco Unity Release 5.0(1)
- Mitel 3300 ICP Release 7.1 UR2

Features

This section lists supported and unsupported features.

Features Supported

- Basic Call
- Disconnect Supervision
- Calling Line (Number) Identification Presentation (CLIP)
- Calling Line (Number) Identification Restriction (CLIR)
- Calling Name Identification Presentation (CNIP)
- Calling Name Identification Restriction (CNIR)
- Consultation Transfer – Local and Network/External
- Early-Attended Transfer – Local and Network/External
- Call Forward Unconditional – Local and Network/External
- Call Forward on Busy – Local and Network/External
- Call Forward on No Answer – Local and Network/External
- Call Hold and Resume with MoH – See Limitations section on page 3.
- Call Conference – Local and Network/External – See Limitations section on page 3.
- Voice Mail integration with Cisco Unity, including MWI

Features Not Supported

- Connected Line (Number) Identification Presentation (COLP)
- Connected Name Identification Presentation (CONP)
- Alerting Name
- Blind Transfer
- Voice Mail Integration with the Mitel 3300 ICP as Message Center PINX

Configuration

This section contains configuration menus and commands and describes configuration sequences and tasks.

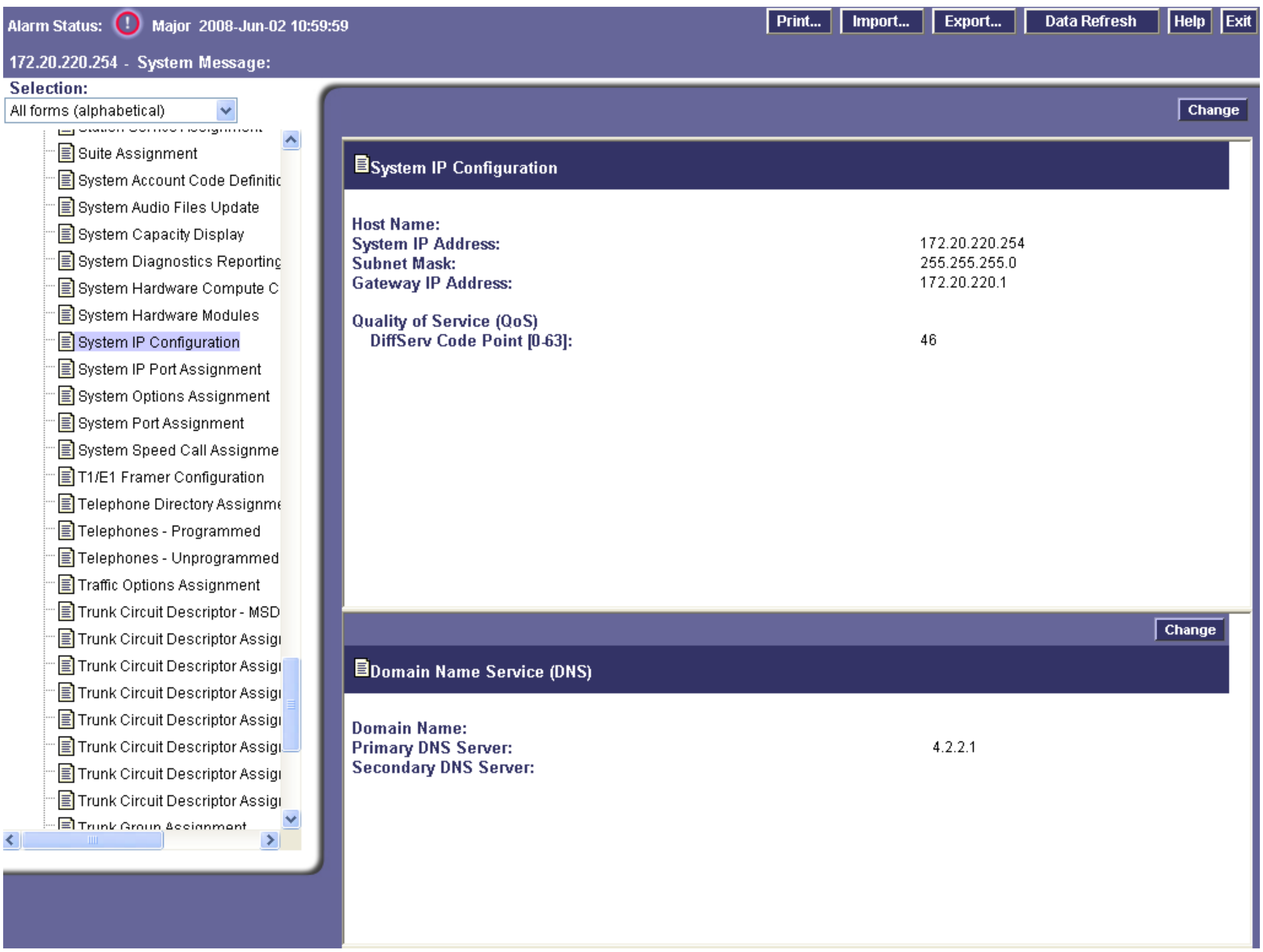
Configuring the Mitel 3300 ICP Release 7.1

Software Version

The specific software version was 7.1.5.13.

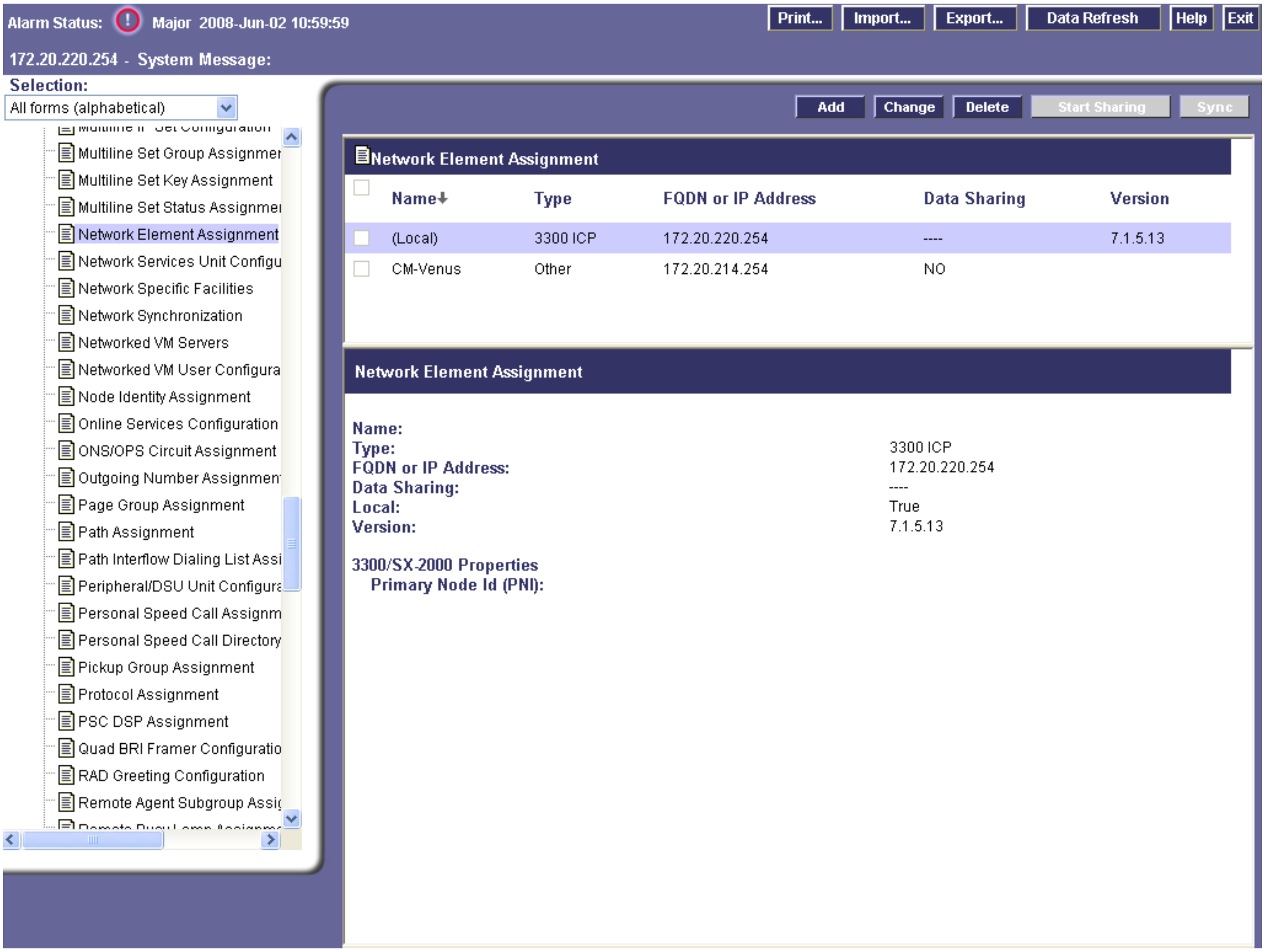
System IP Configuration

System IP Configuration – 1 of 1.

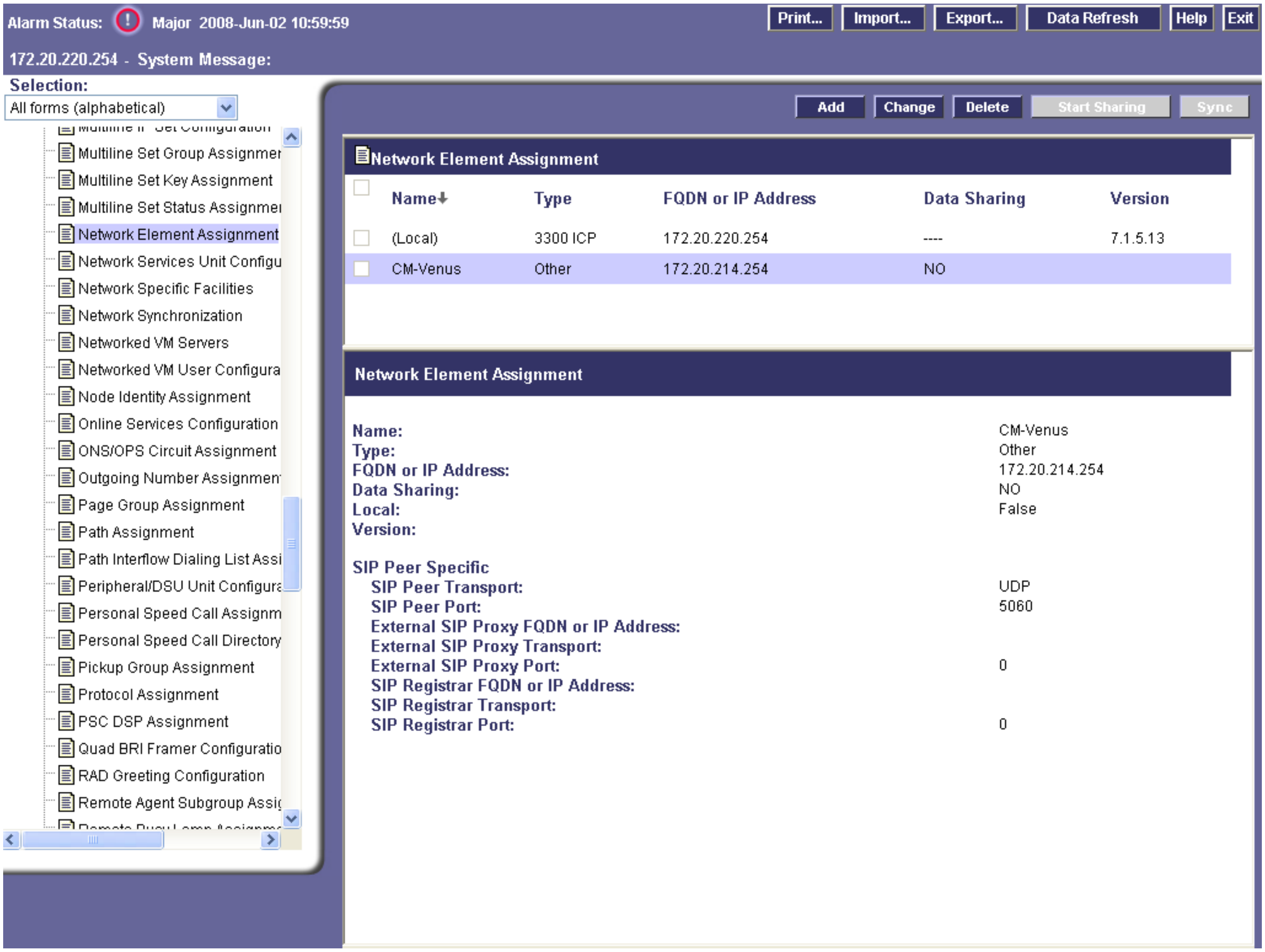


SIP Trunk Setup

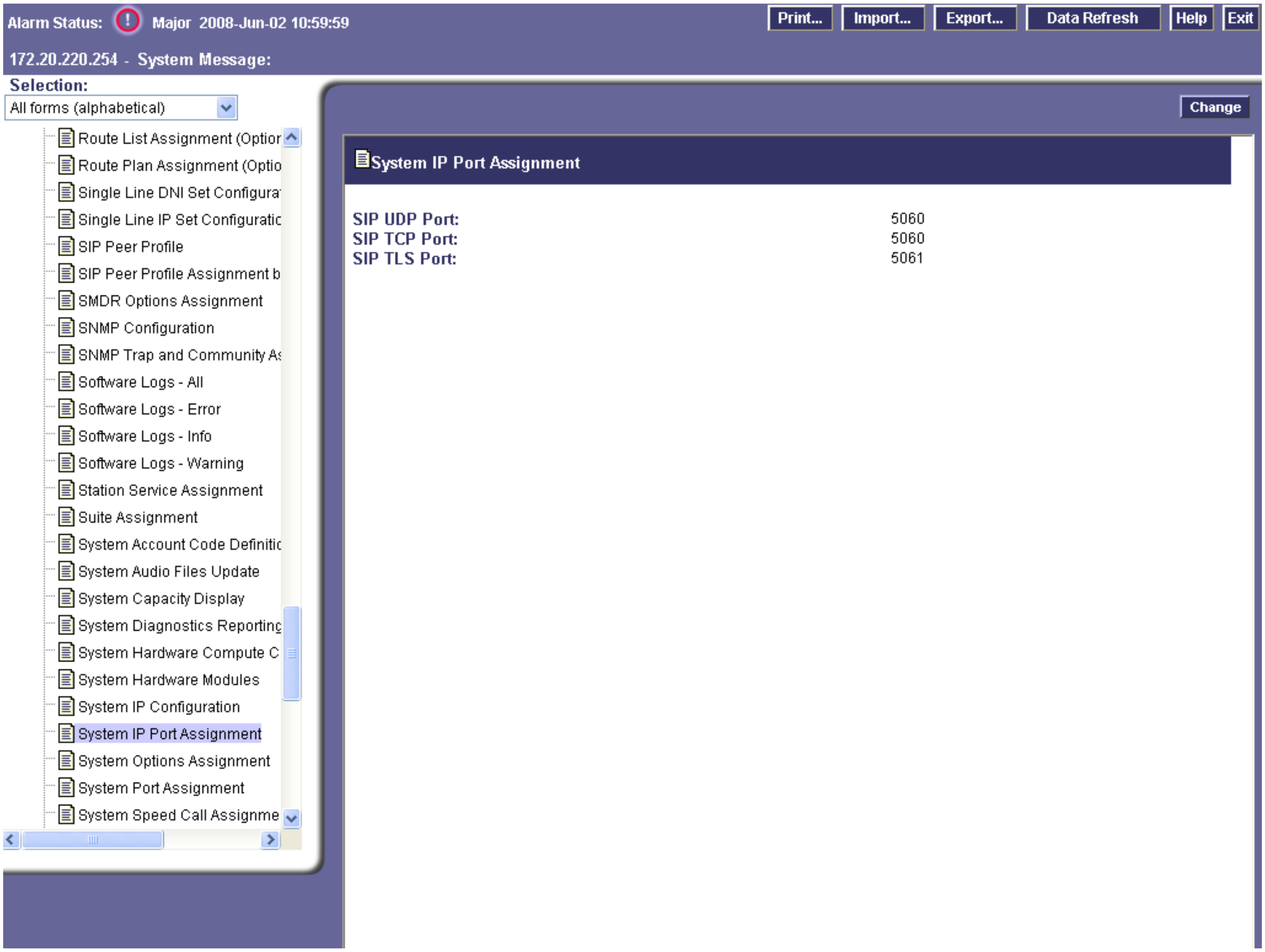
Network Element Assignment – 1 of 2.



Network Element Assignment – 2 of 2.

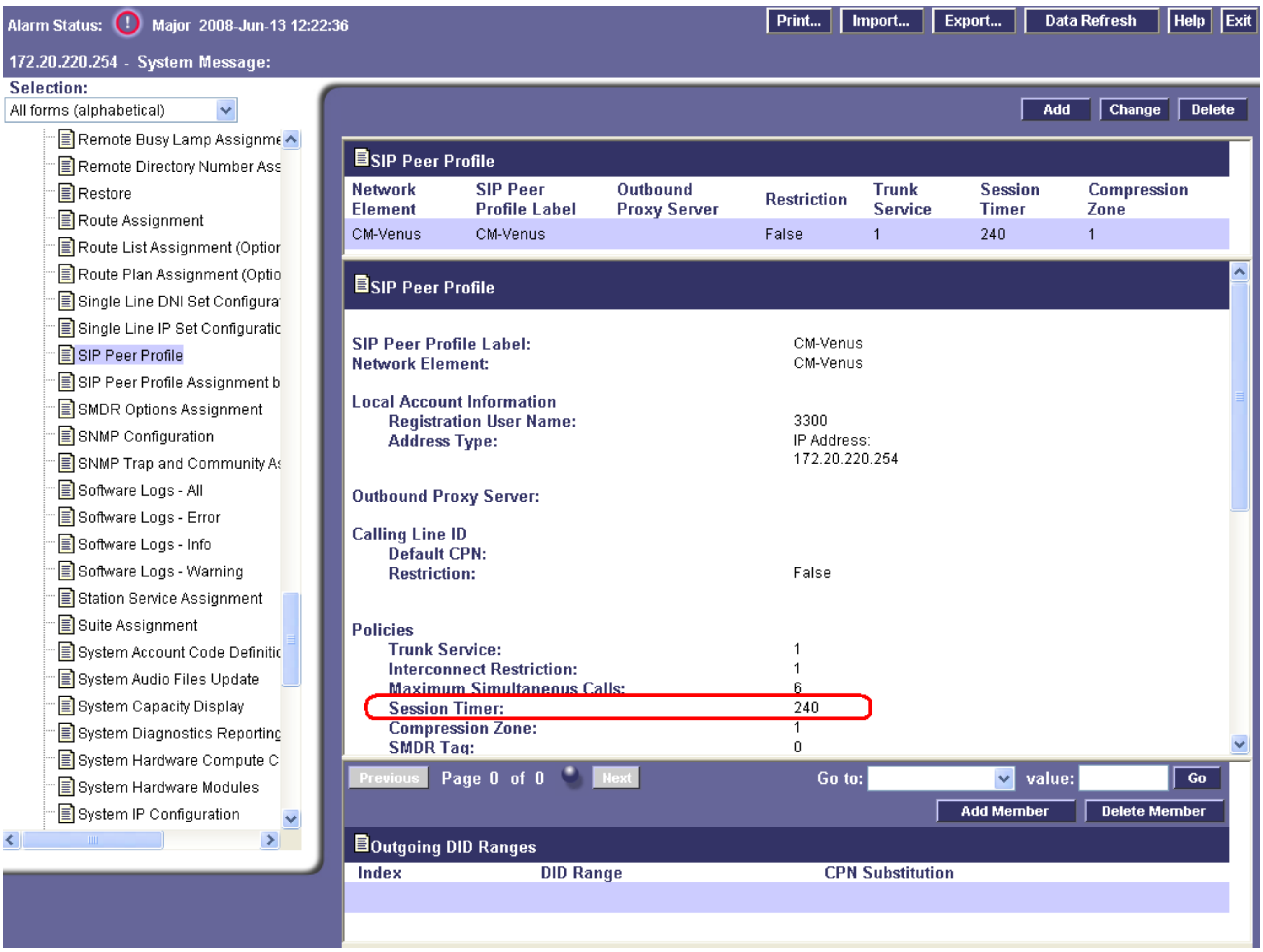


System IP Port Assignment – 1 of 1.

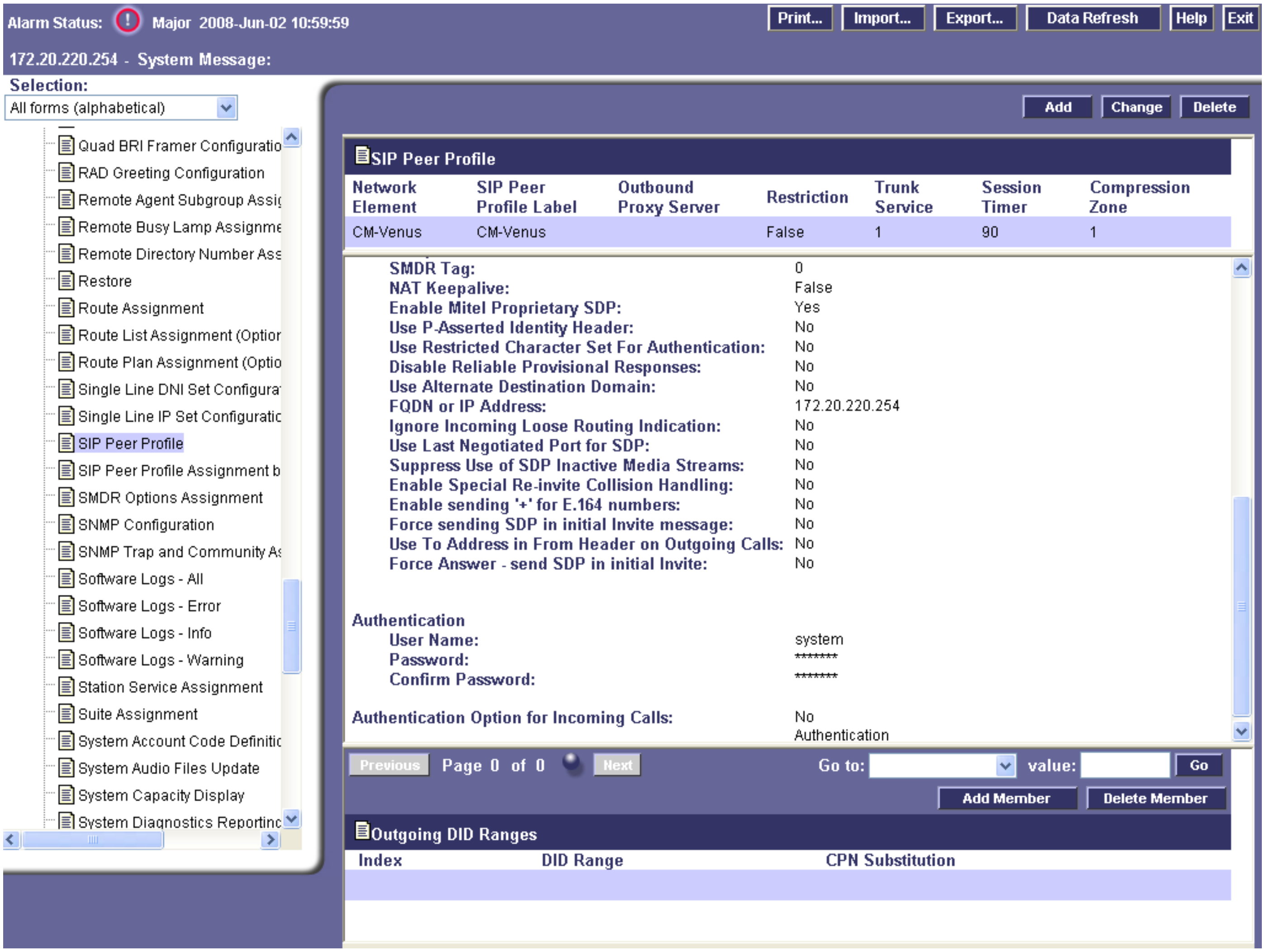


SIP Peer Profile – 1 of 2.

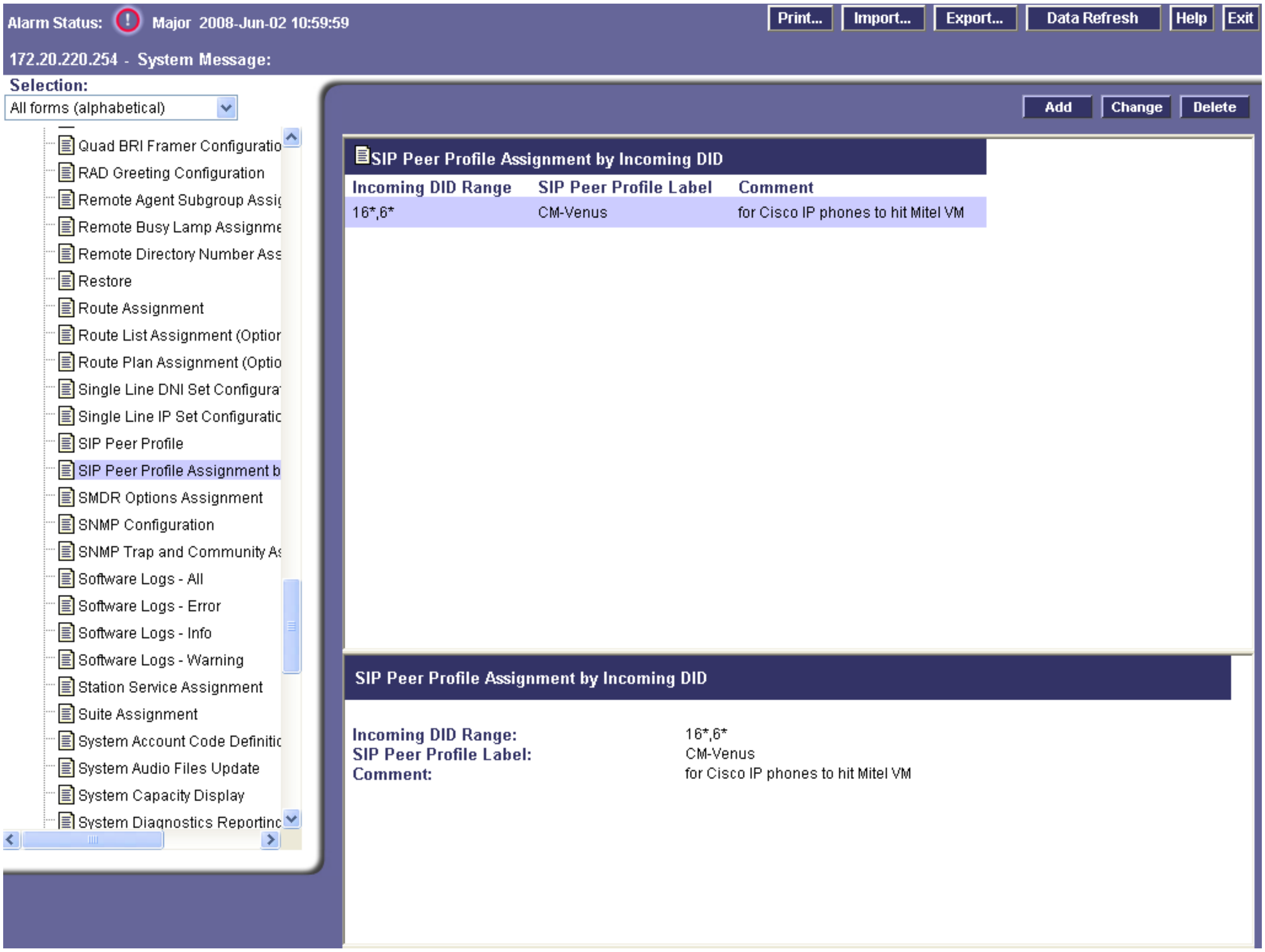
Note: The Mitel 3300 ICP SIP Session Timer value must equal or exceed the Cisco Unified Communications Manager 6.1 SIP Min-SE Value. The range for the Mitel 3300 ICP SIP Session Timer is 90 – 9999 seconds, and the default is 90 seconds. The range for the Cisco Unified Communications Manager 6.1 SIP Min-SE Value is 60 – 86400 seconds, and the default is 1800 seconds. The value shown here is 240 seconds, because that was the value on Cisco Unified Communications Manager 6.1 at the time of testing. However, if the Cisco Unified Communications Manager 6.1 SIP Min-SE Value had been set to default, the Mitel SIP Session Timer value would have been set to 1800 – 9999. See note on page 126.



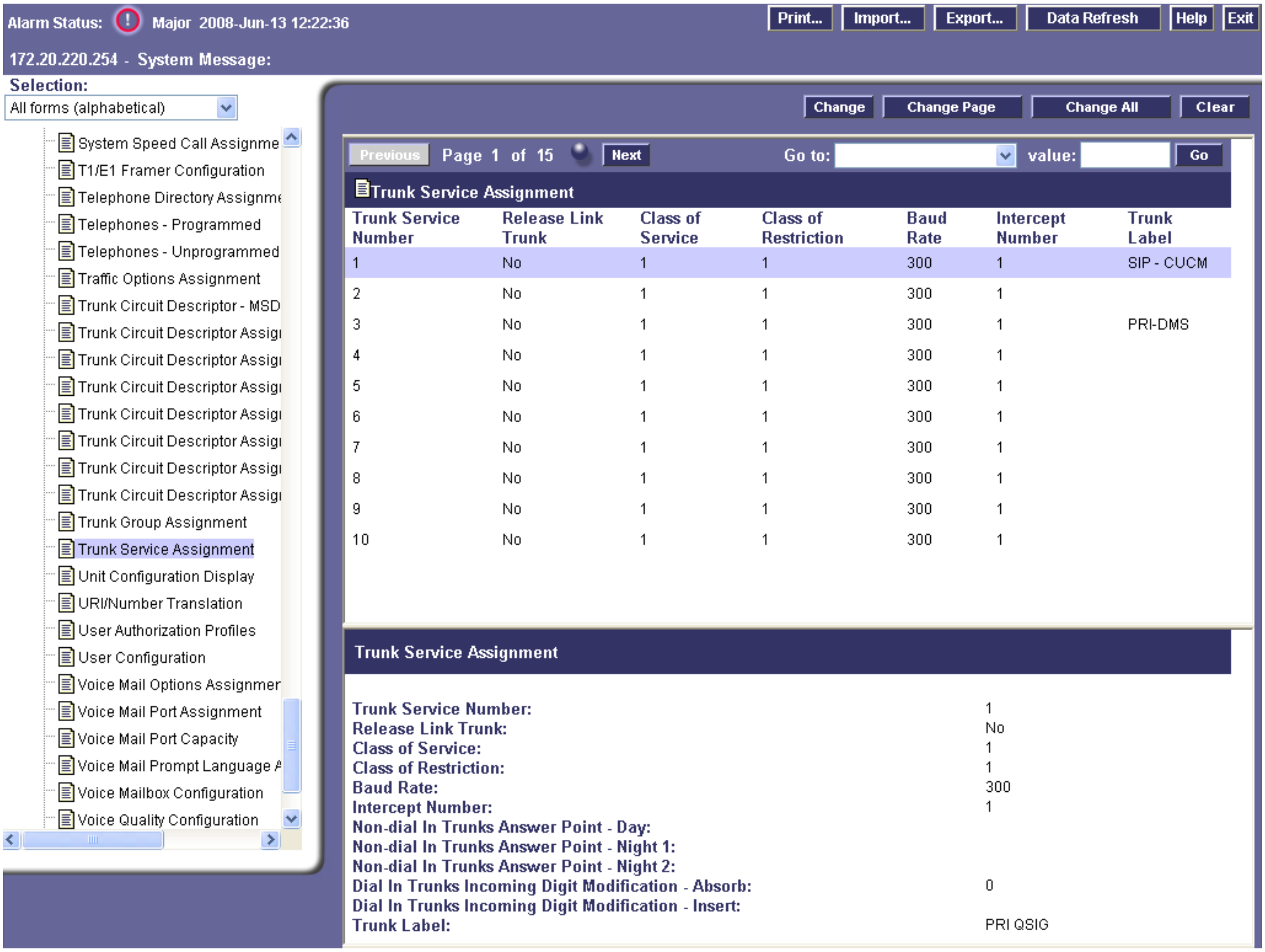
SIP Peer Profile – 2 of 2.



SIP Peer Profile Assignment by Incoming DID– 1 of 1.

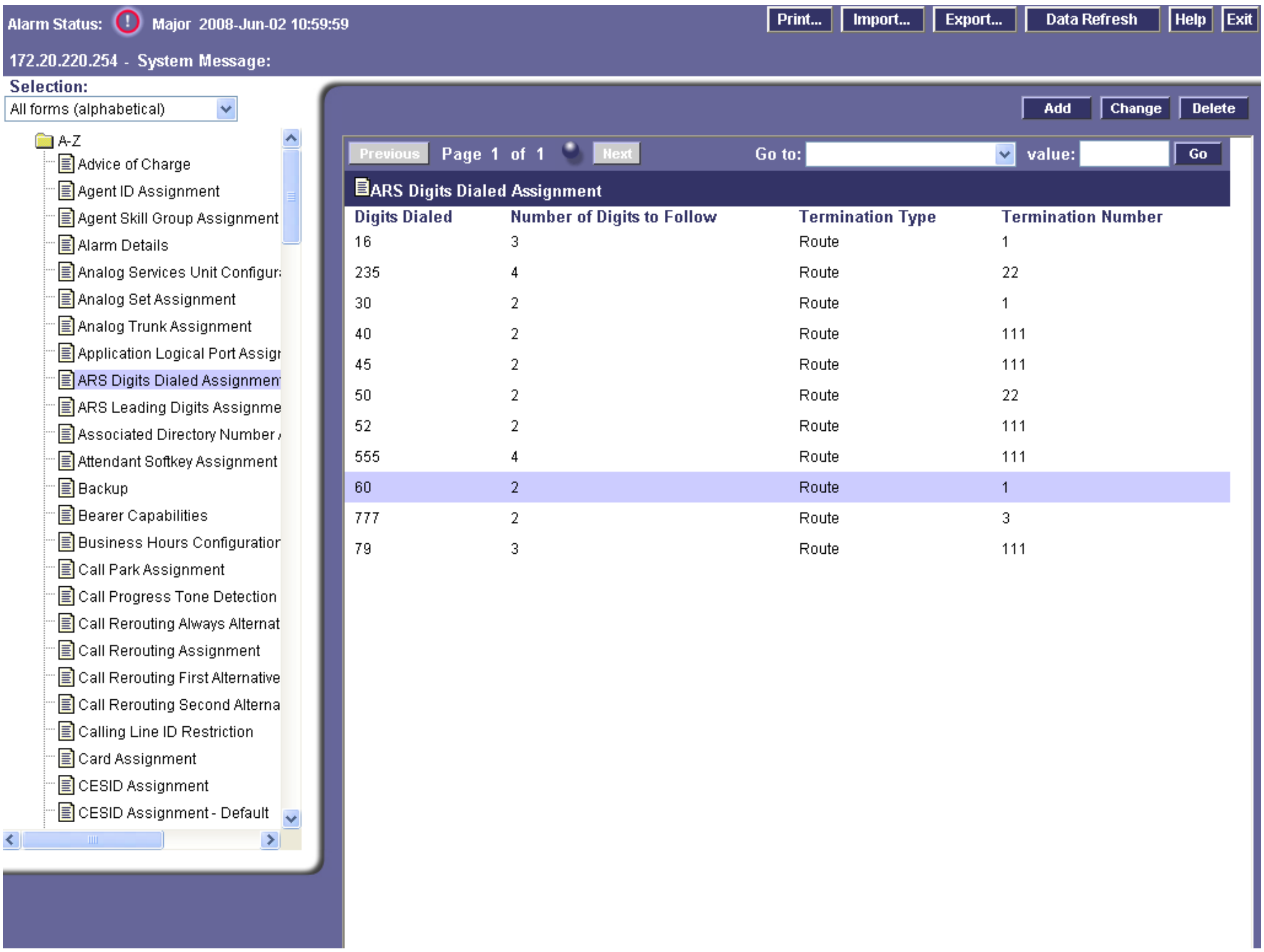


Trunk Service Assignment– 1 of 1.

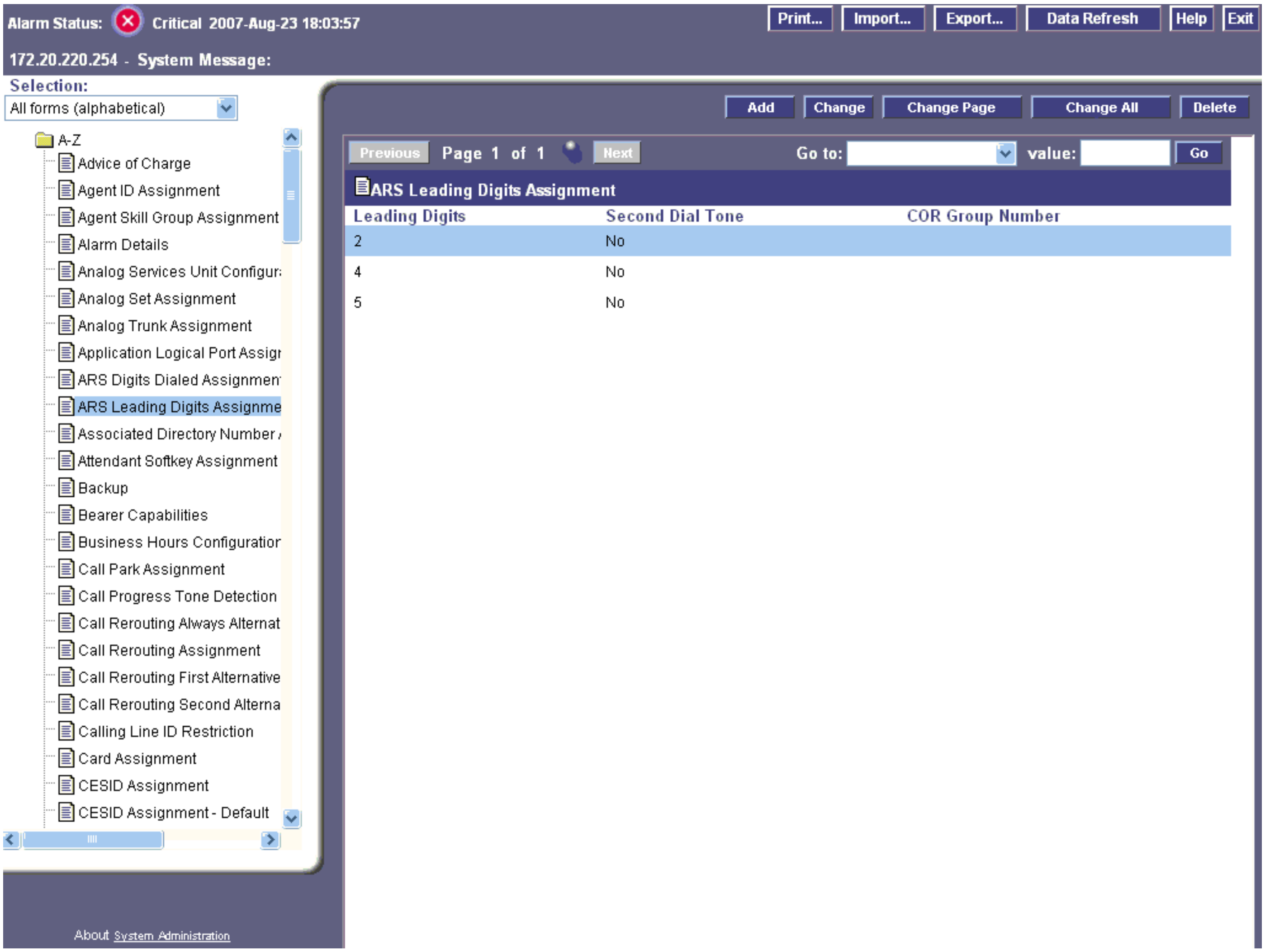


Routing

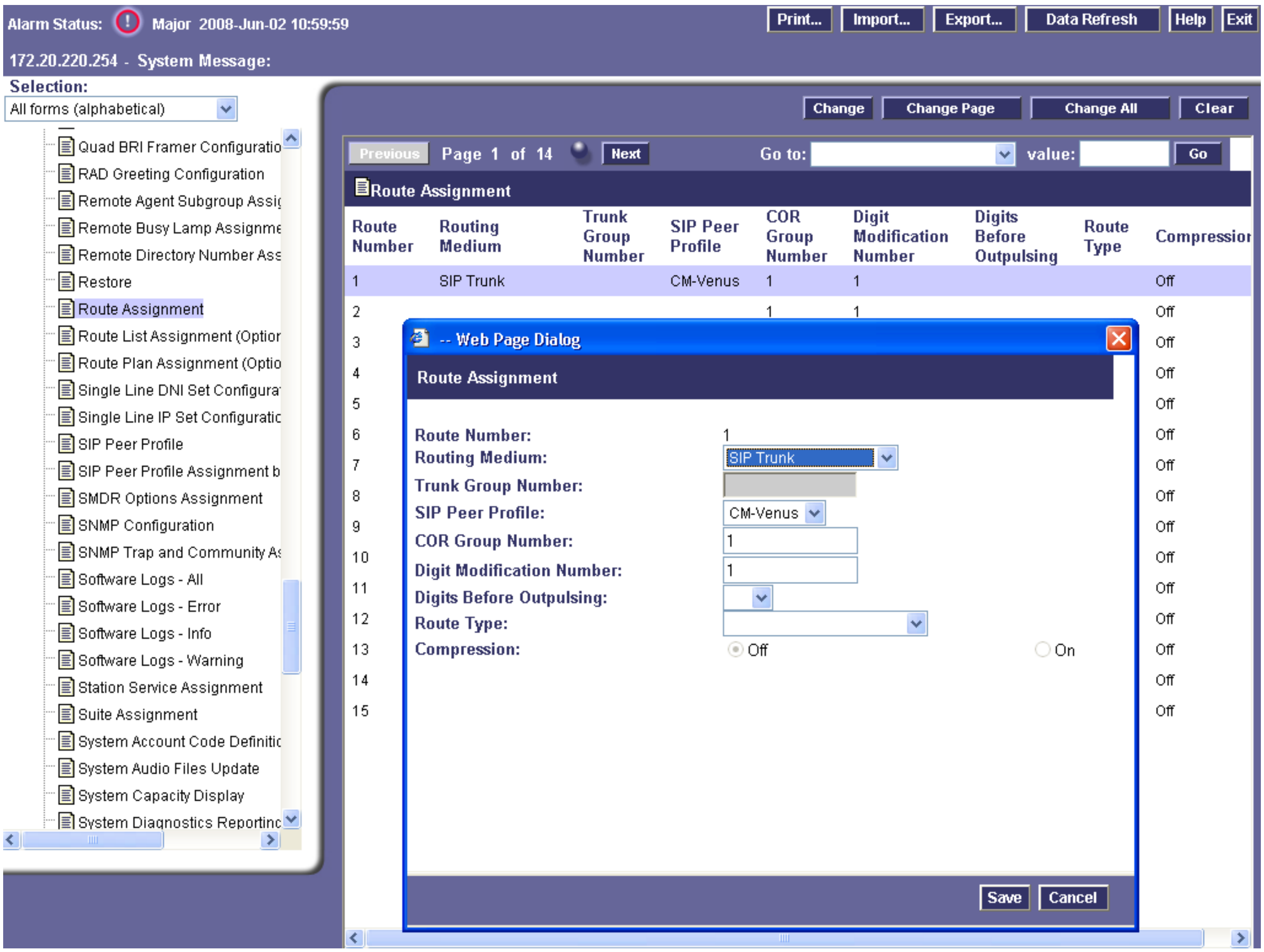
ARS Digits Dialed Assignment – 1 of 1.



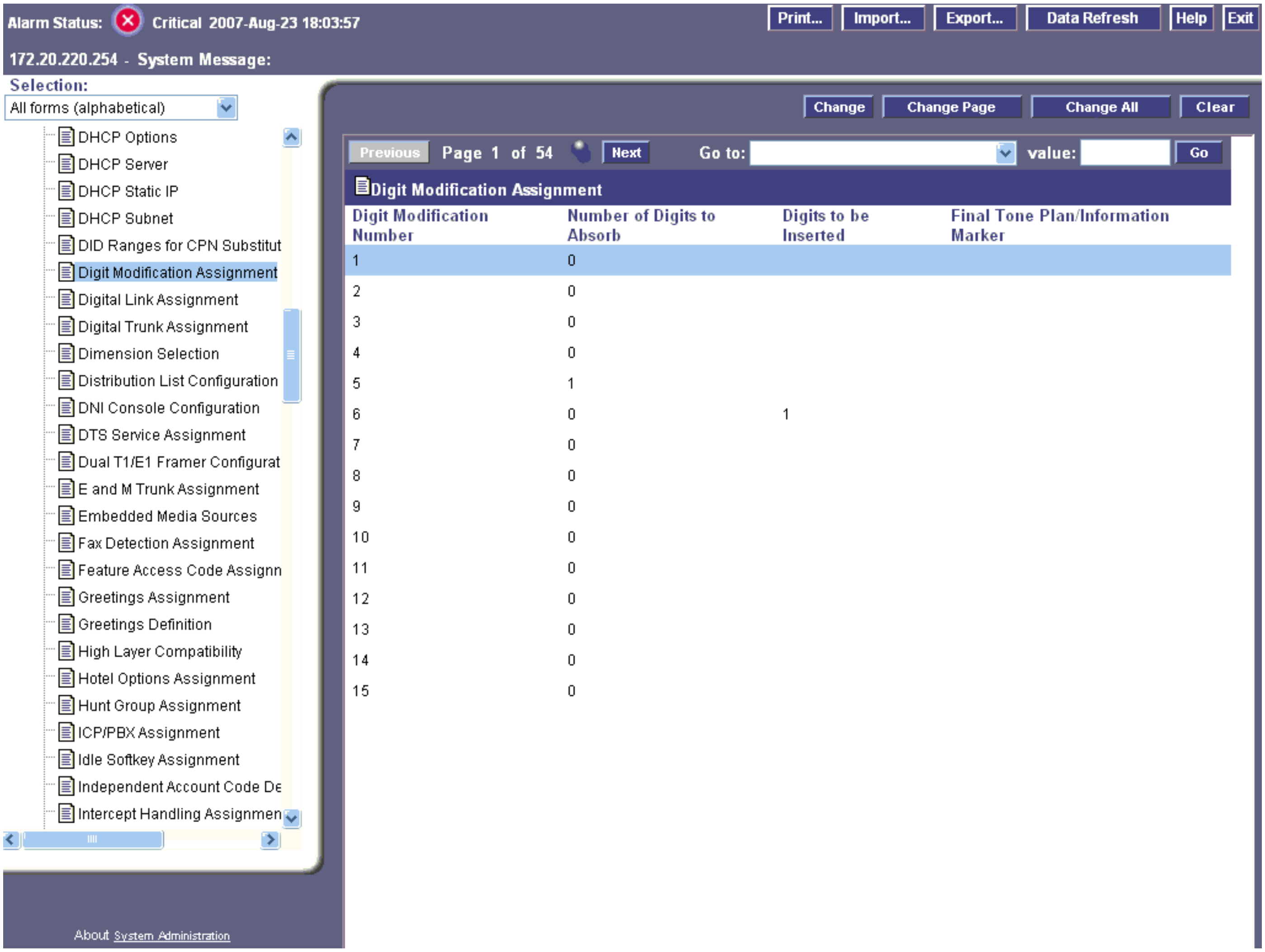
ARS Leading Digits Assignment – 1 of 1.



Route Assignment: Route 1 to Cisco Unified Communications Manager 6.1 – 1 of 1.



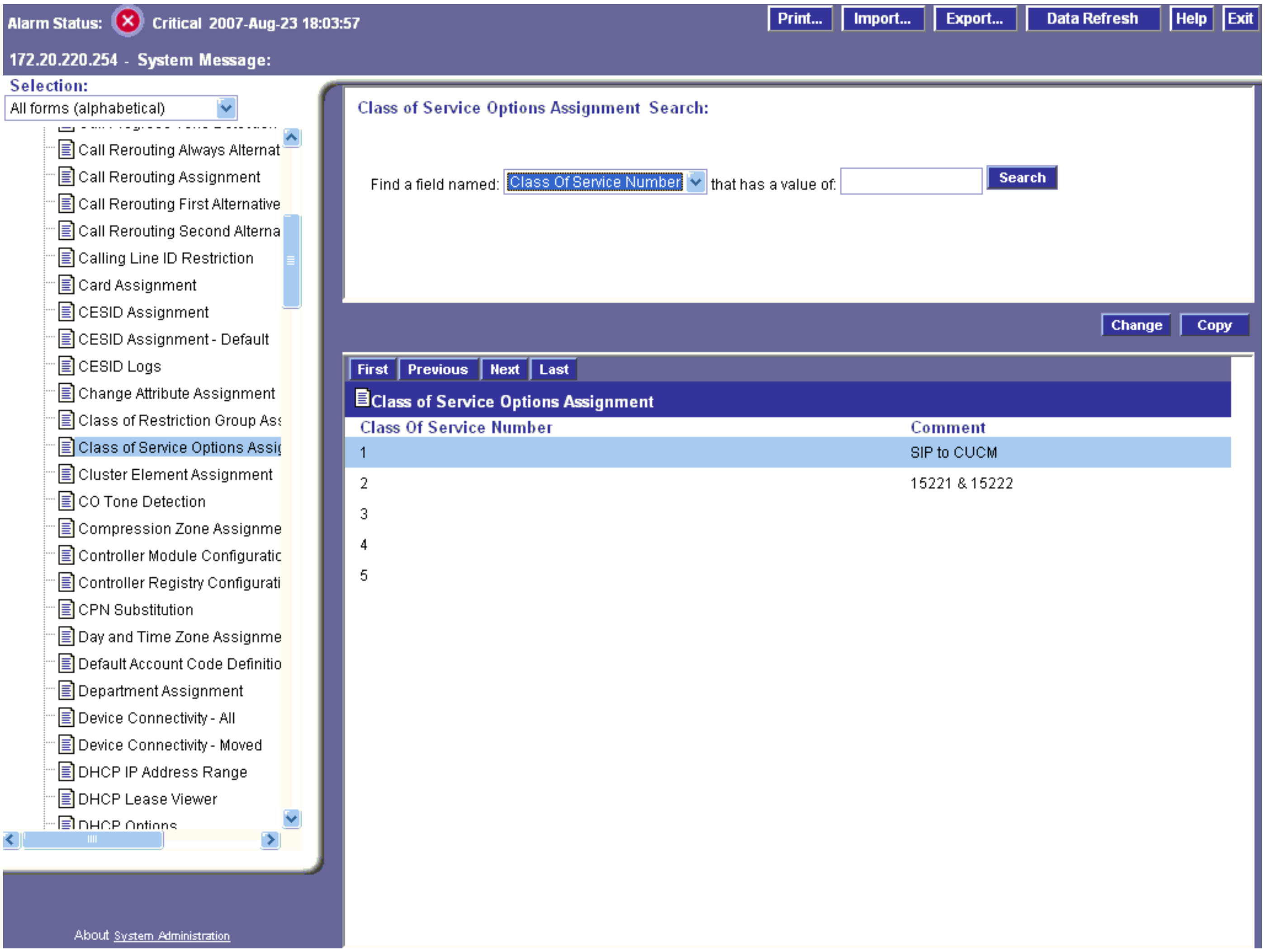
Digit Modification Assignment – 1 of 1.



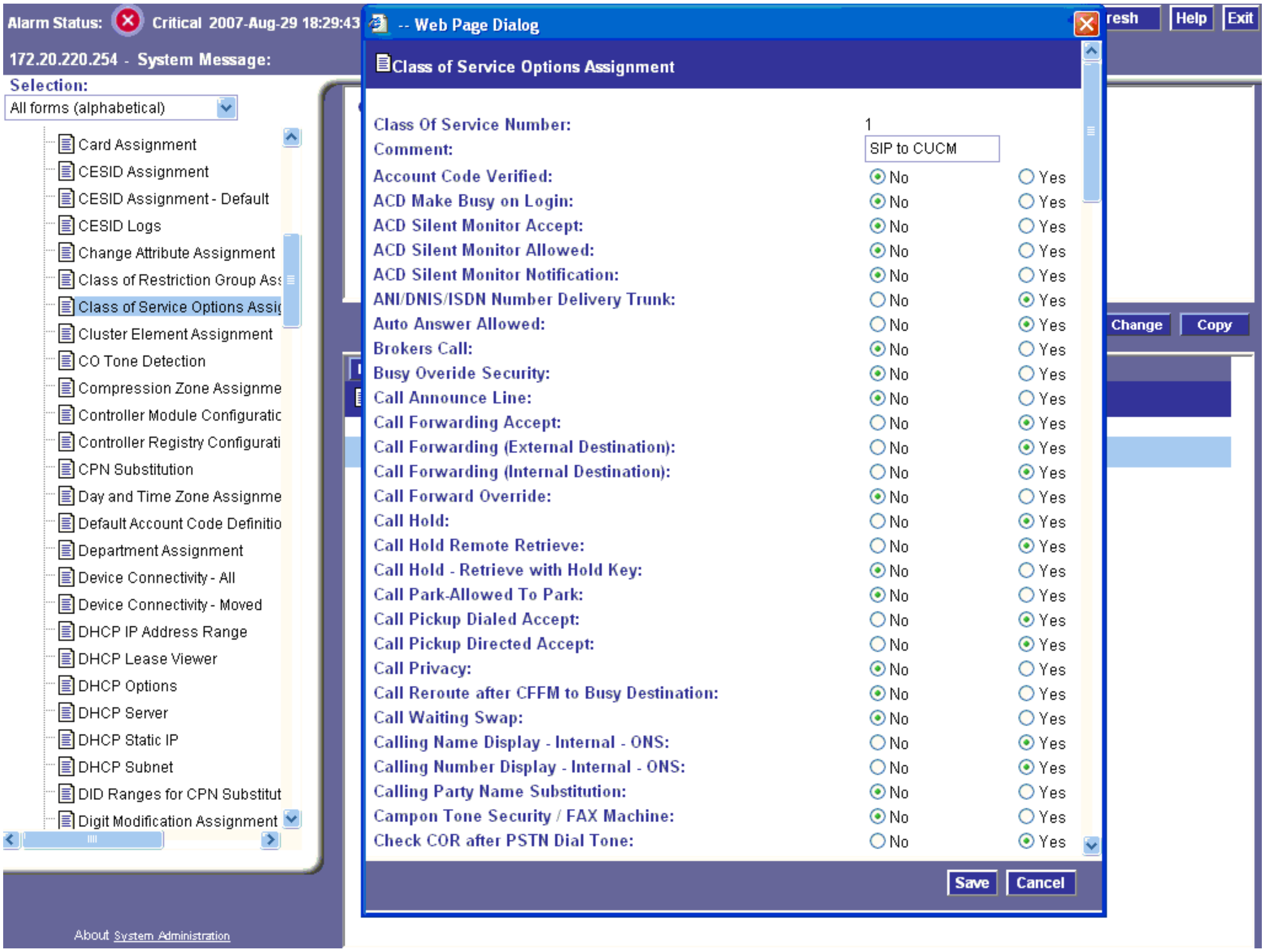
Digit Modification Number	Number of Digits to Absorb	Digits to be Inserted	Final Tone Plan/Information Marker
1	0		
2	0		
3	0		
4	0		
5	1		
6	0	1	
7	0		
8	0		
9	0		
10	0		
11	0		
12	0		
13	0		
14	0		
15	0		

Classes of Service

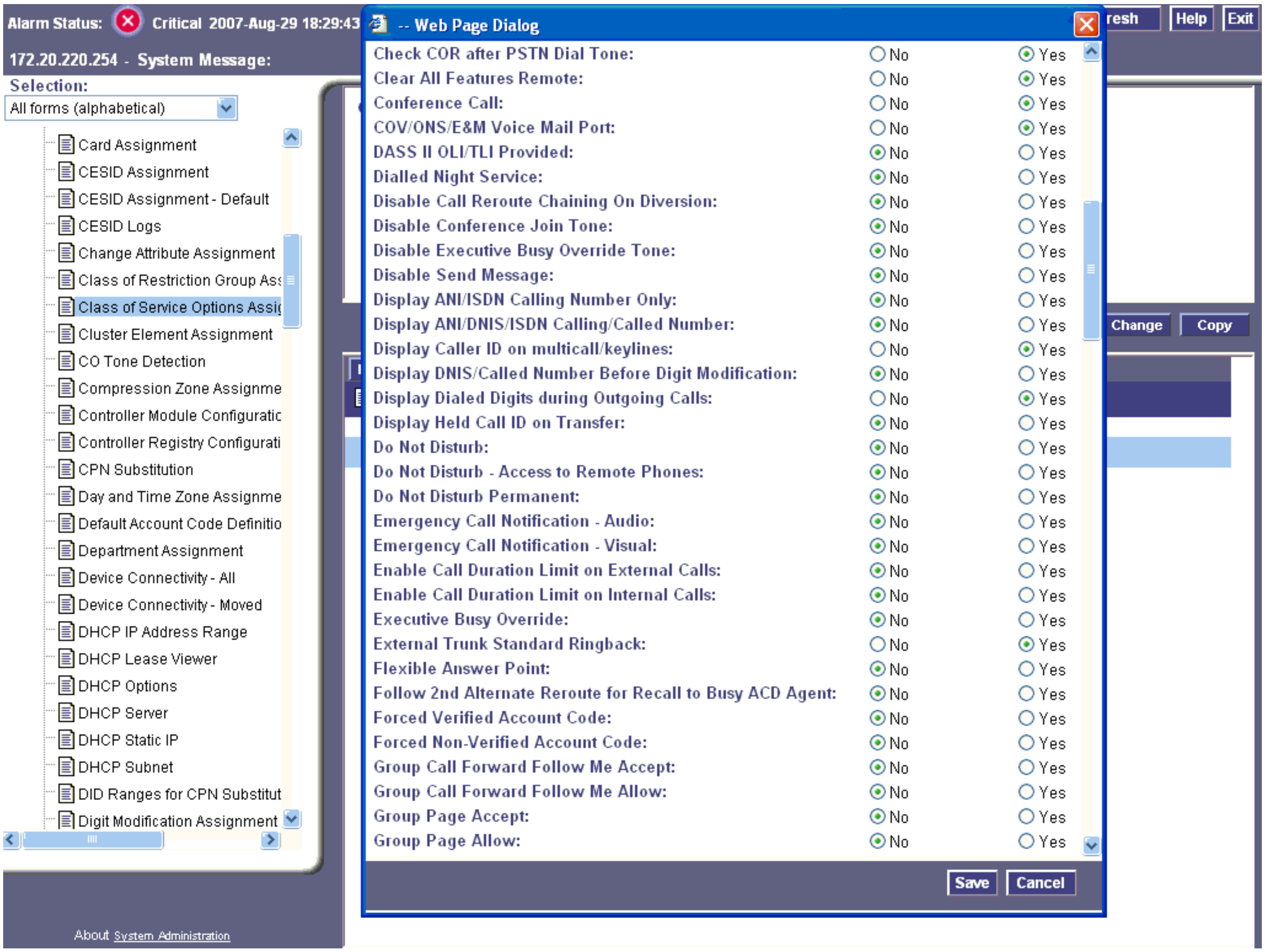
Main COS Form – 1 of 1.



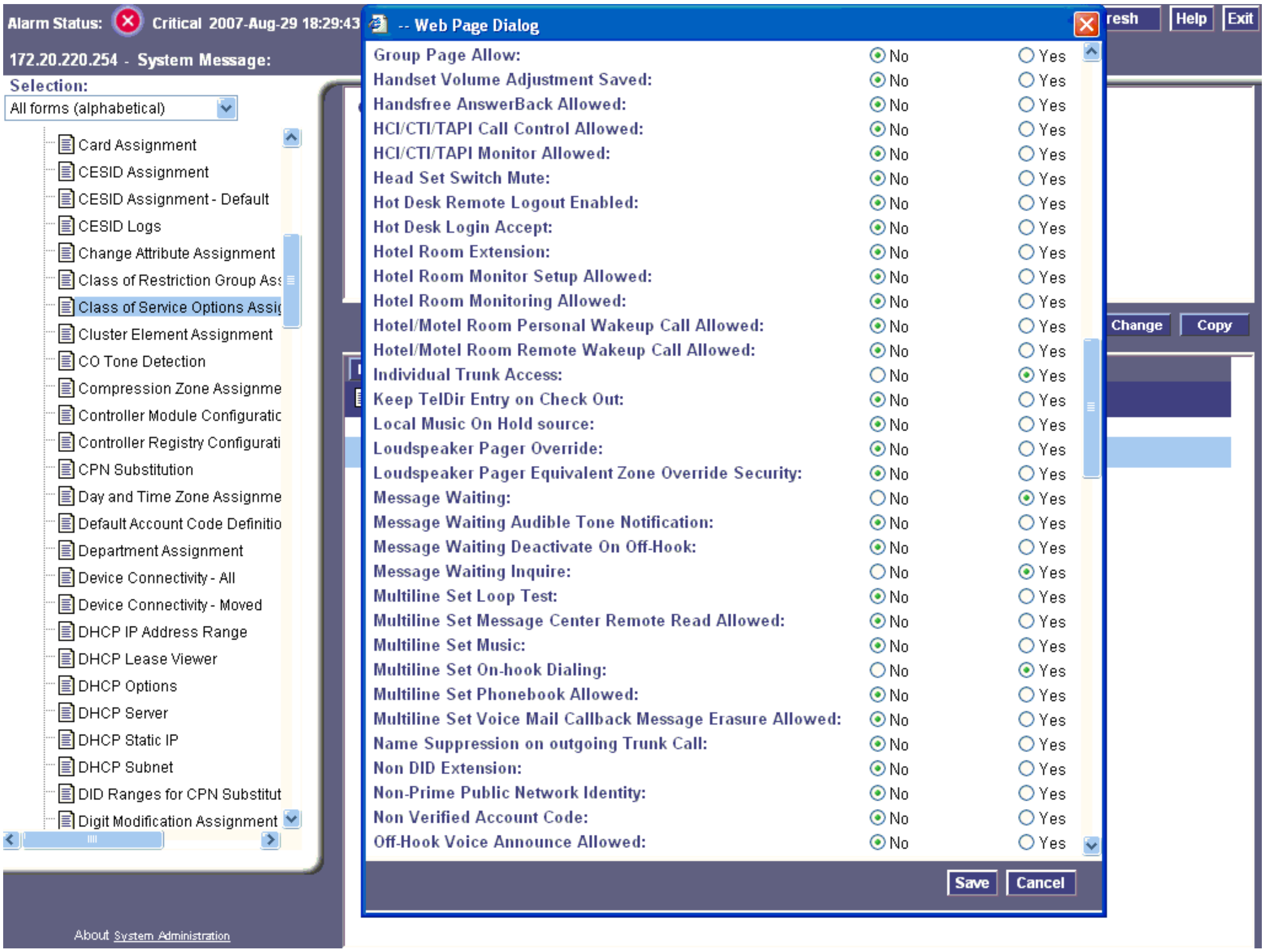
COS-Trunks – 1 of 6.



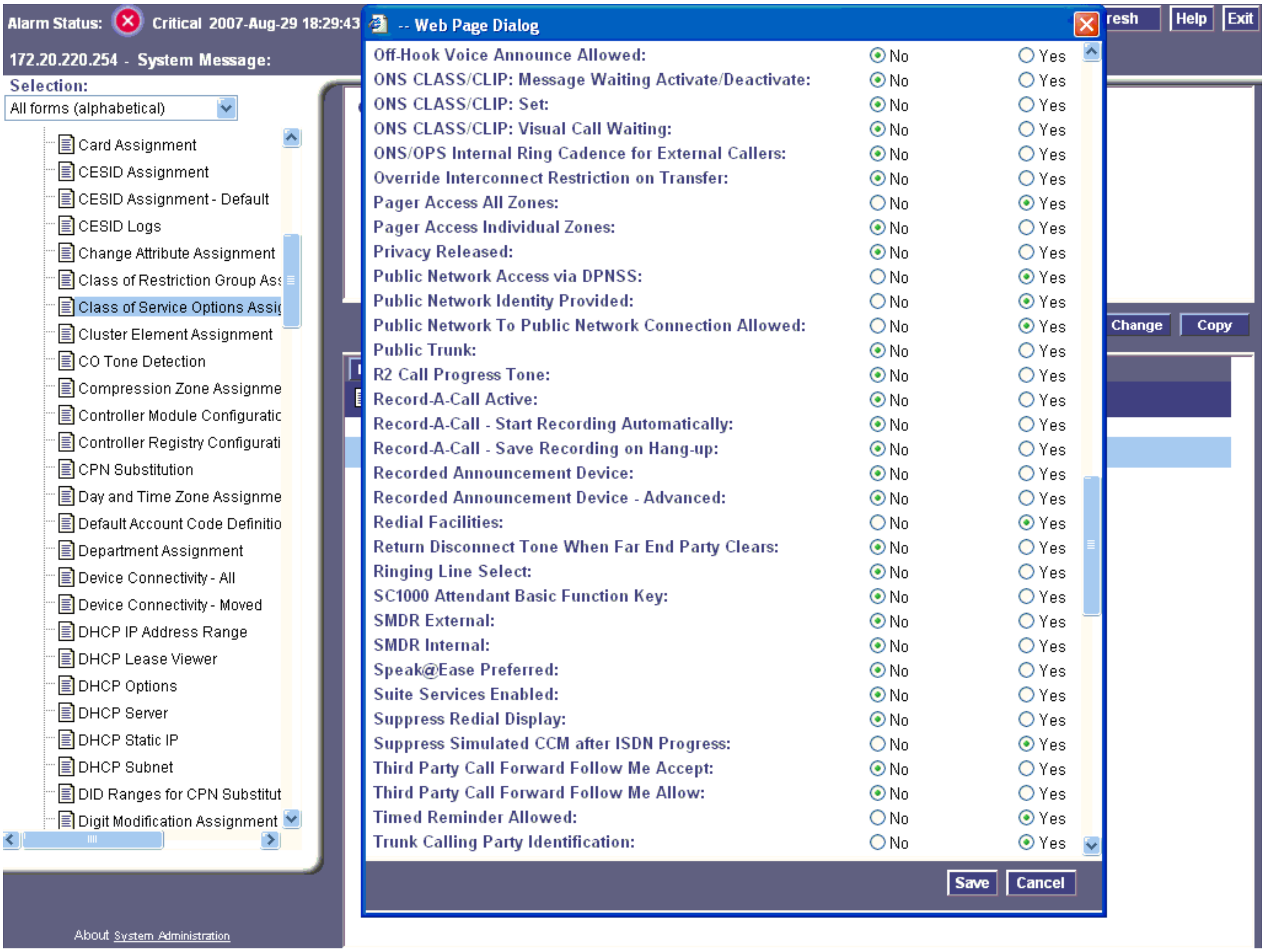
COS-Trunks – 2 of 6.



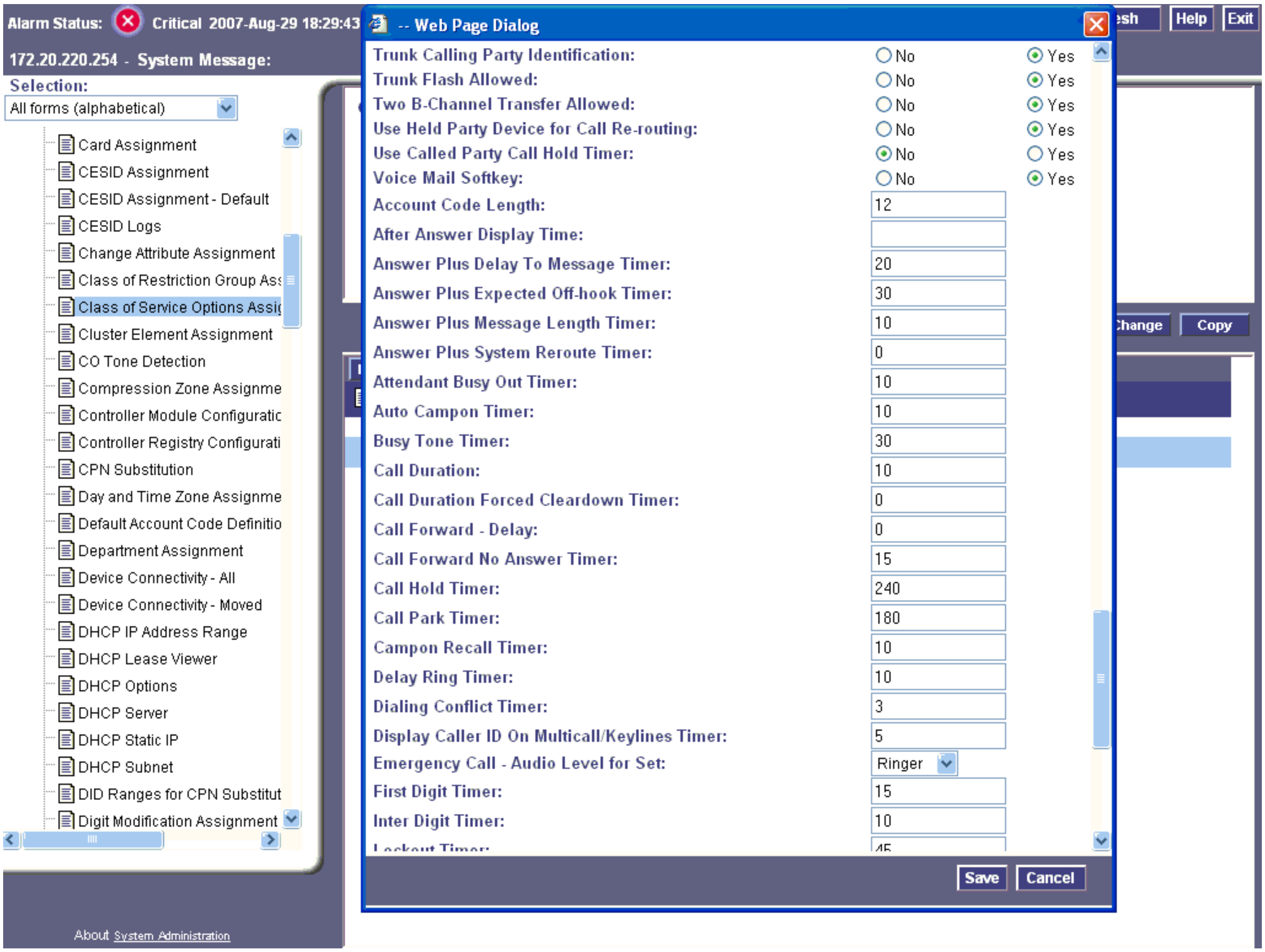
COS-Trunks – 3 of 6.



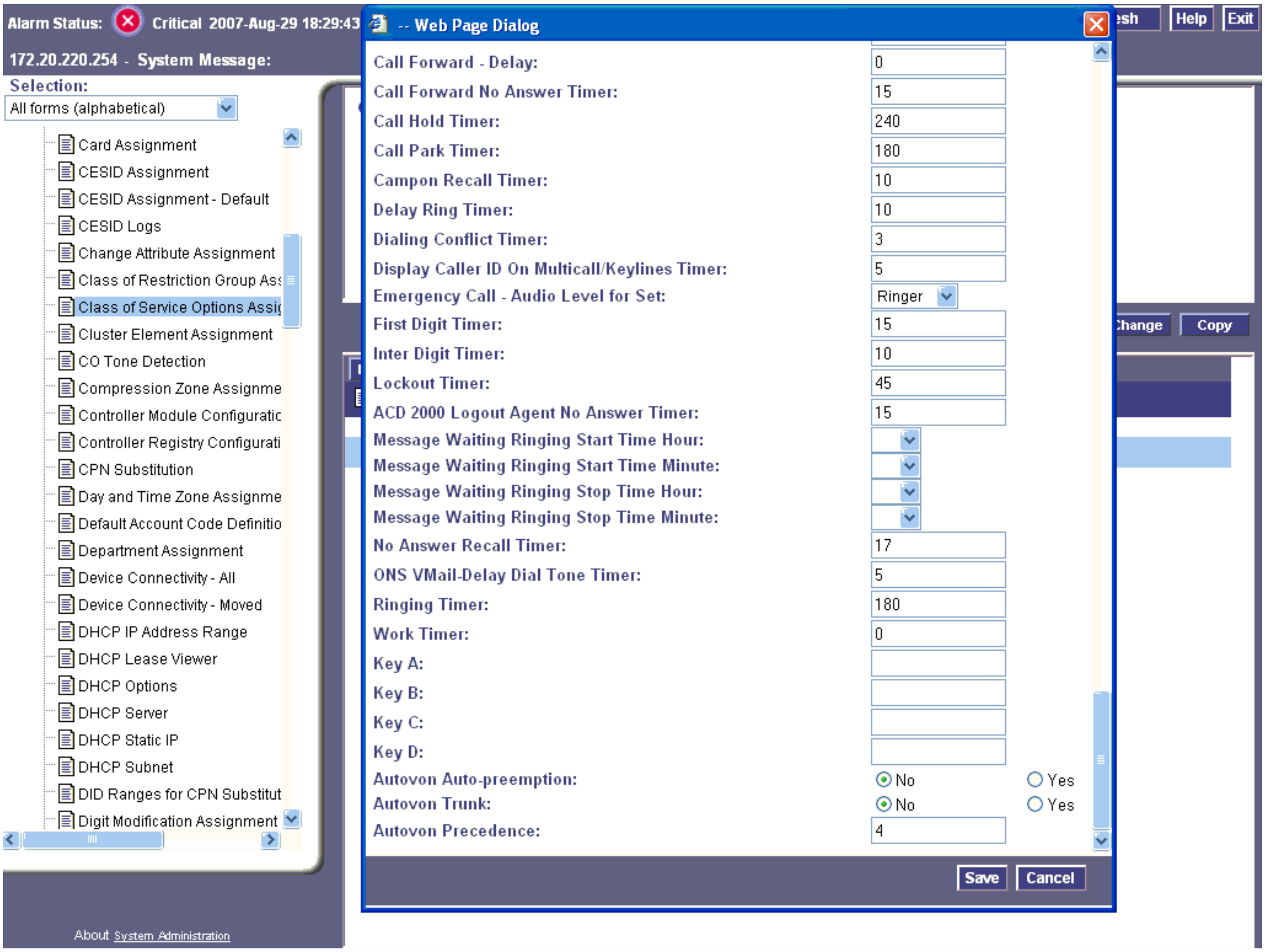
COS-Trunks – 4 of 6.



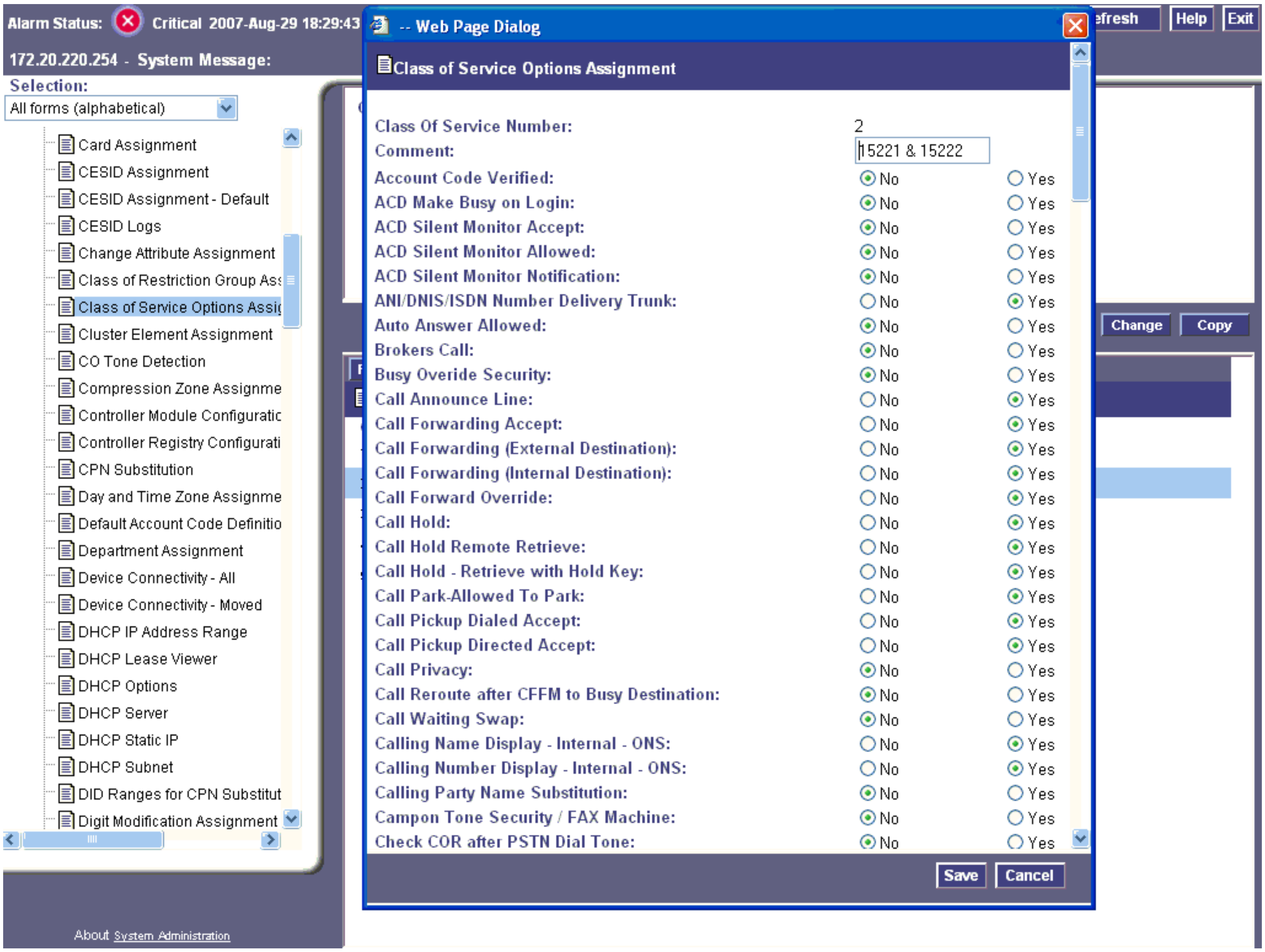
COS-Trunks – 5 of 6.



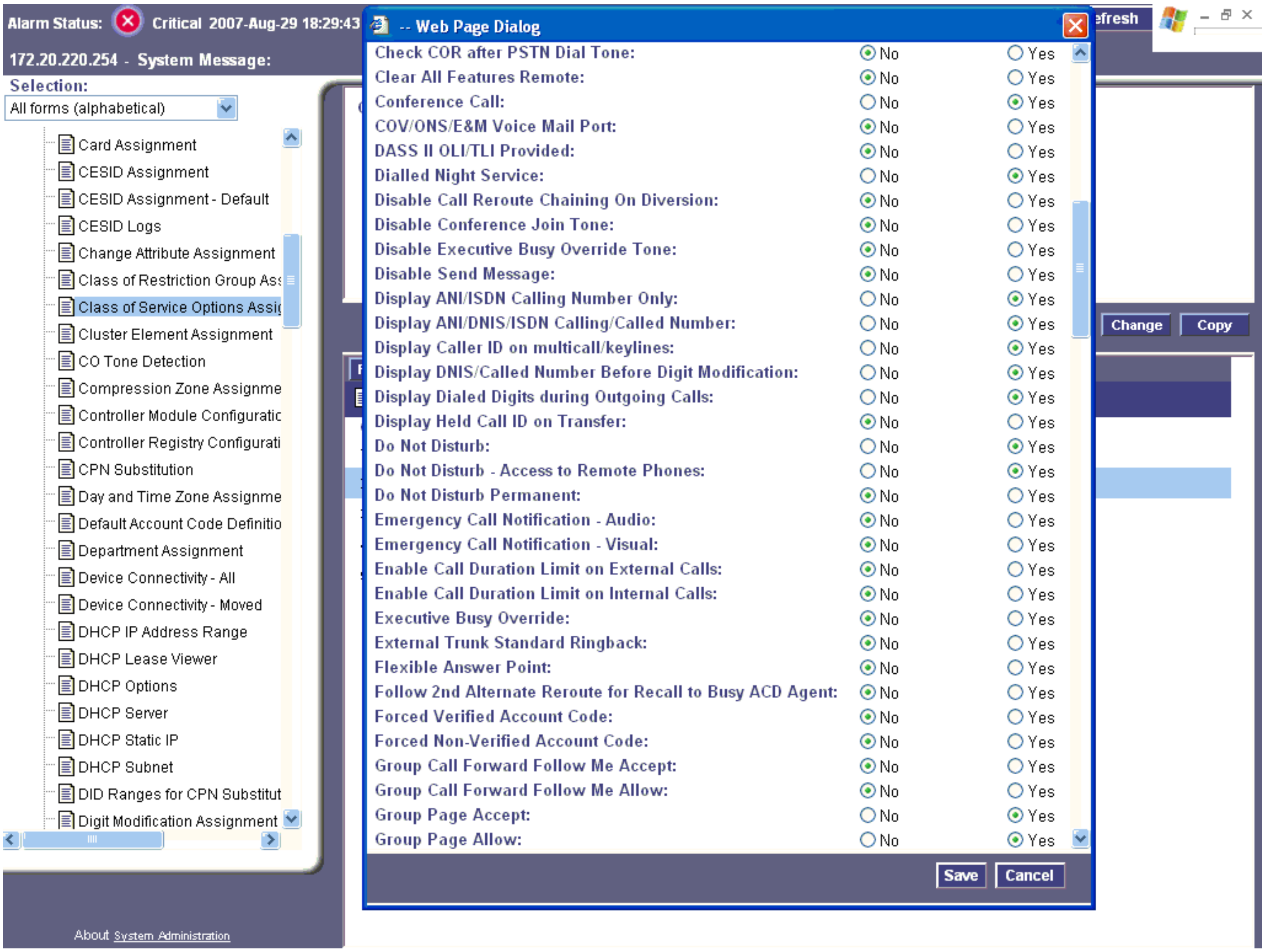
COS-Trunks – 6 of 6.



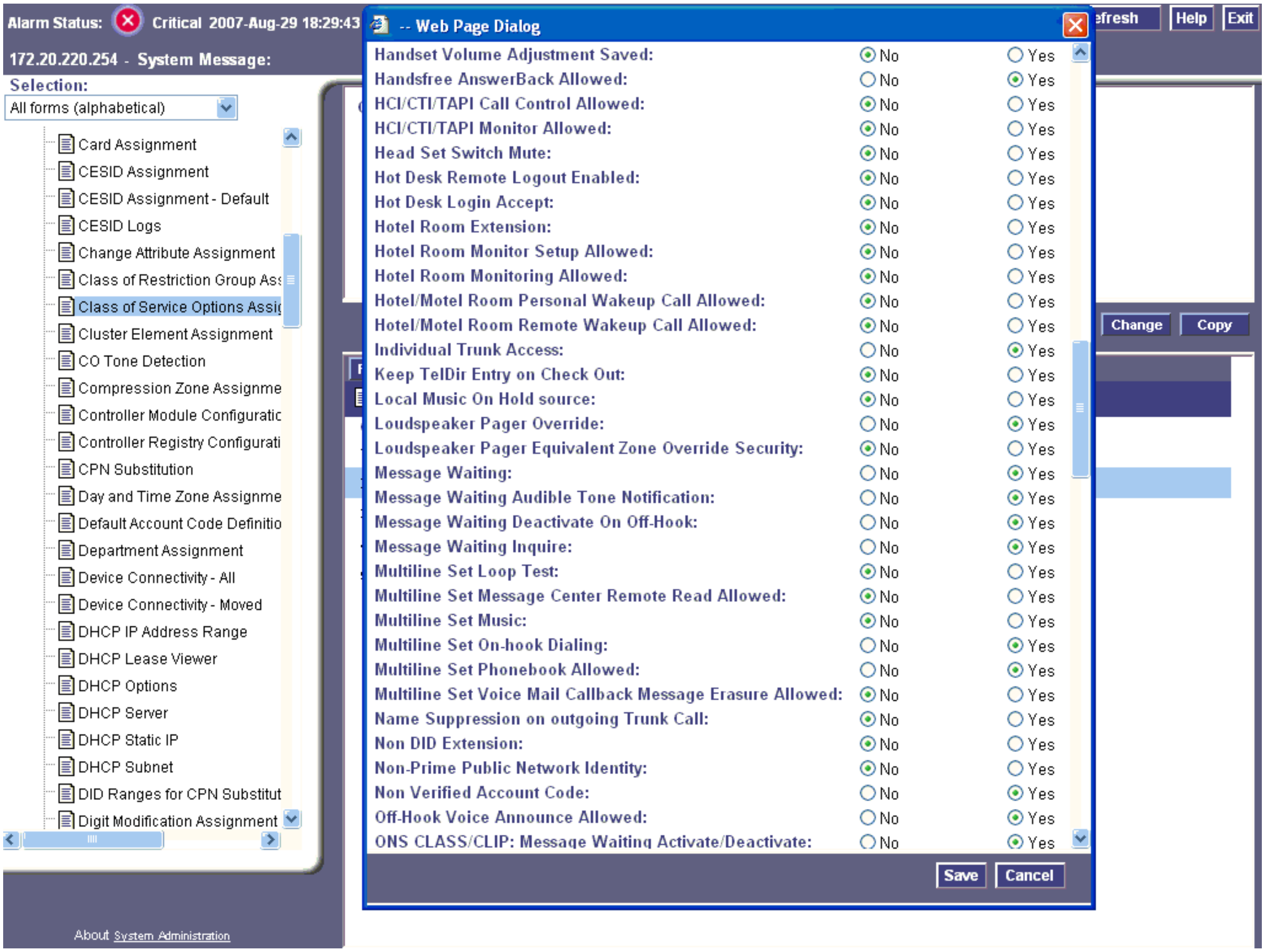
COS-Phones – 1 of 6.



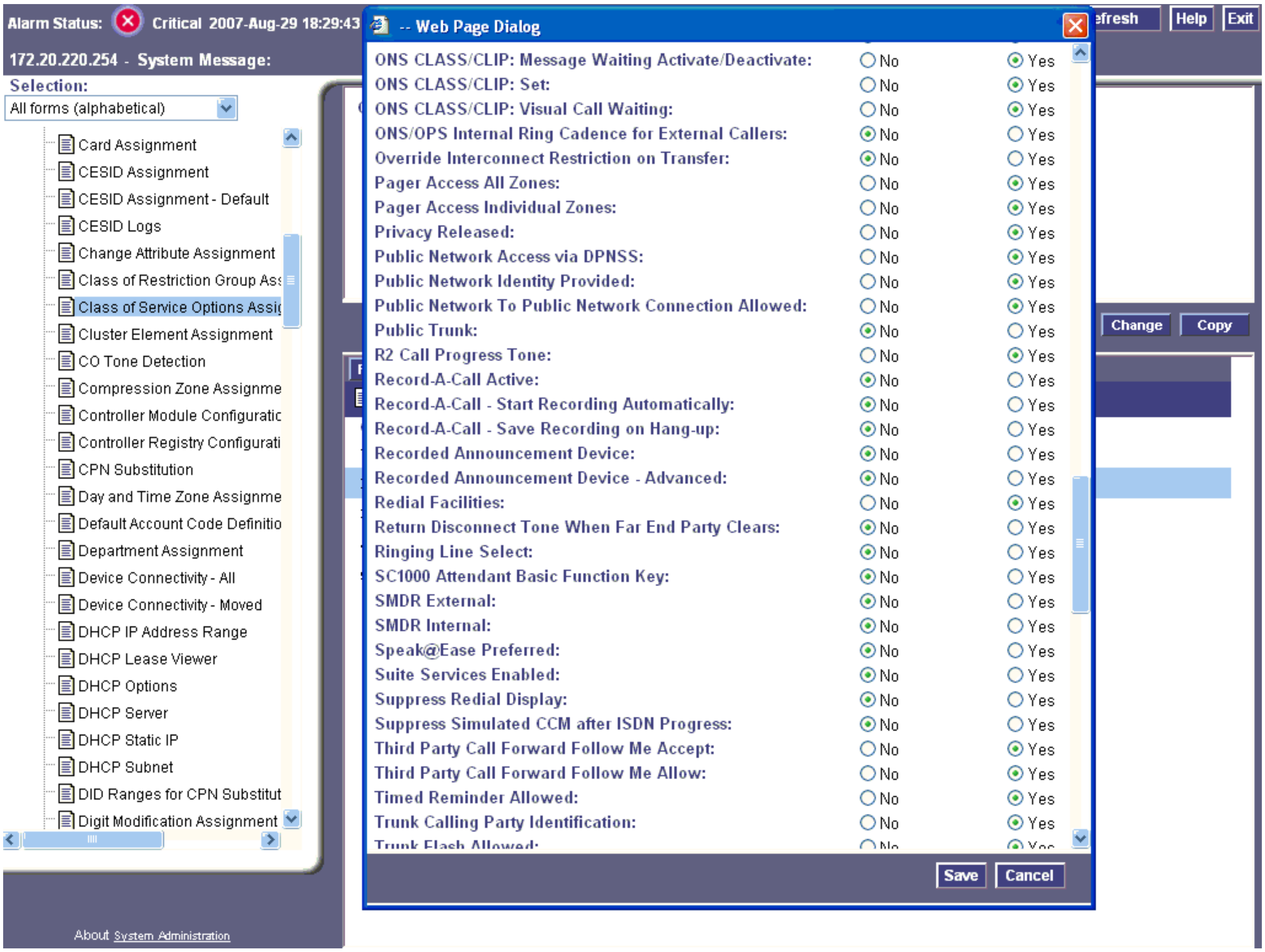
COS-Phones – 2 of 6.



COS-Phones – 3 of 6.



COS-Phones – 4 of 6.



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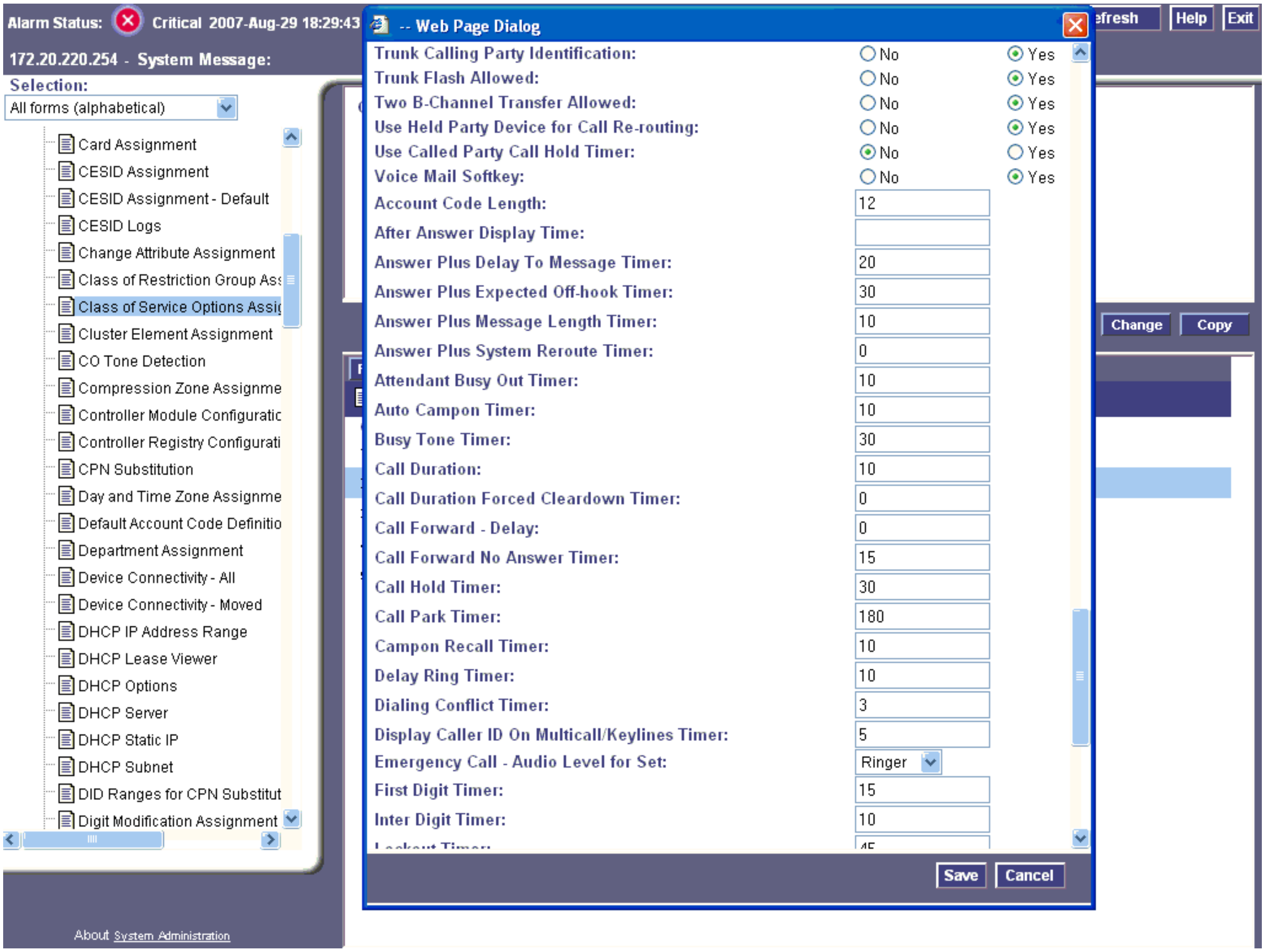

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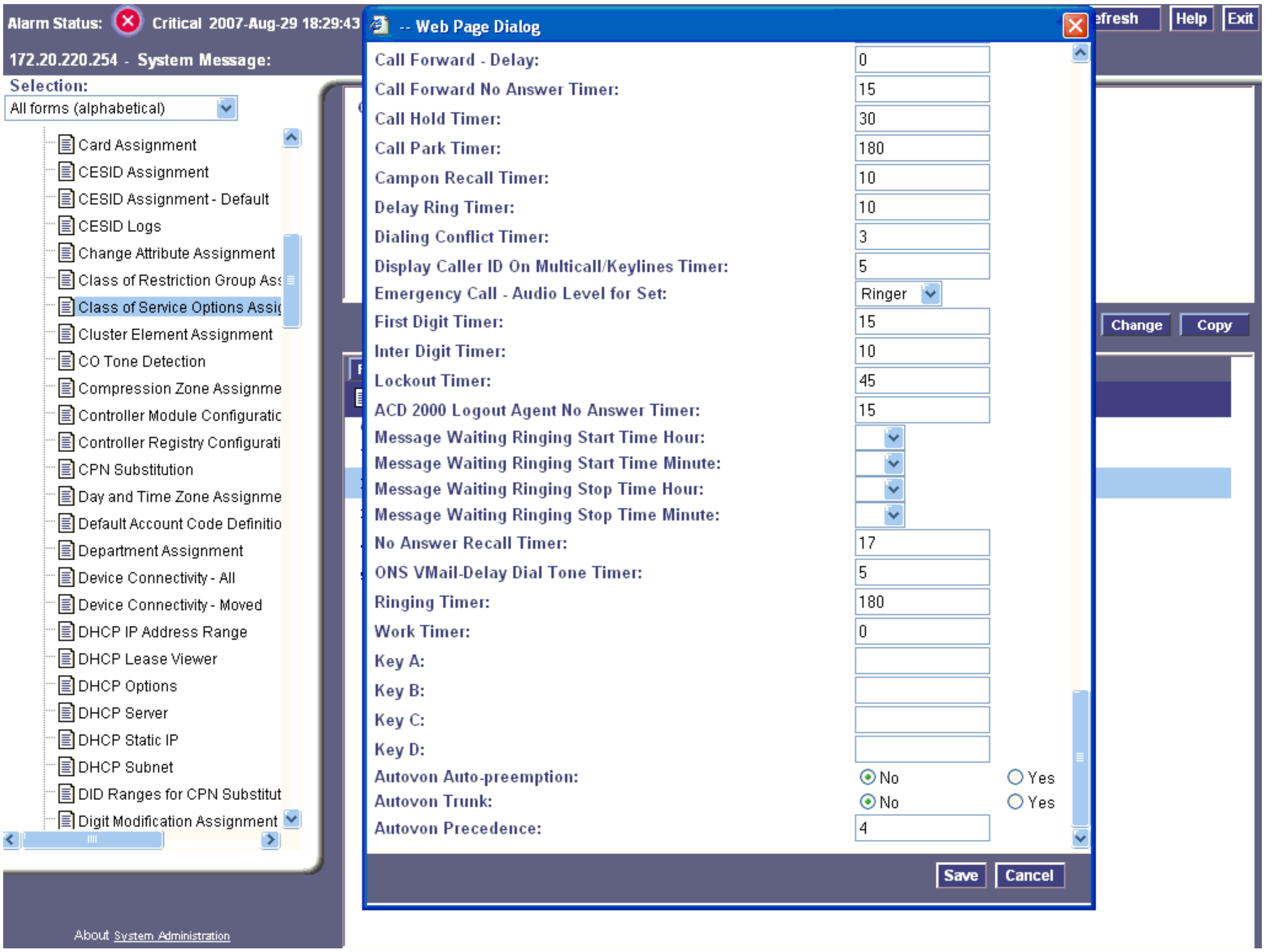
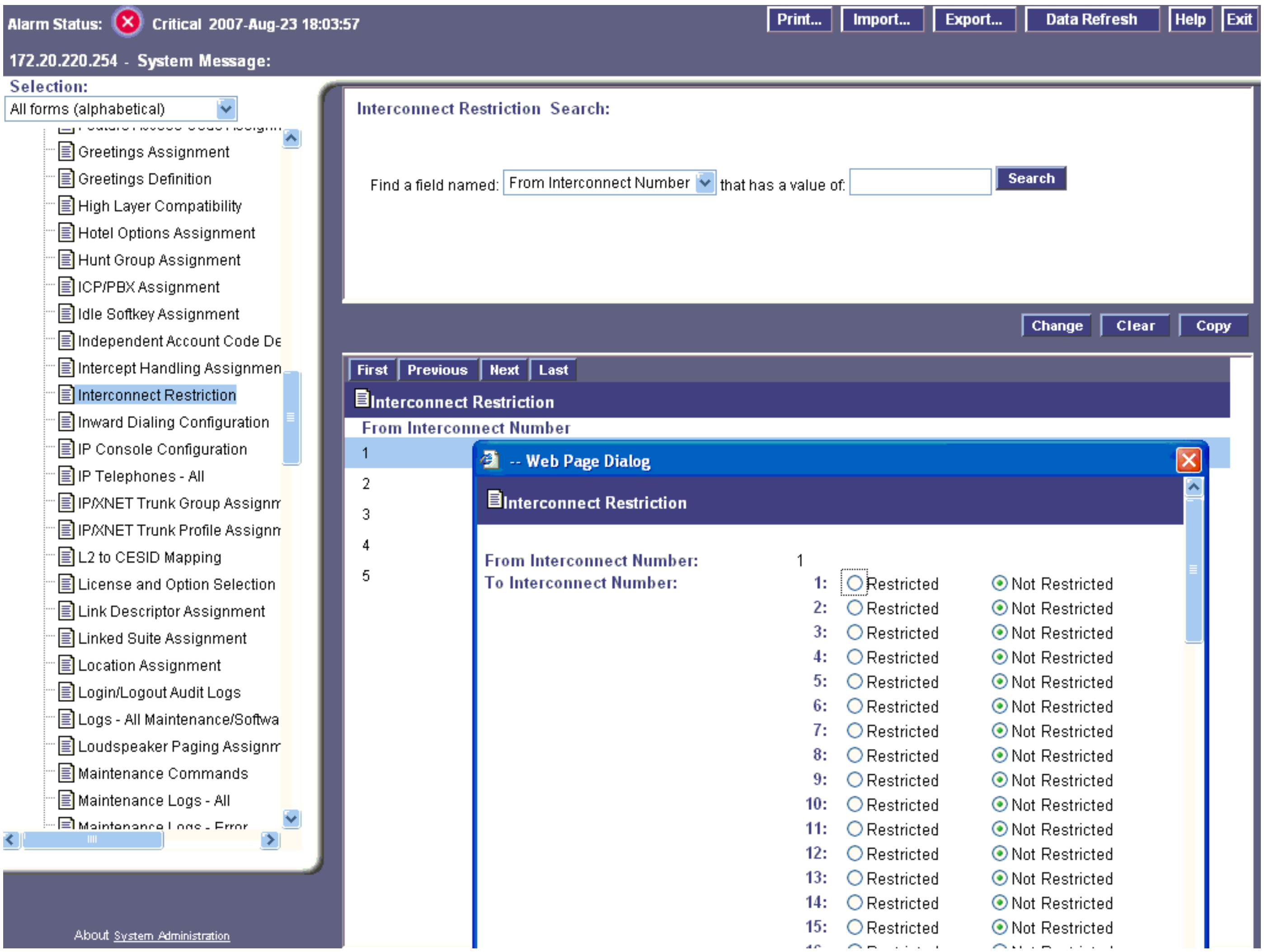


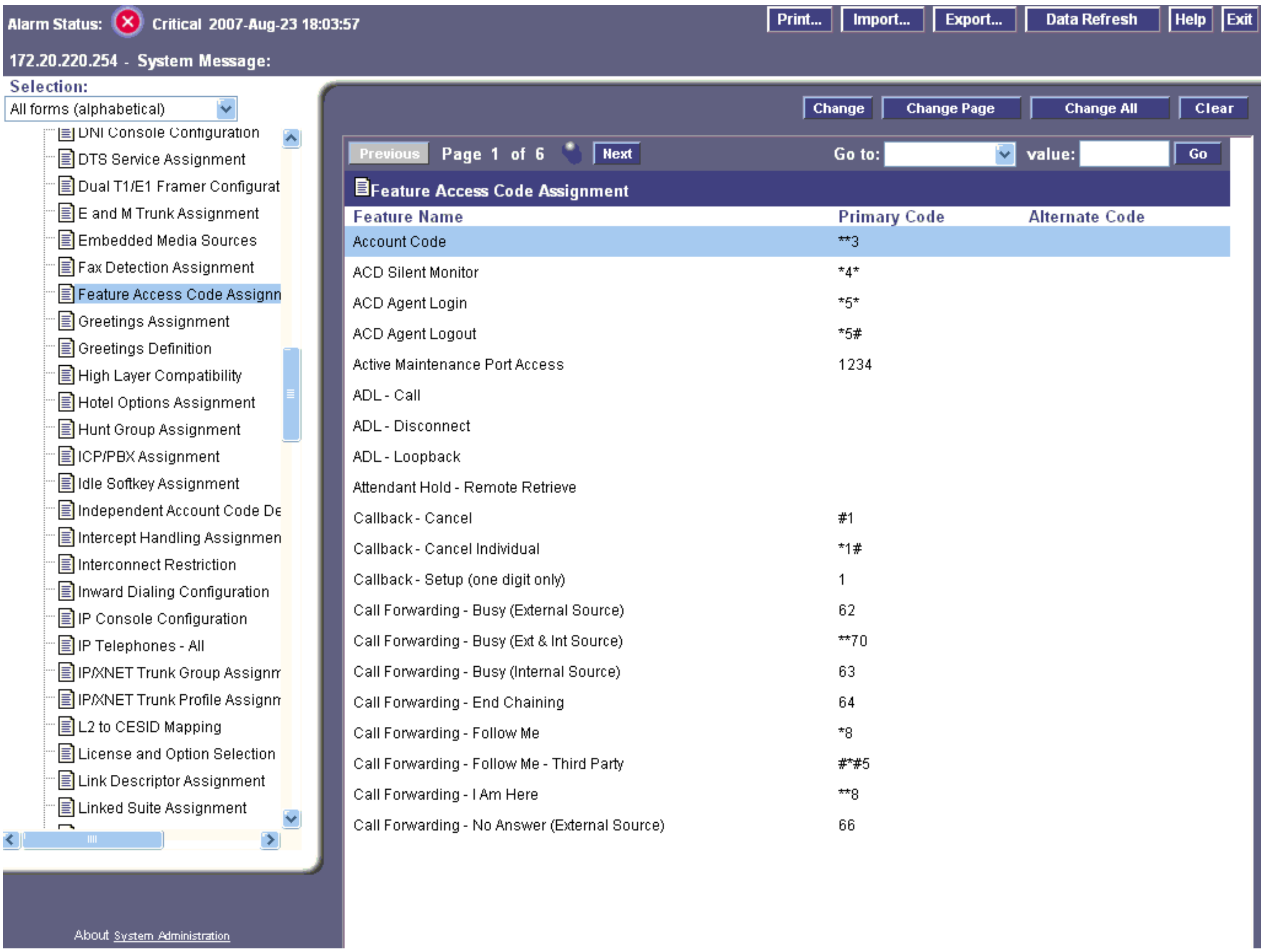
Interconnect Restriction

Interconnect Restriction – 1 of 1.

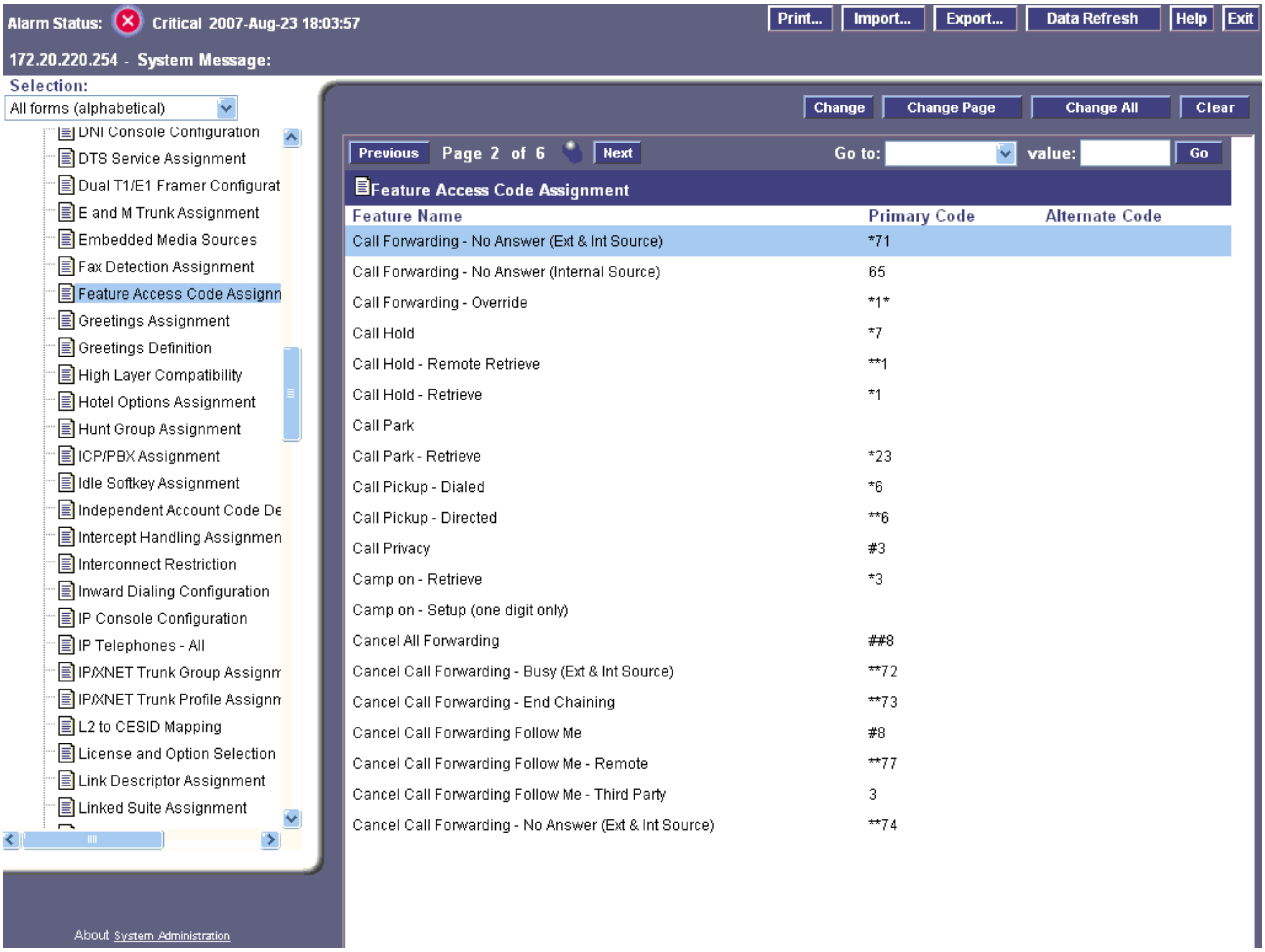


Feature Access Codes

Feature Access Codes – 1 of 6.



Feature Access Codes – 2 of 6.



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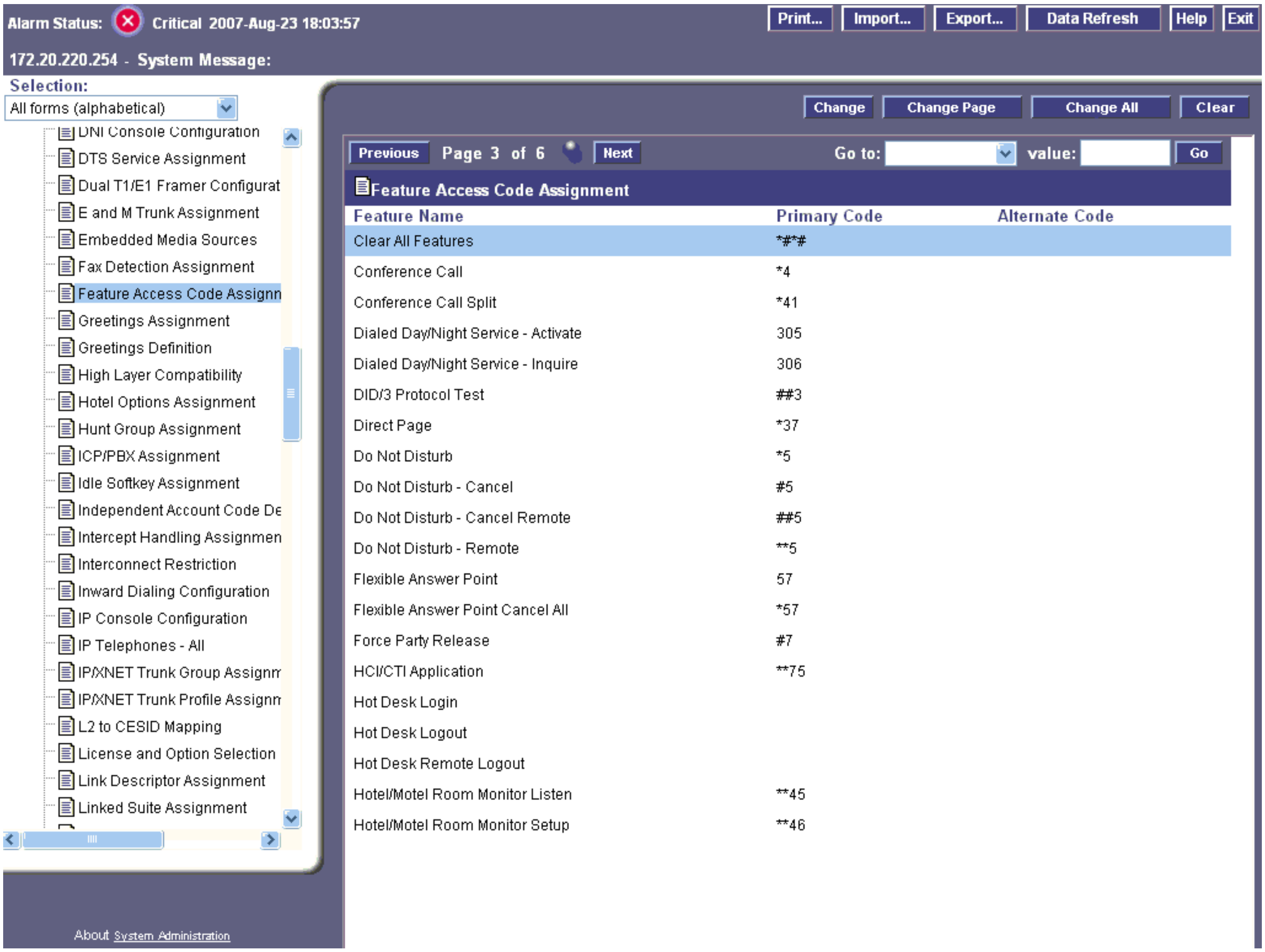

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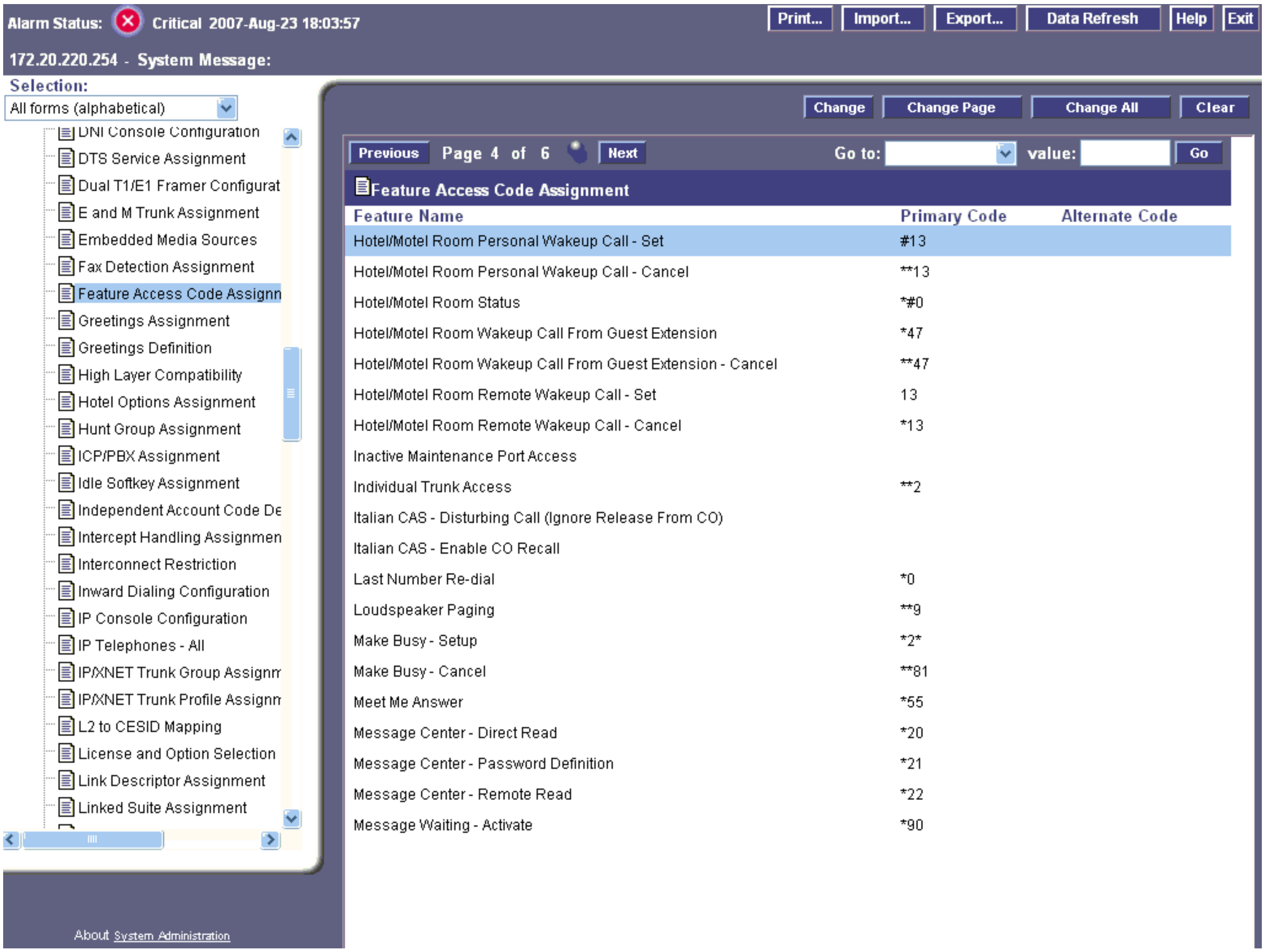

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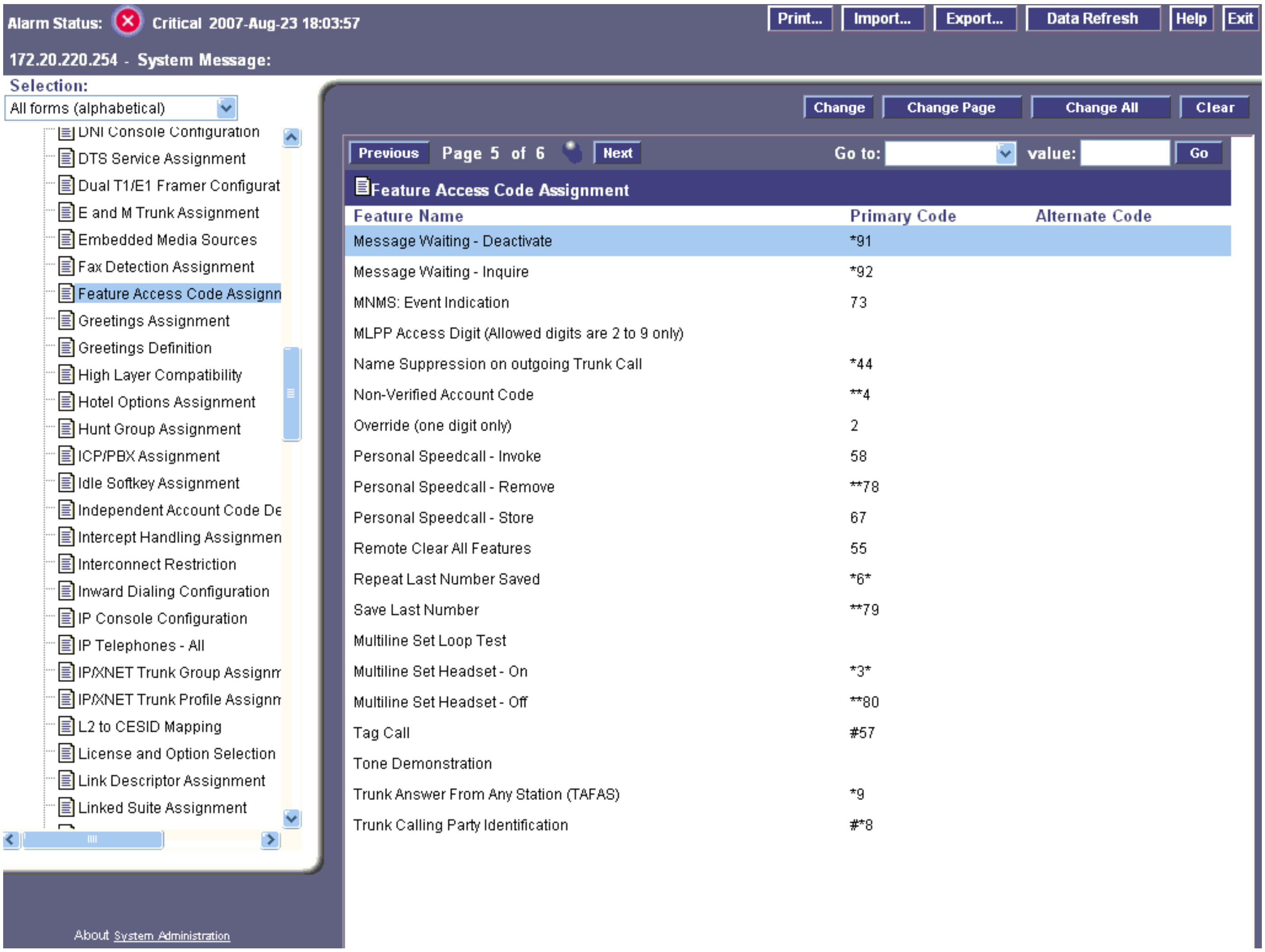
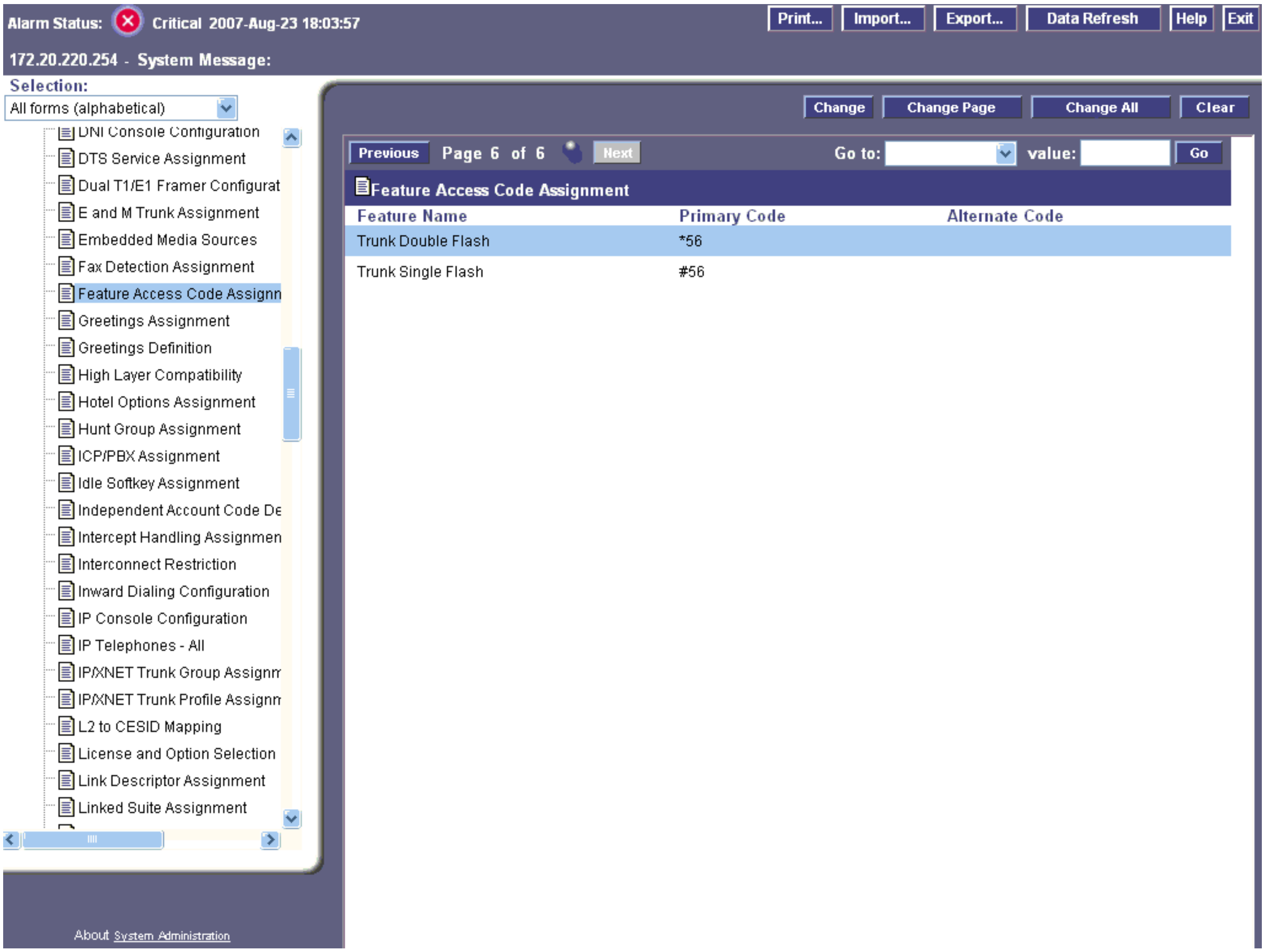
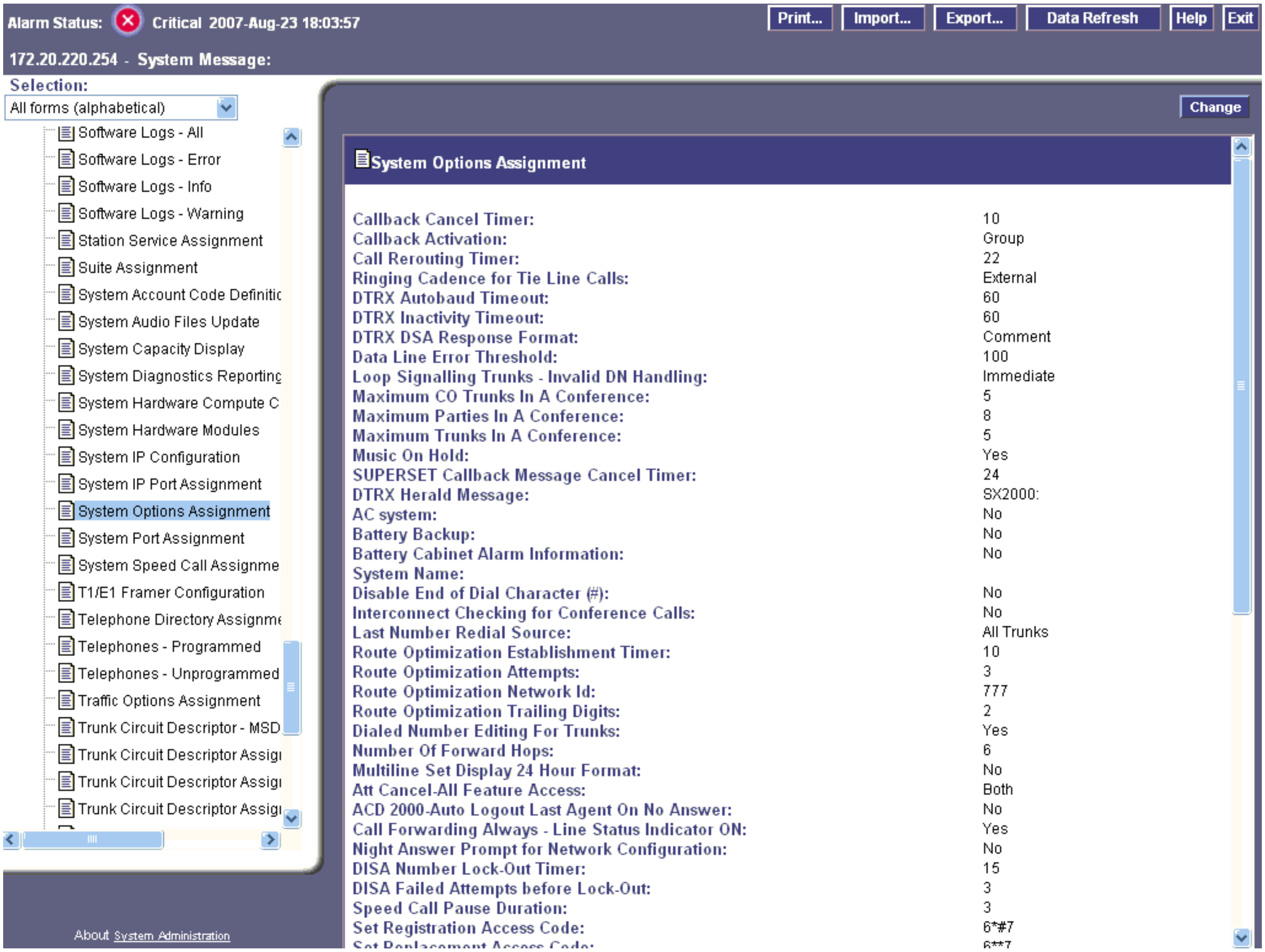


Feature Access Codes – 6 of 6.



System Options

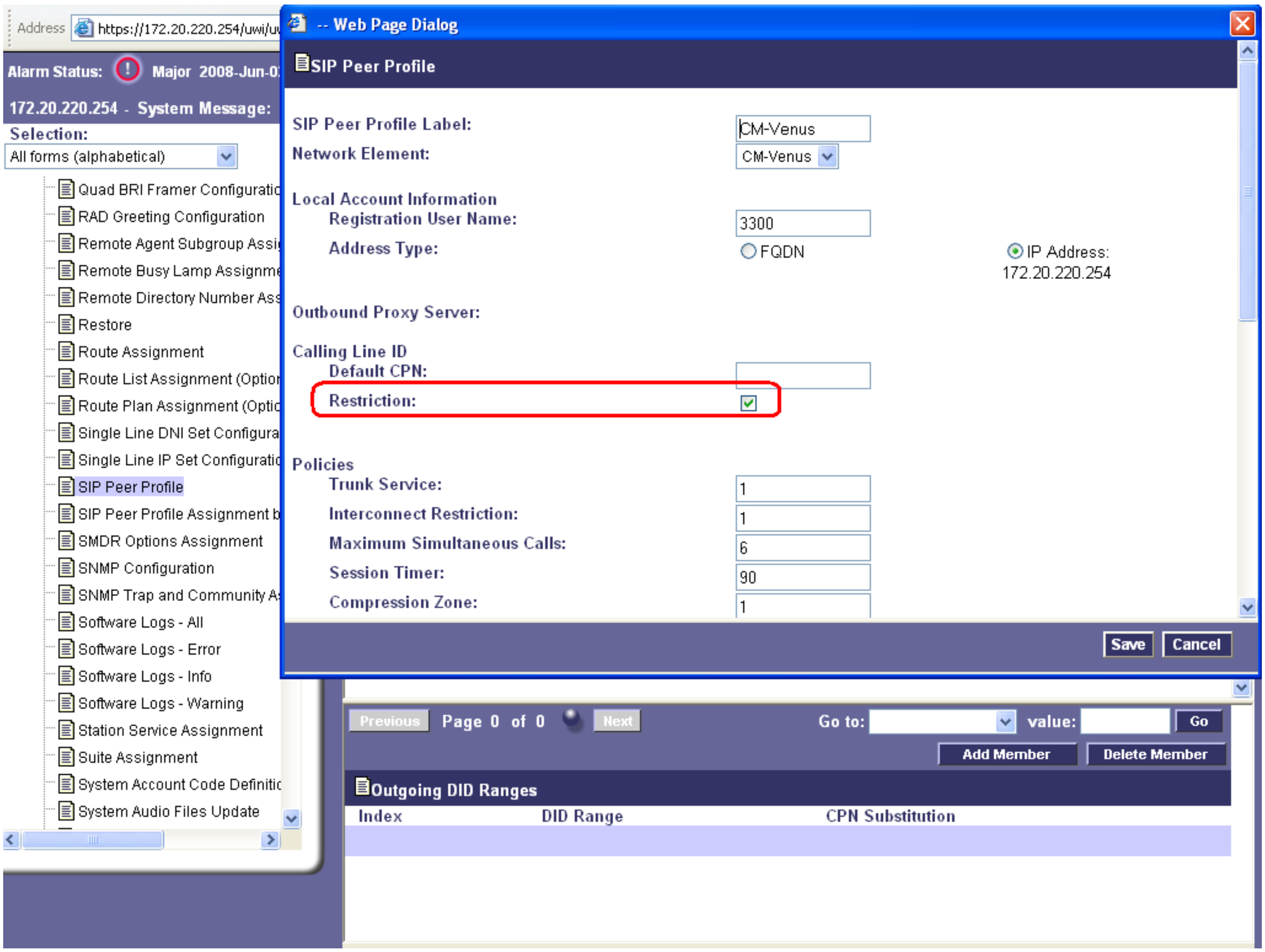
System Options – 1 of 2.



System Options – 2 of 2.

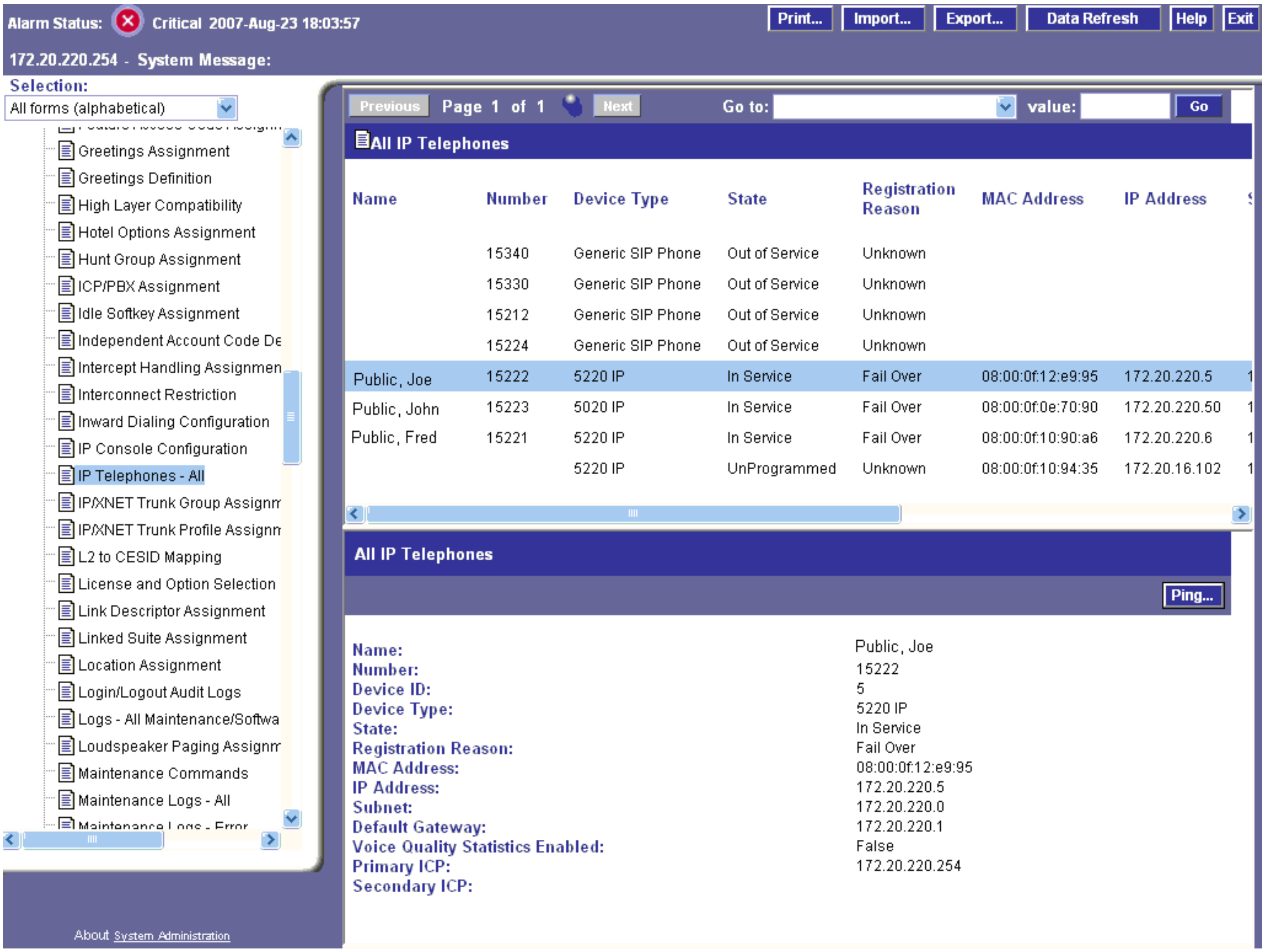
Caller ID Restriction

CLIR on SIP Peer Profile form – 1 of 1.

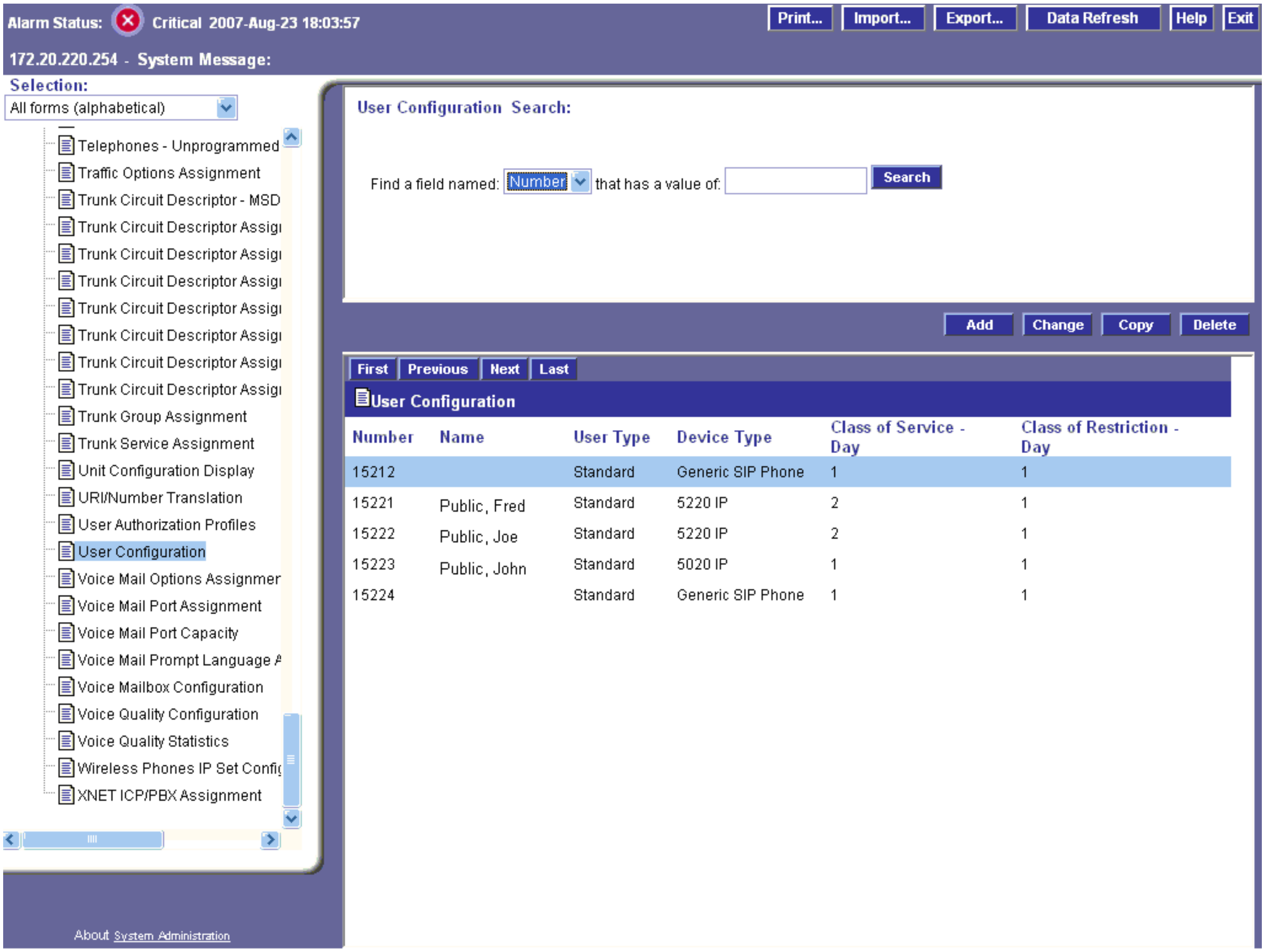


Extensions

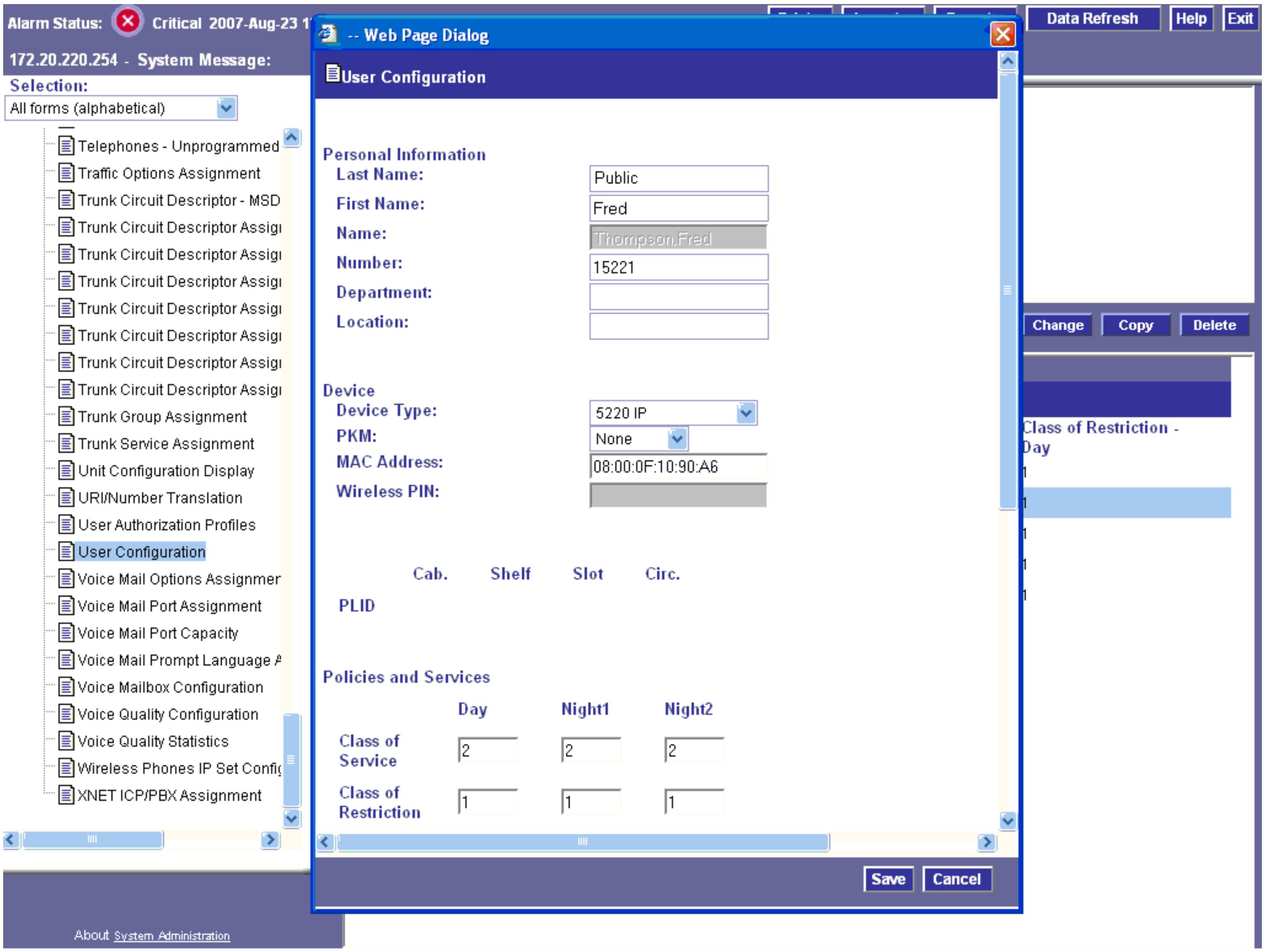
IP Telephones – 1 of 1.



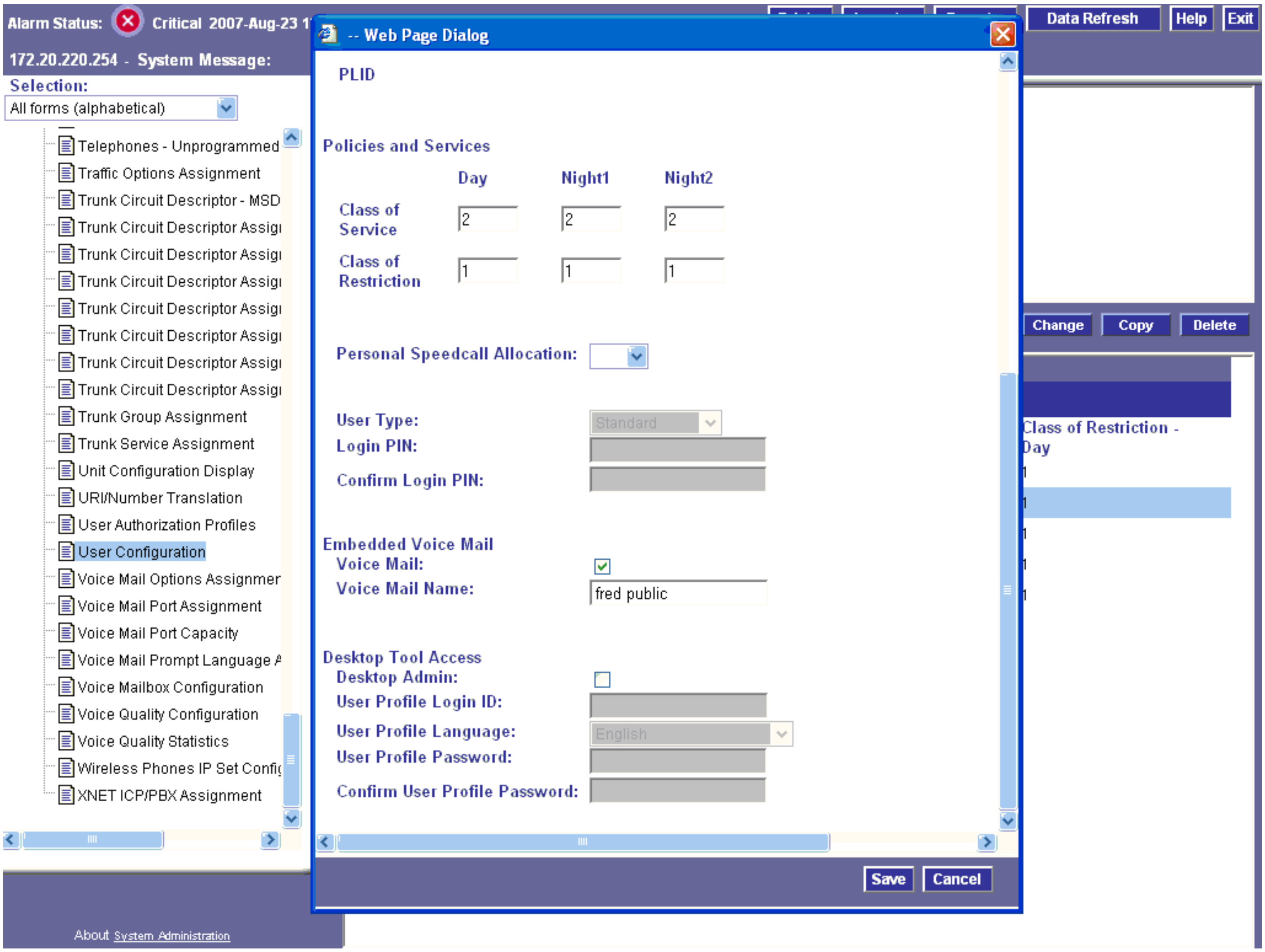
User Configurations – 1 of 1.



User 15221 Configuration – 1 of 2.

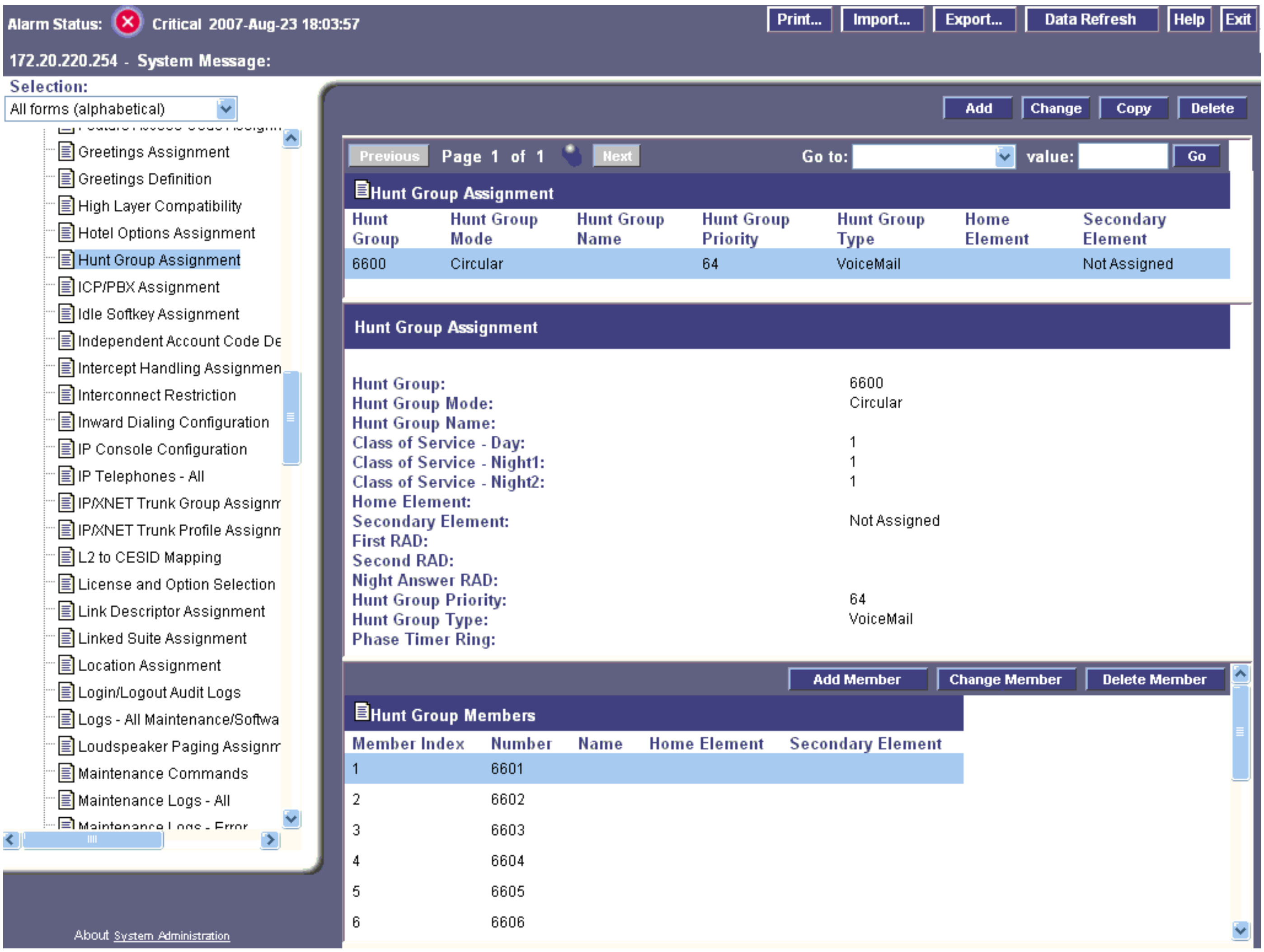


User 15222 Configuration – 2 of 2.

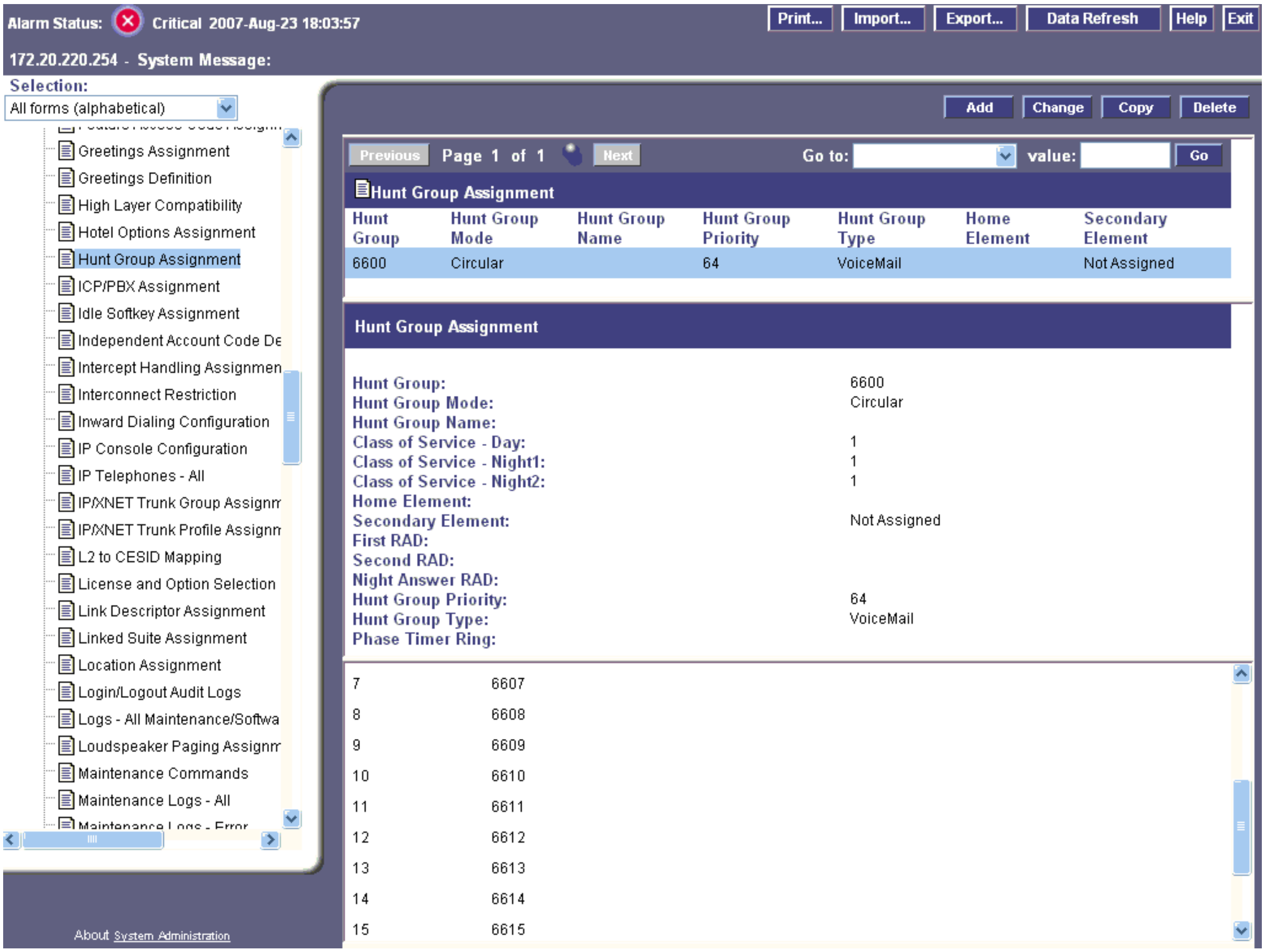


Voice Mail

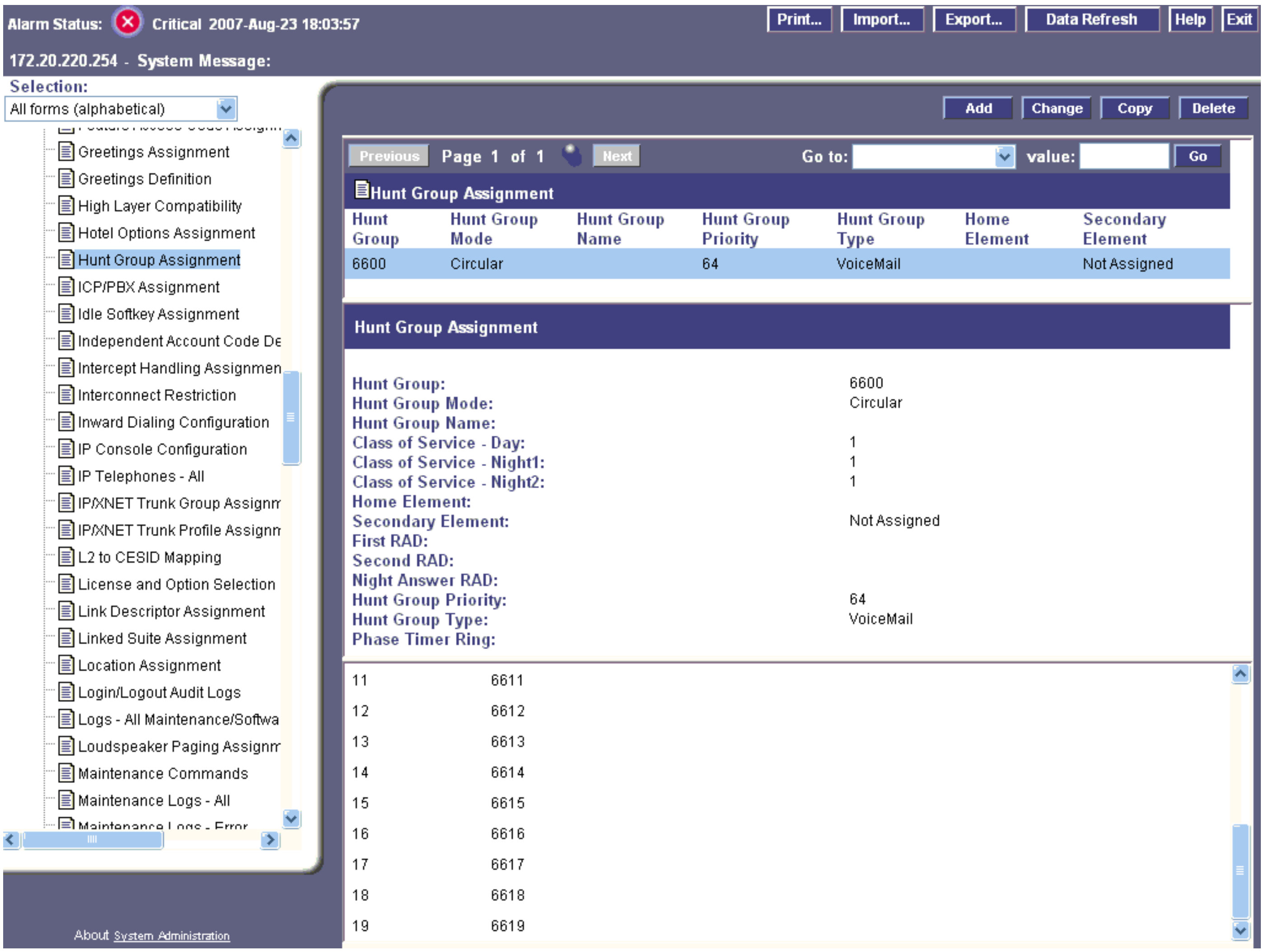
Hunt Groups – 1 of 3.



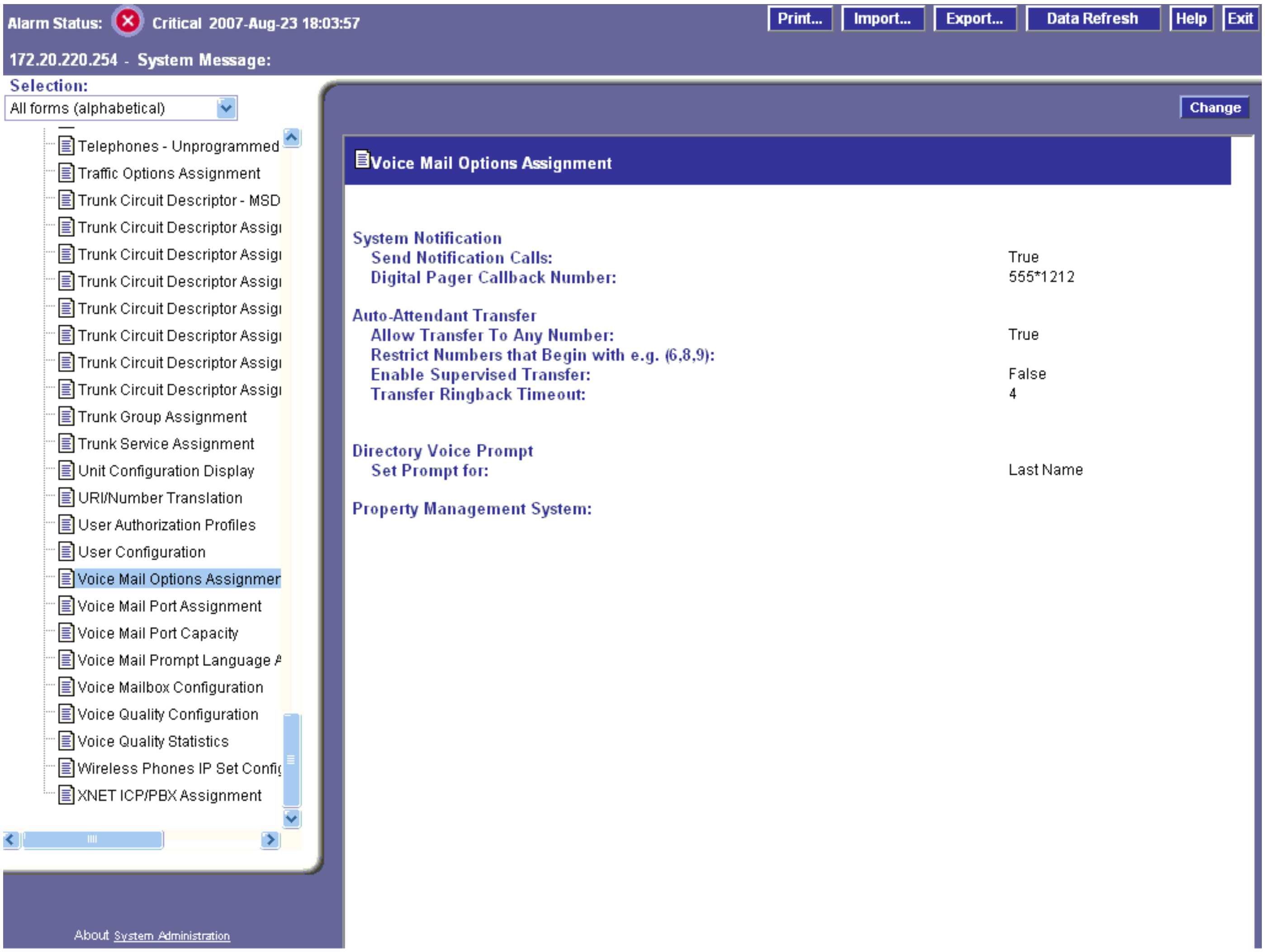
Hunt Groups – 2 of 3.



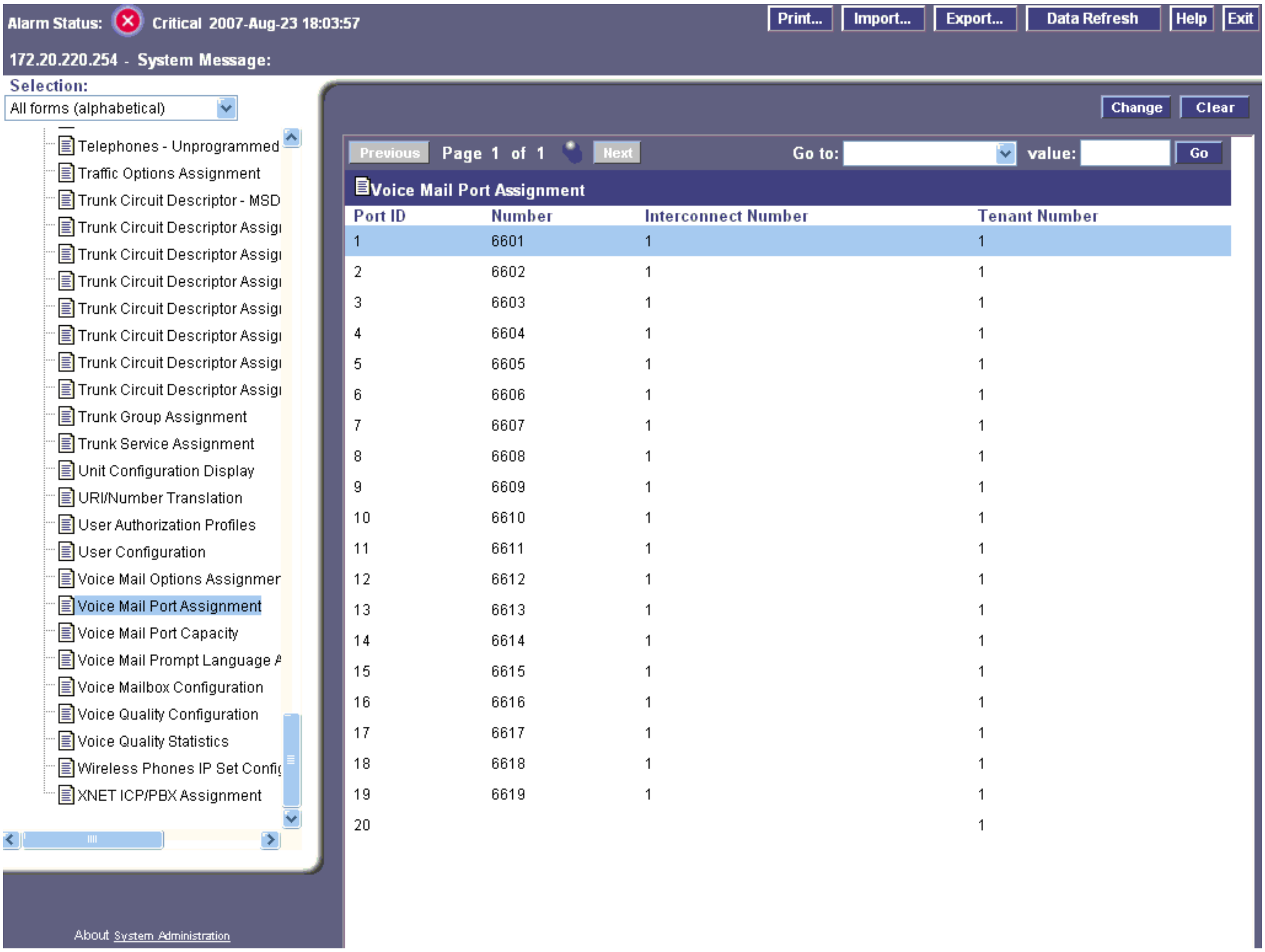
Hunt Groups – 3 of 3.



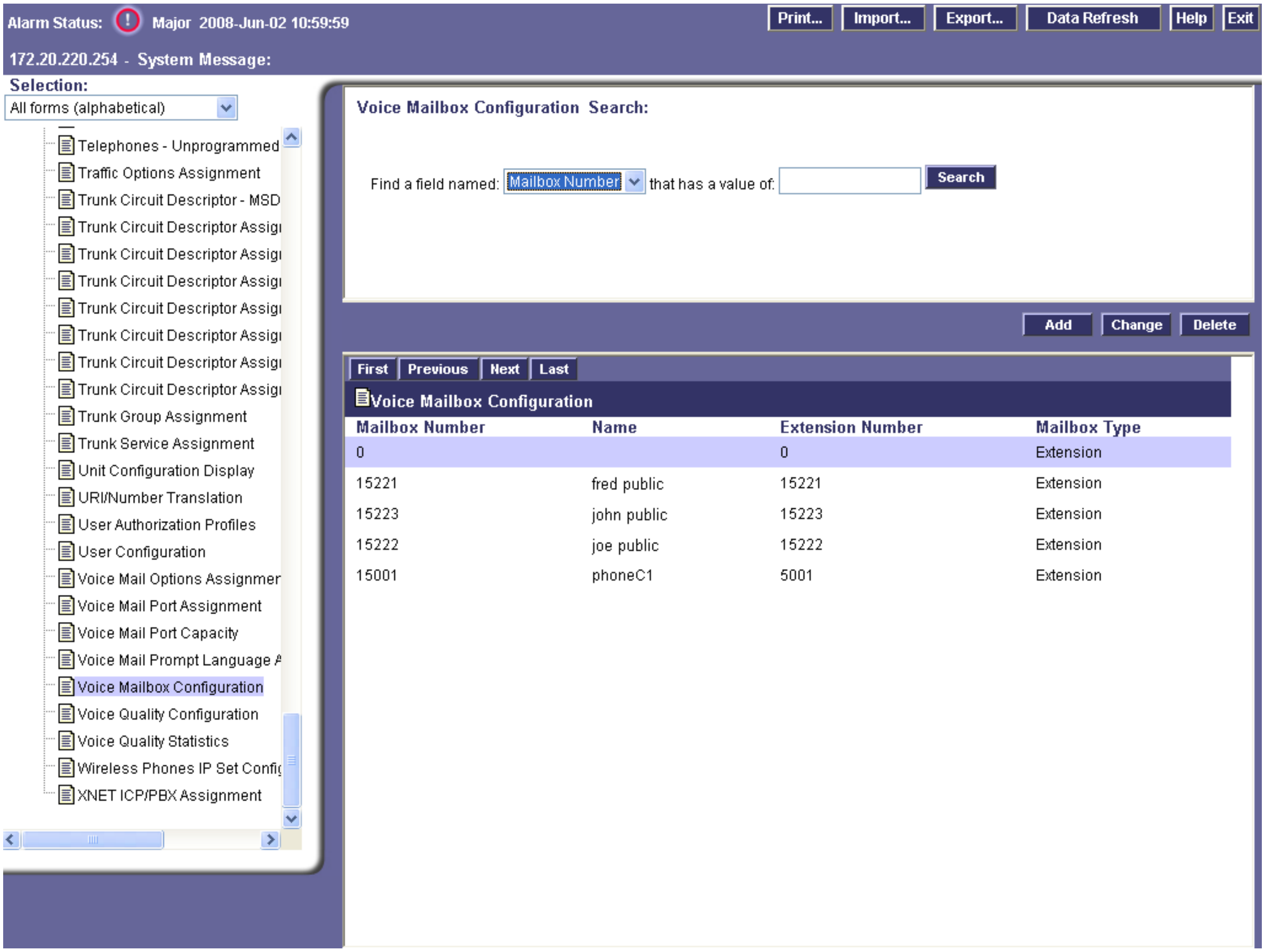
Voice Mail Options Assignment – 1 of 1.



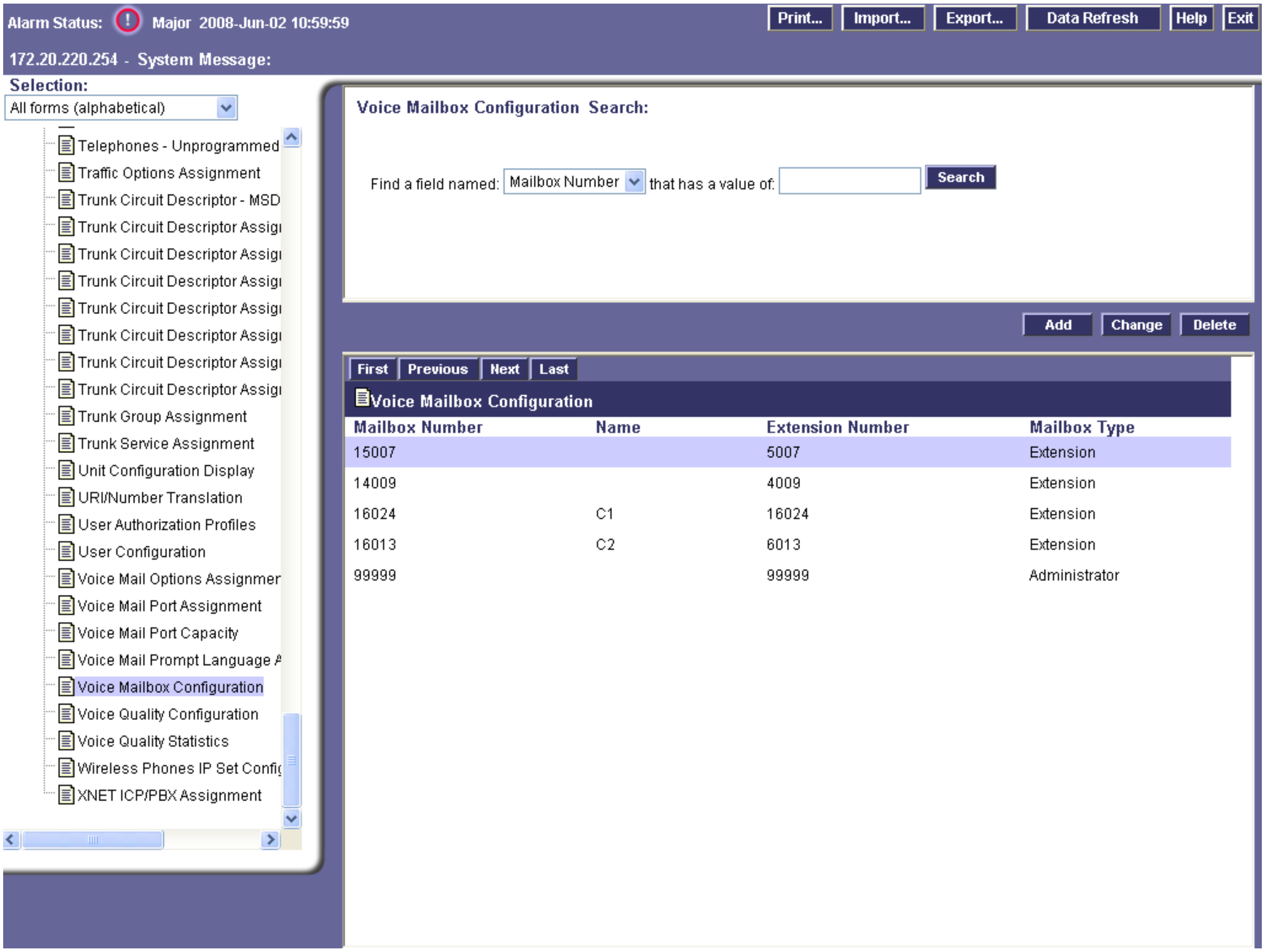
Voice Mail Port Assignment – 1 of 1.



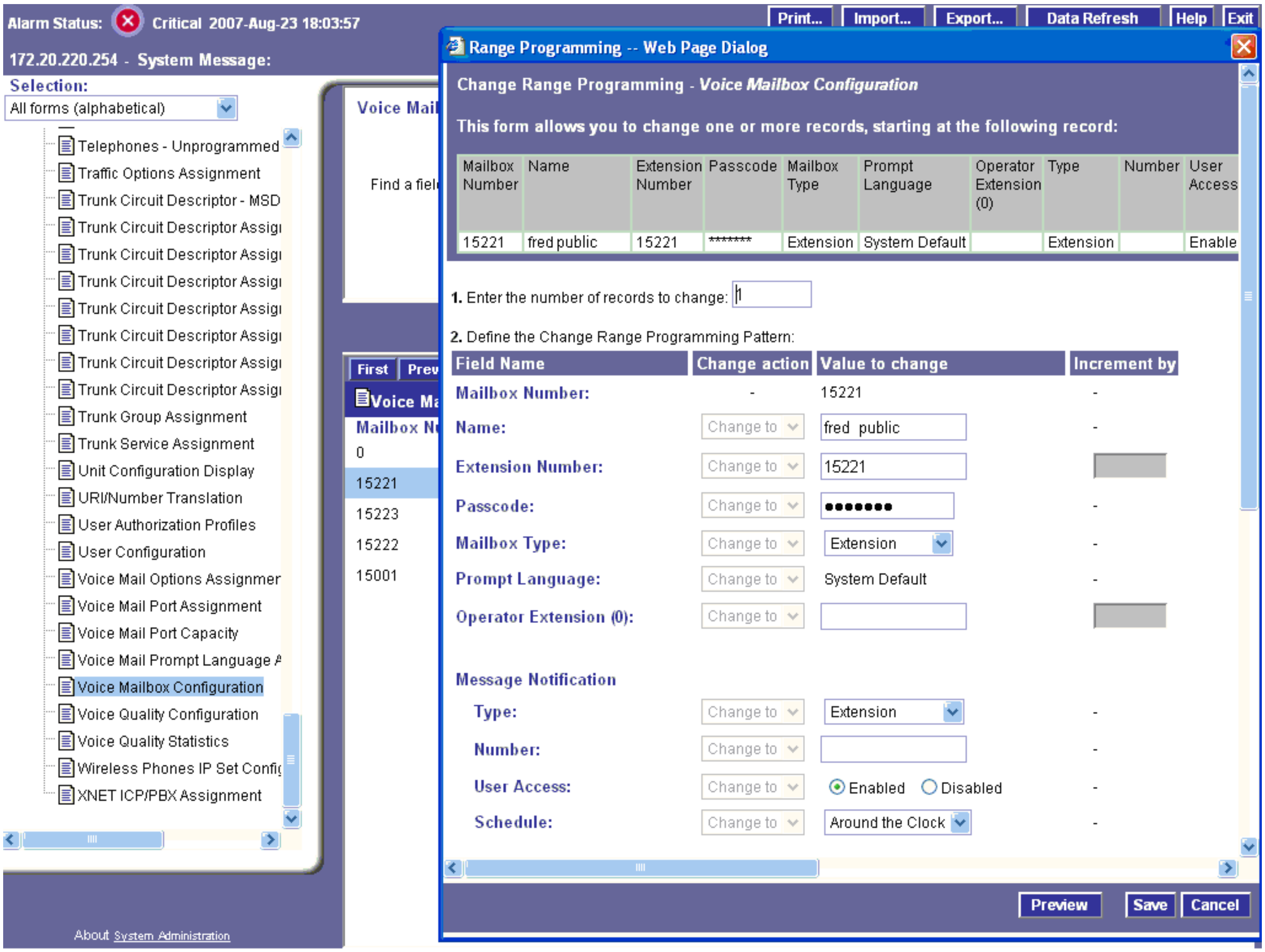
Voice Mailbox Configuration – 1 of 2.



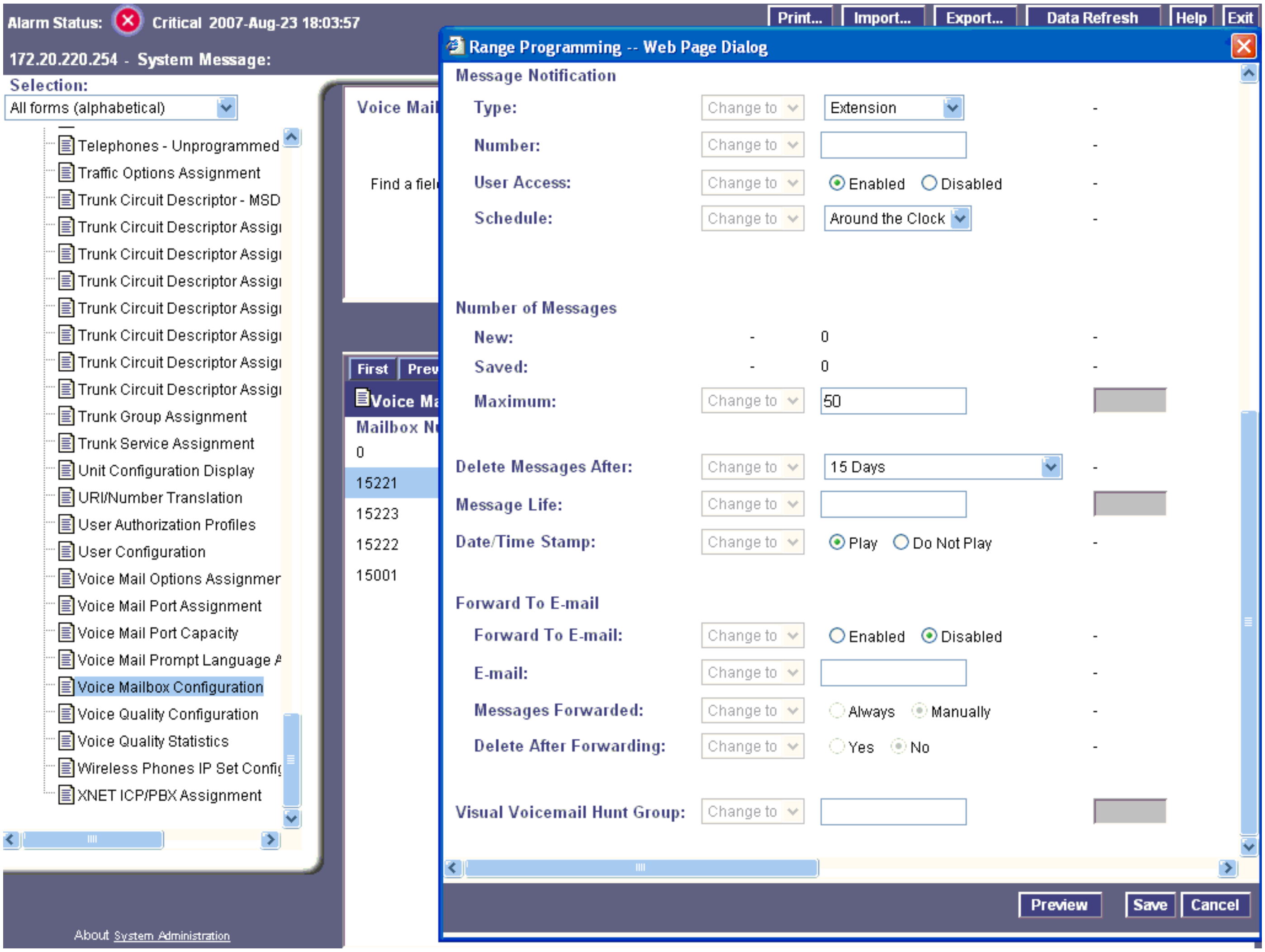
Voice Mailbox Configuration – 2 of 2.



Voice Mailbox 15221 Configuration – 1 of 2.



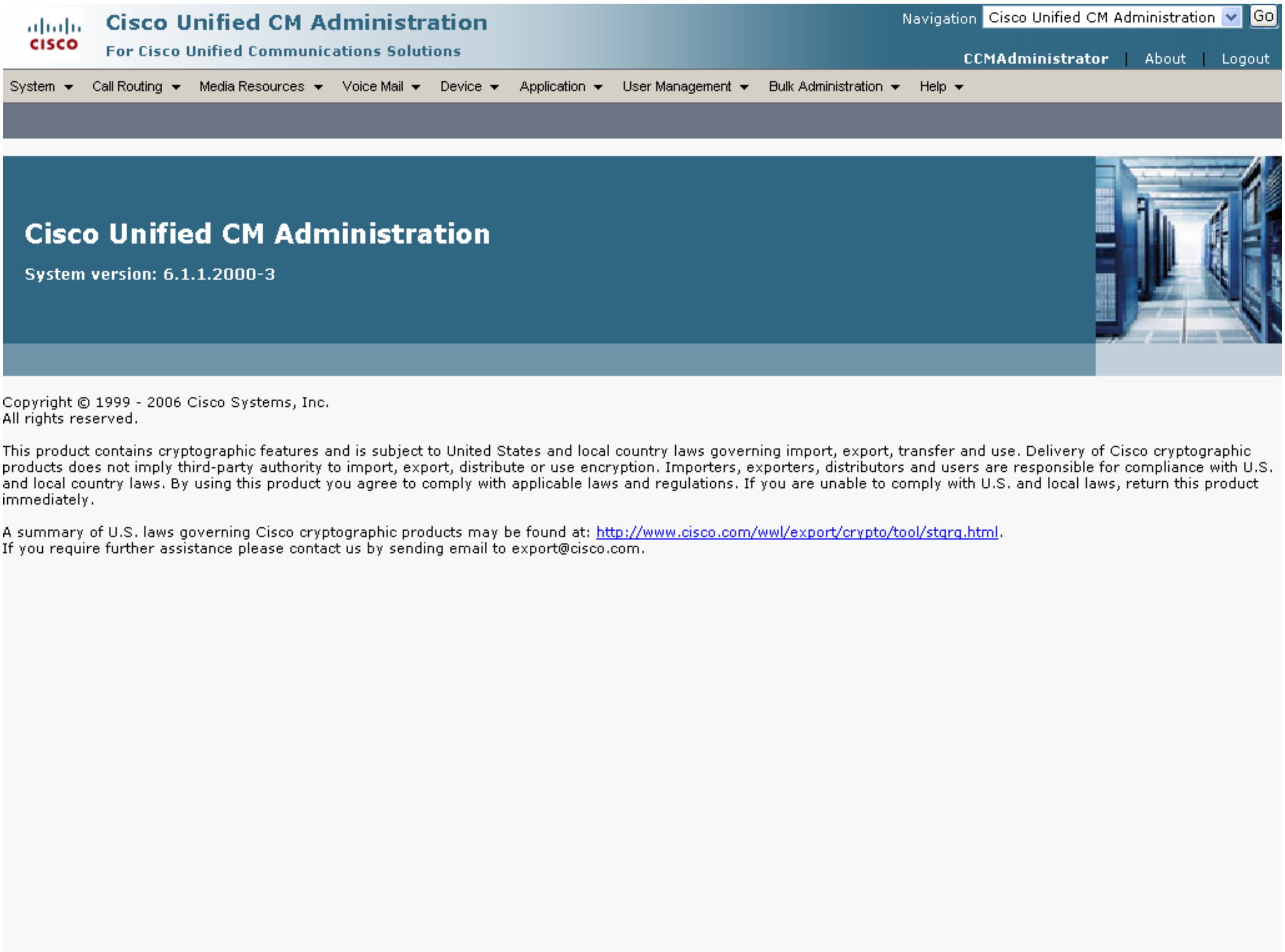
Voice Mailbox 15221 Configuration – 2 of 2.



Configuring the Cisco Unified Communications Manager 6.1

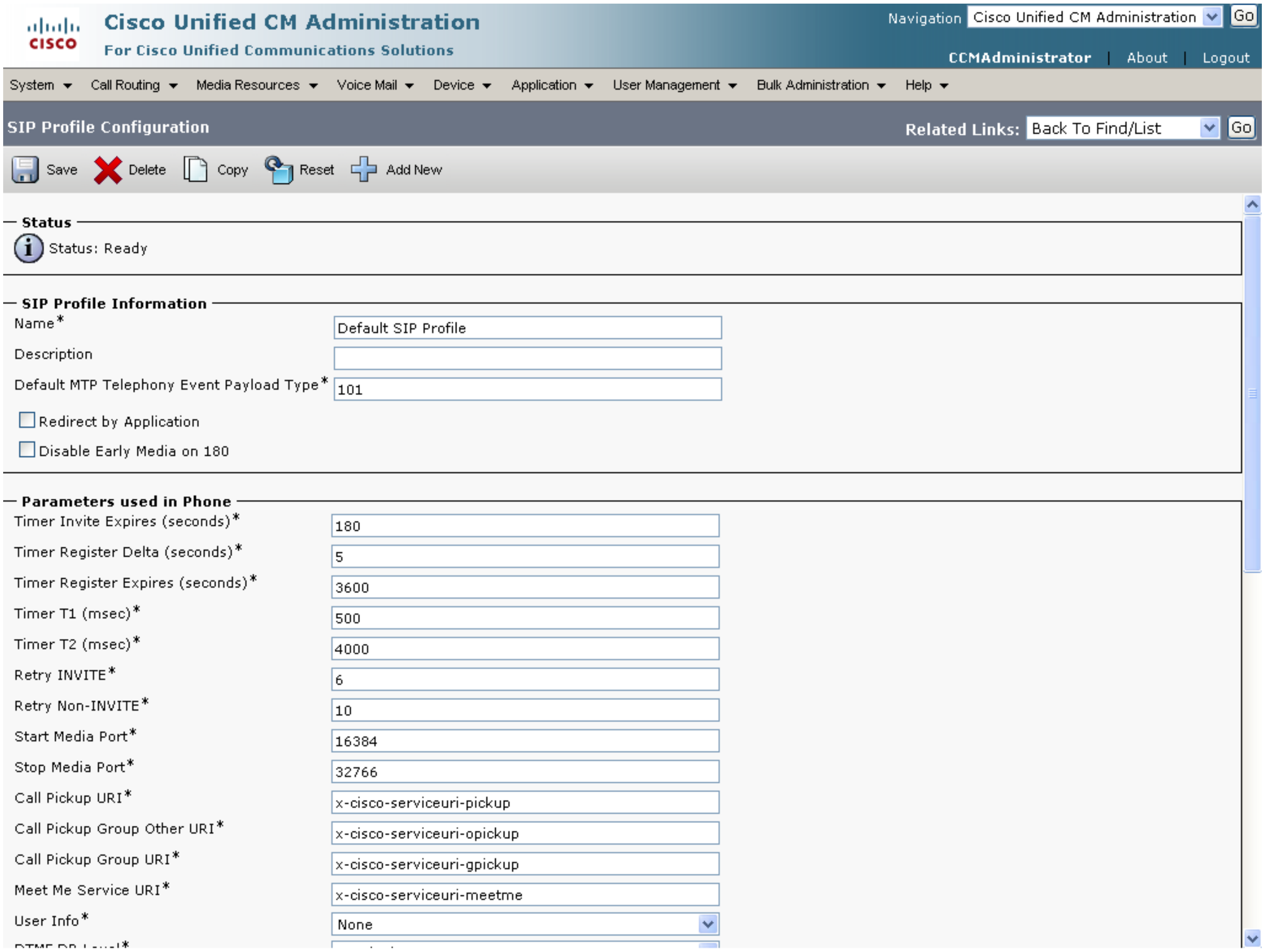
Software Version

Software Version – 1 of 1.

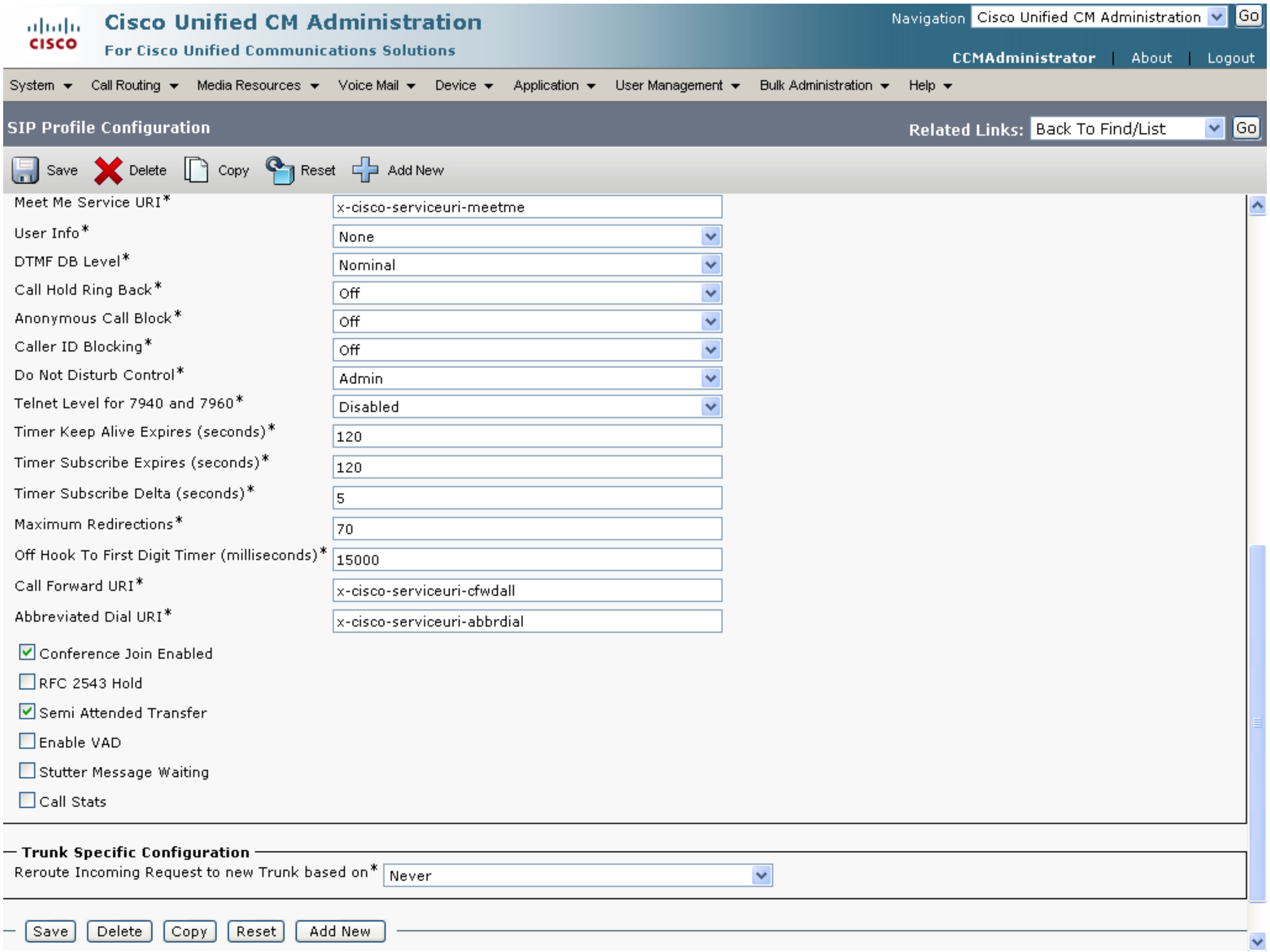


SIP Profile

Default SIP Profile – 1 of 2.

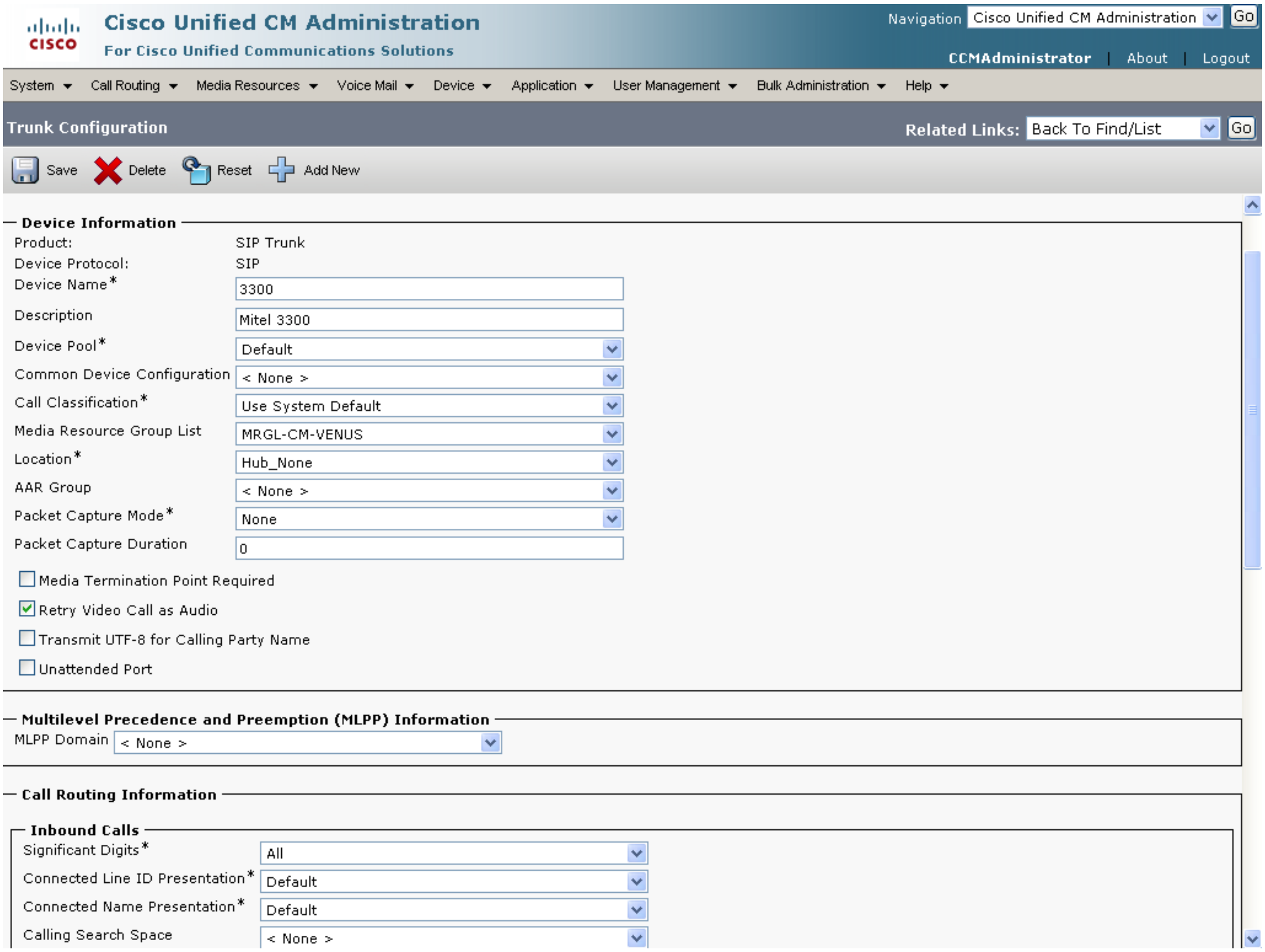


Default SIP Profile – 2 of 2.

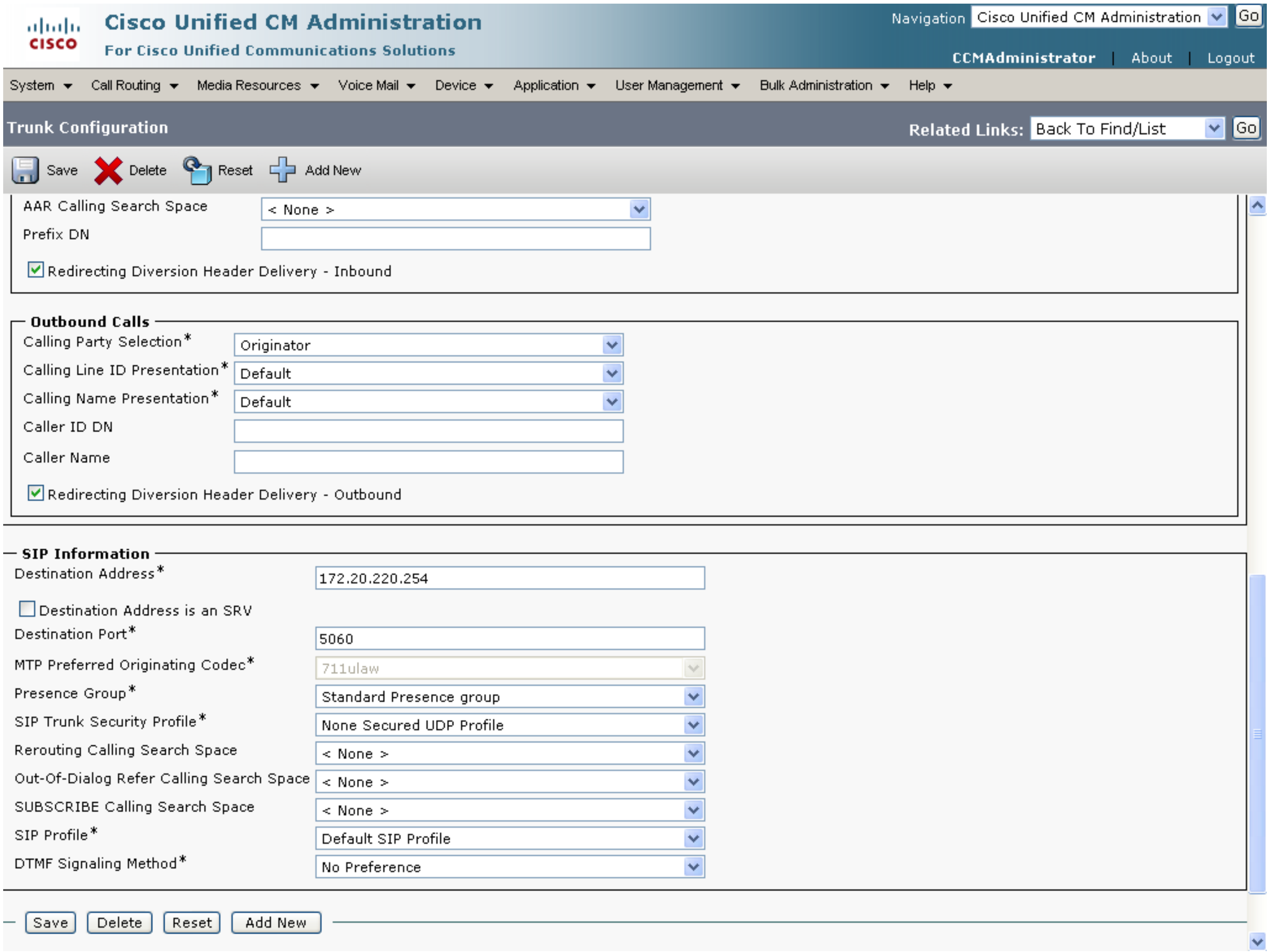


SIP Trunk

SIP Trunk – 1 of 2.

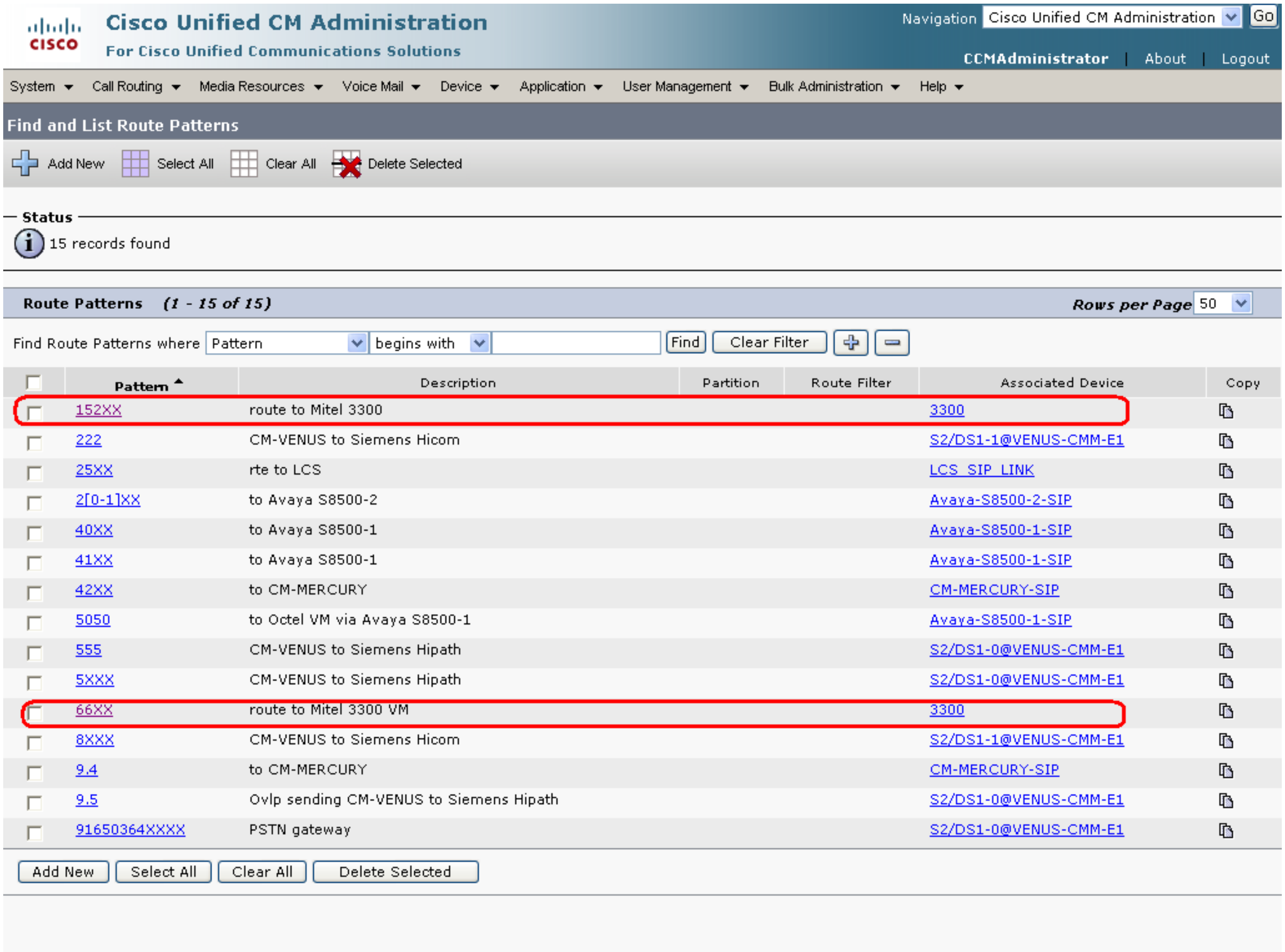


SIP Trunk – 2 of 2.



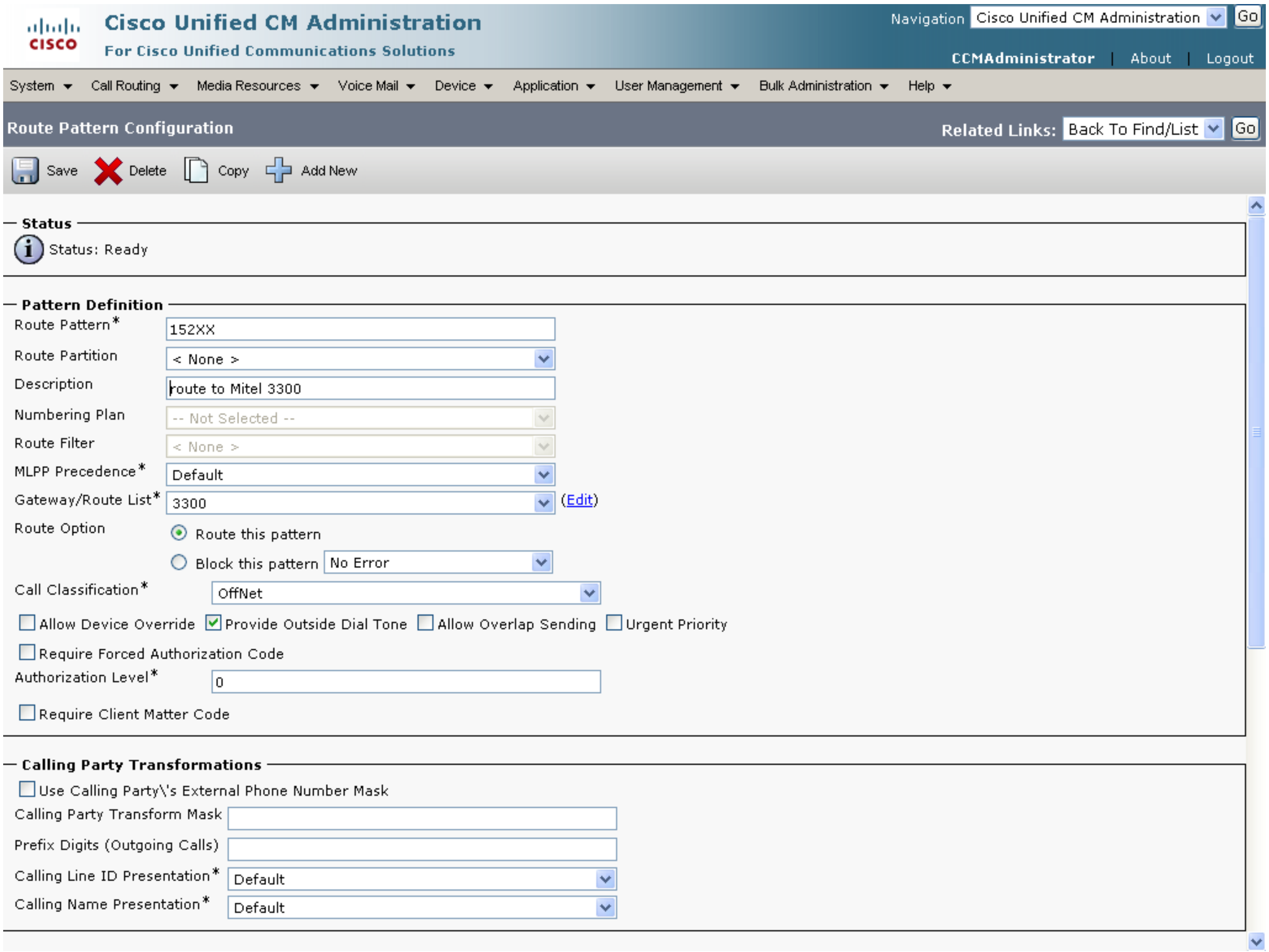
Route Patterns

Route Patterns – 1 of 1.



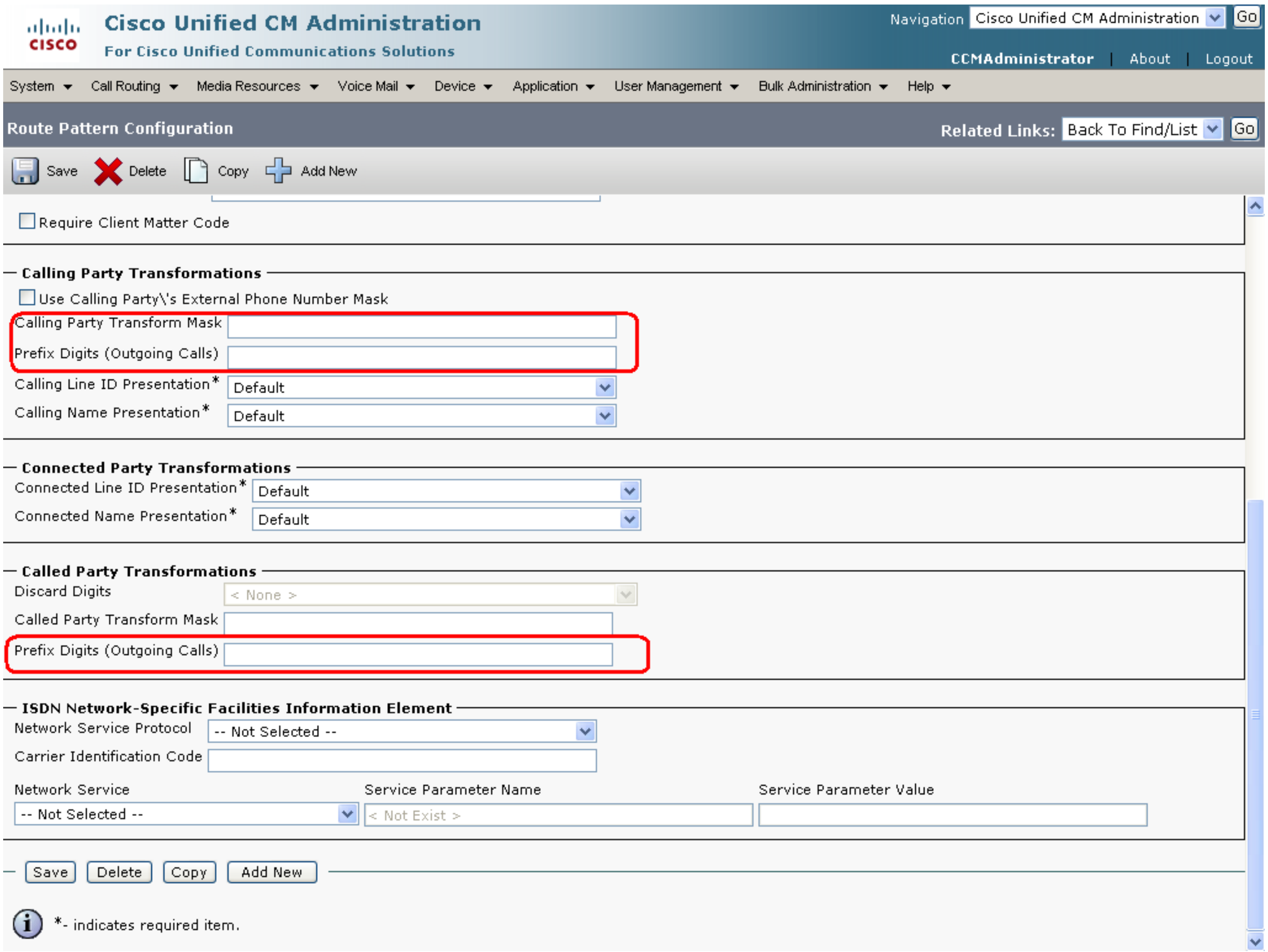
Pattern	Description	Partition	Route Filter	Associated Device	Copy
152XX	route to Mitel 3300			3300	
222	CM-VENUS to Siemens Hicom			S2/DS1-1@VENUS-CMM-E1	
25XX	rte to LCS			LCS_SIP_LINK	
2[0-1]XX	to Avaya S8500-2			Avaya-S8500-2-SIP	
40XX	to Avaya S8500-1			Avaya-S8500-1-SIP	
41XX	to Avaya S8500-1			Avaya-S8500-1-SIP	
42XX	to CM-MERCURY			CM-MERCURY-SIP	
5050	to Octel VM via Avaya S8500-1			Avaya-S8500-1-SIP	
555	CM-VENUS to Siemens Hipath			S2/DS1-0@VENUS-CMM-E1	
5XXX	CM-VENUS to Siemens Hipath			S2/DS1-0@VENUS-CMM-E1	
66XX	route to Mitel 3300 VM			3300	
8XXX	CM-VENUS to Siemens Hicom			S2/DS1-1@VENUS-CMM-E1	
9.4	to CM-MERCURY			CM-MERCURY-SIP	
9.5	Ovlp sending CM-VENUS to Siemens Hipath			S2/DS1-0@VENUS-CMM-E1	
91650364XXXX	PSTN gateway			S2/DS1-0@VENUS-CMM-E1	

152XX Route Pattern – 1 of 2.

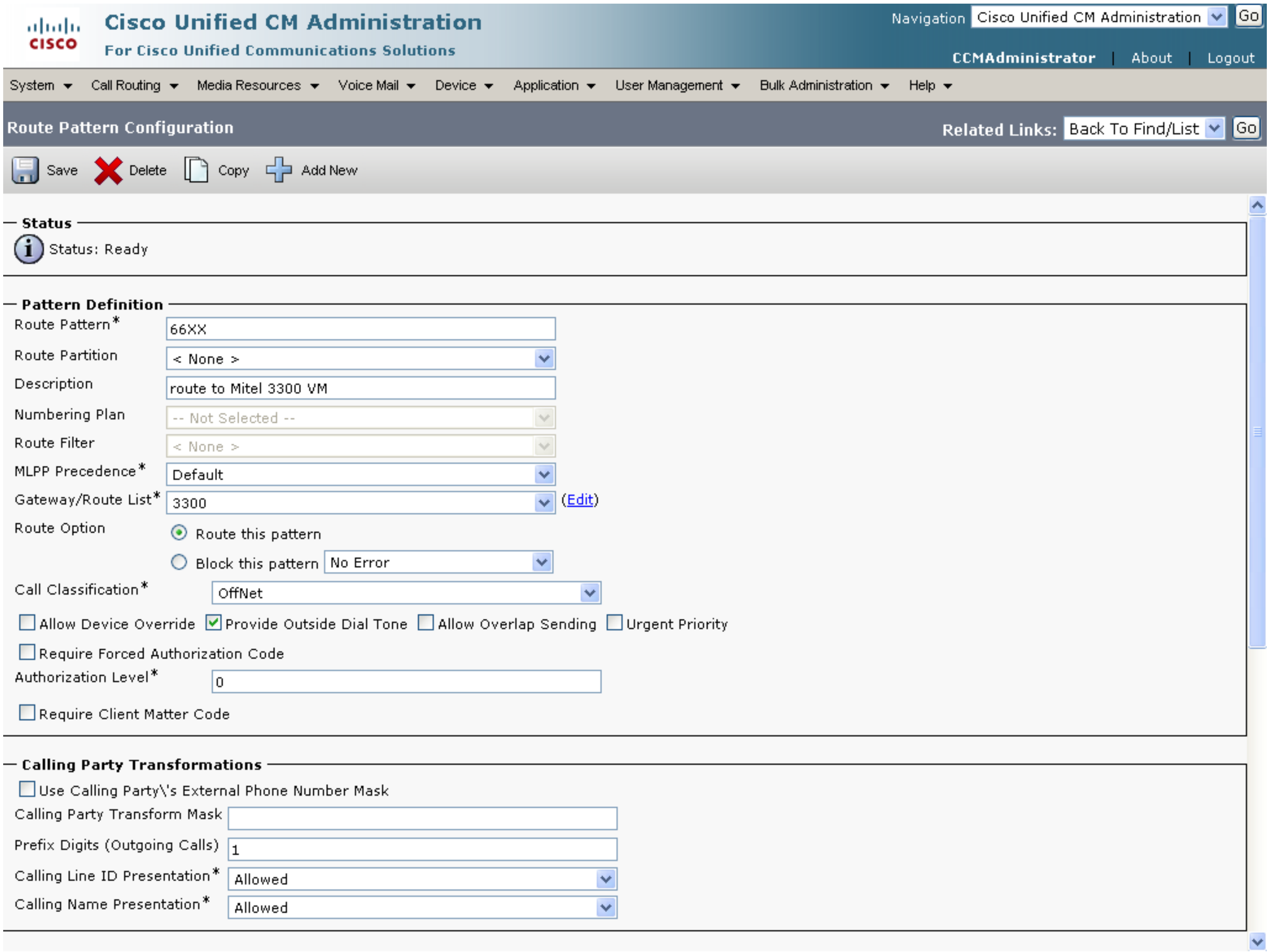


152XX Route Pattern – 2 of 2.

Note: There is no prefix digit "1" in this route pattern, because a 5-digit route pattern (1522X) was used to map directly from Cisco Unified Communications Manager 6.1 to the Mitel 3300 ICP.

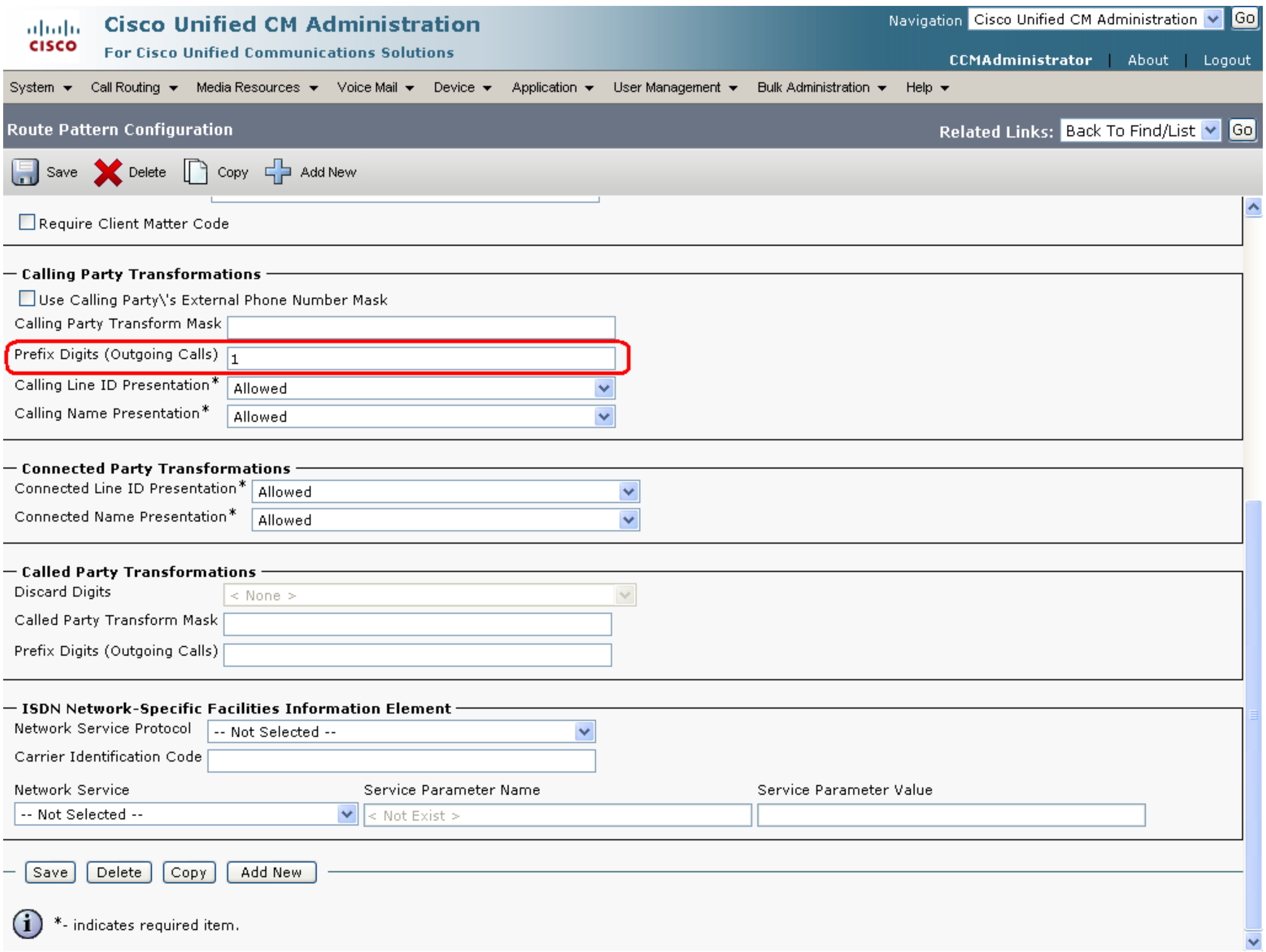


66XX Route Pattern (with Calling Party Transform Mask) for Mitel 3300 ICP Voice Mail – 1 of 2.



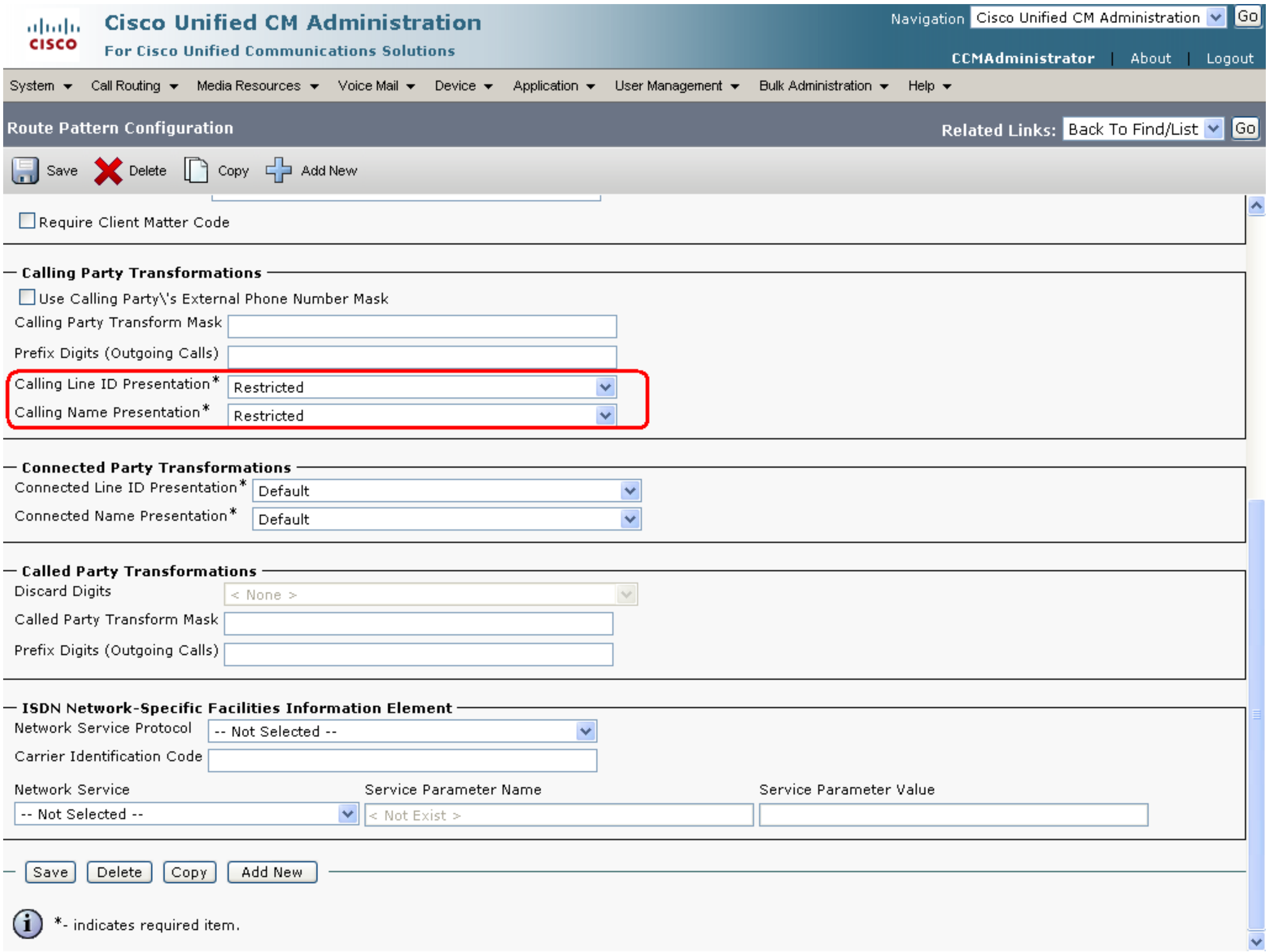
1522X Route Pattern (with Calling Party Transform Mask) for Mitel 3300 ICP Voice Mail – 2 of 2.

Note: The prefix digit "1" was used to adapt the Cisco Unified Communications Manager number plan, which contained 4 digits, to the Mitel 3300 ICP number plan, which contained 5 digits. The Mitel voice mailboxes can not vary in length, and they are set up to match the length of the Mitel extensions, which are 5 digits. Voice mailboxes for Cisco Unified Communications Manager extensions were set up on the Mitel 3300 ICP with an additional "1". For instance, voice mailbox "16024" corresponds to Cisco Unified Communications Manager extension "6024". The prefix digit "1" in this route pattern exists so that the correct number will be sent to the Mitel 3300 ICP in the SIP INVITE.



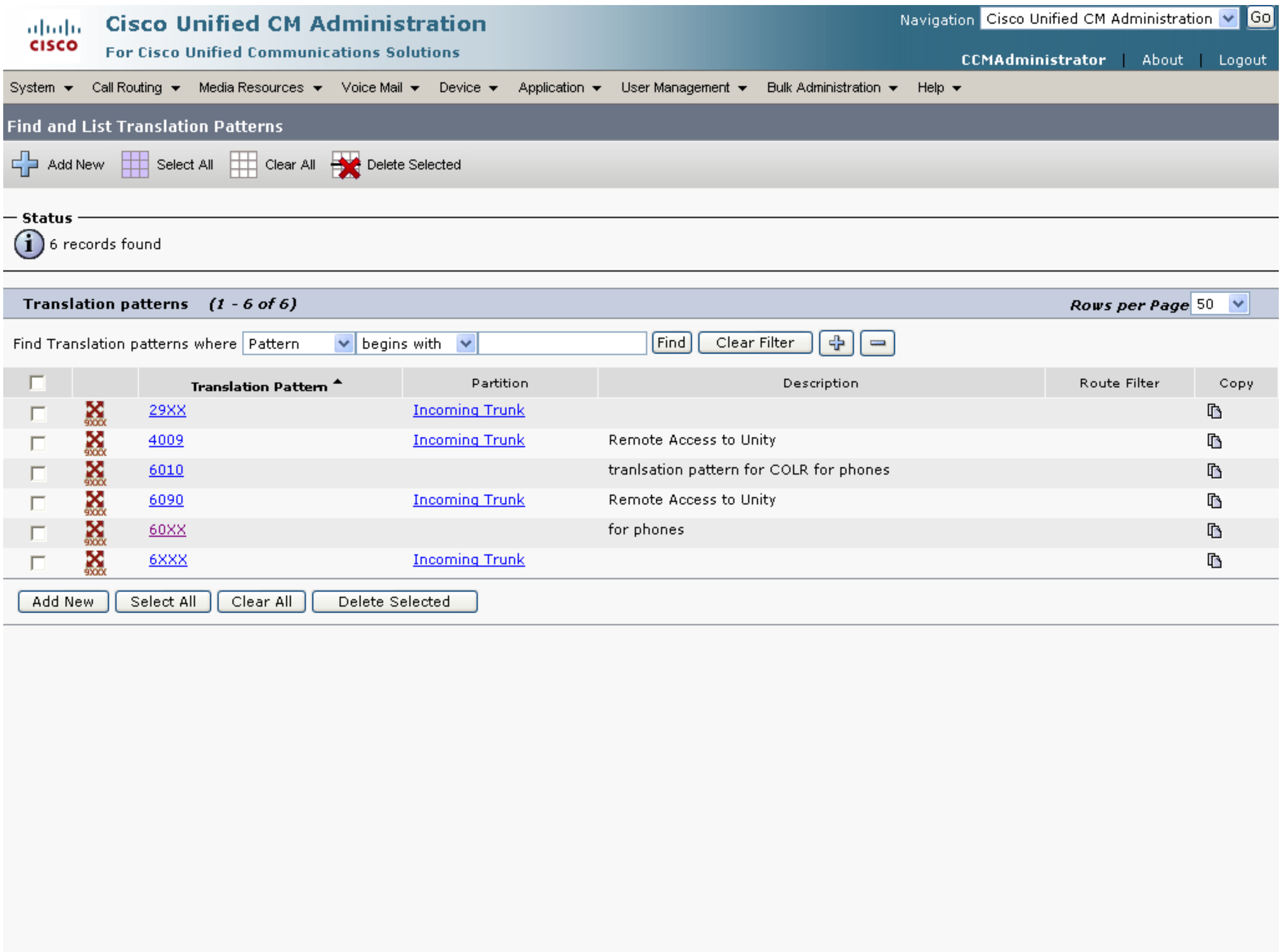
Calling Line ID Restriction

CLIR on 522X Route Pattern – 1 of 1.

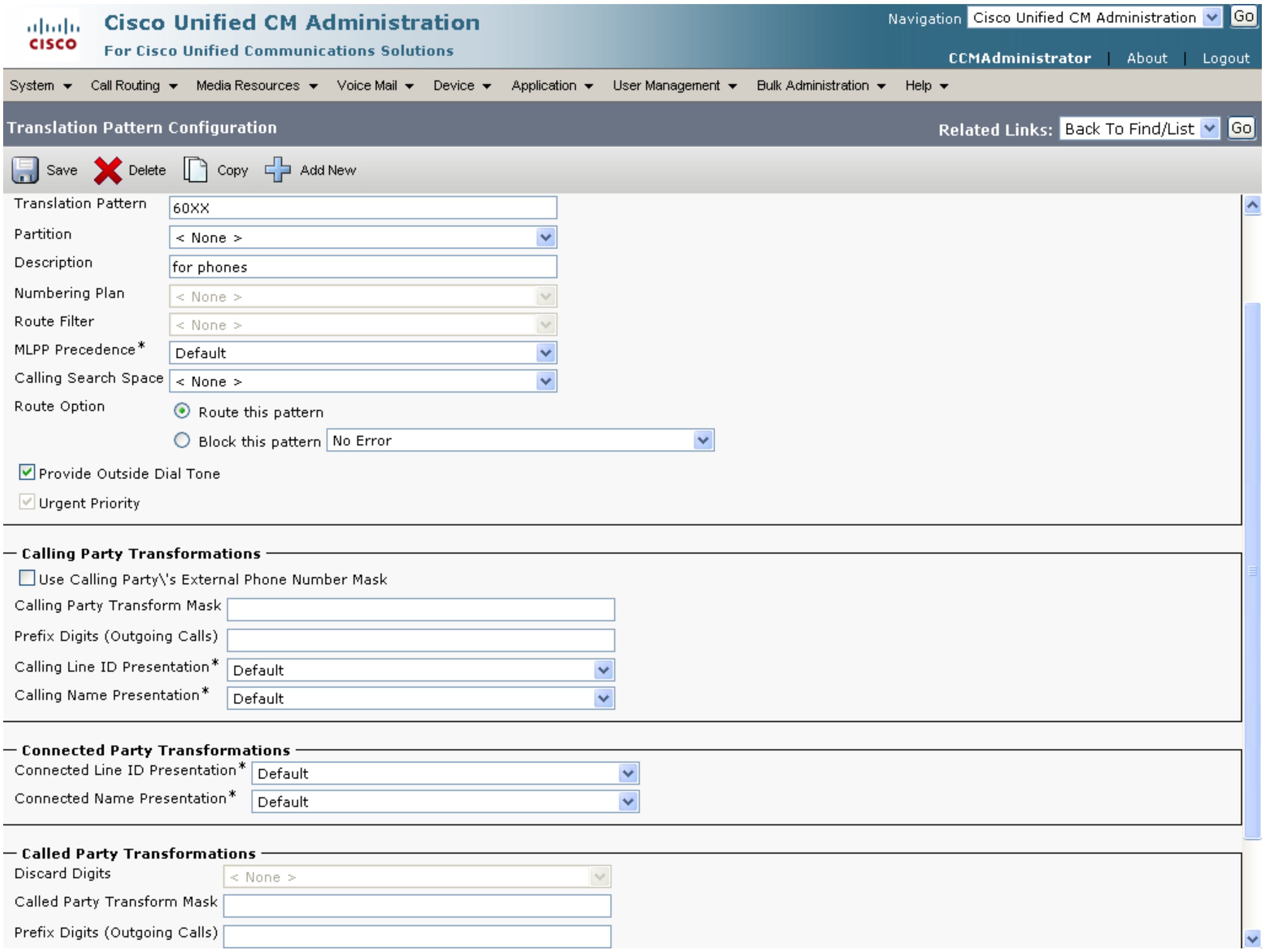


Translation Patterns

Translation Patterns – 1 of 1.

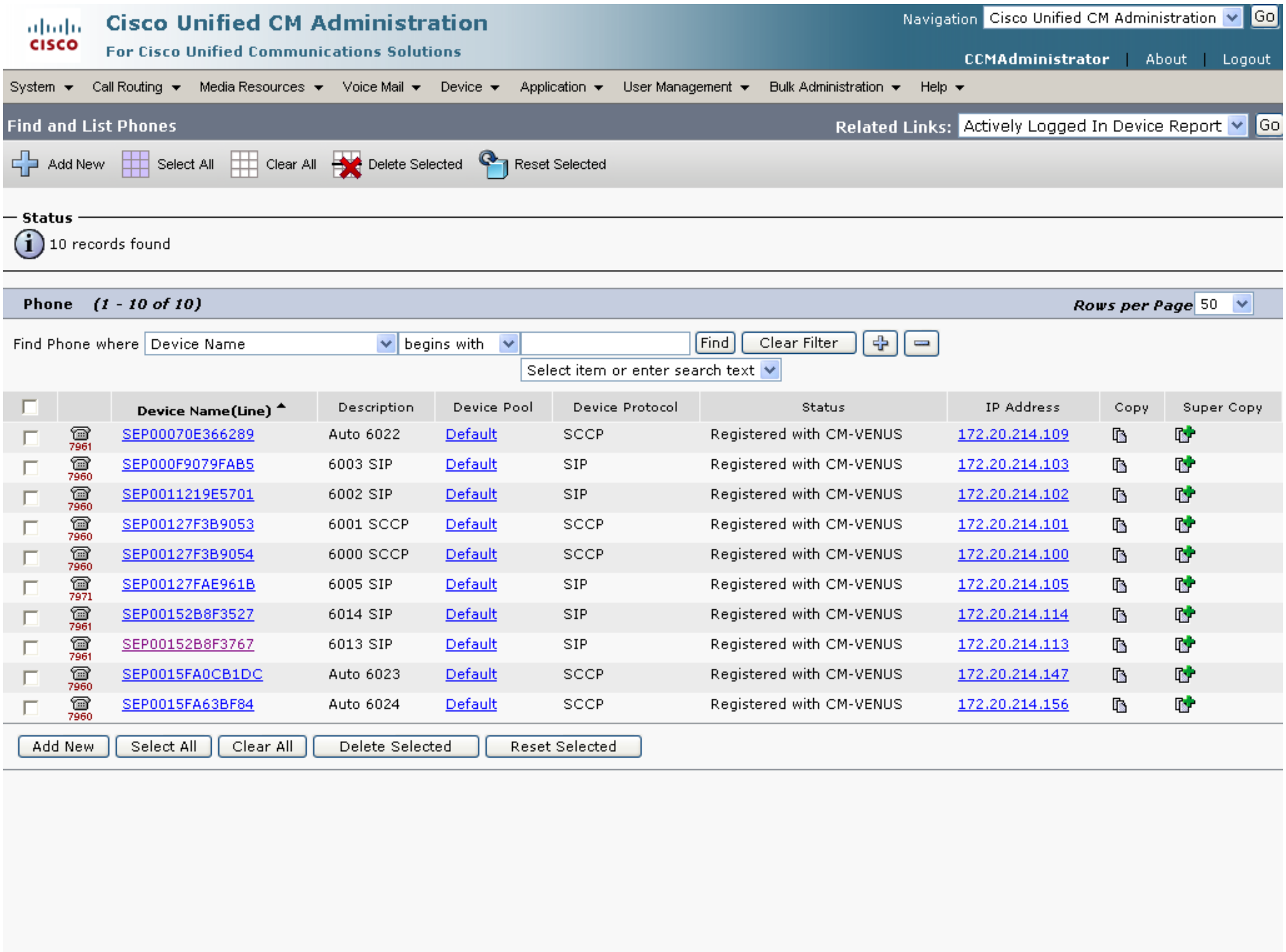


Translation Pattern 60XX for Cisco Unified IP Phones – 1 of 1.



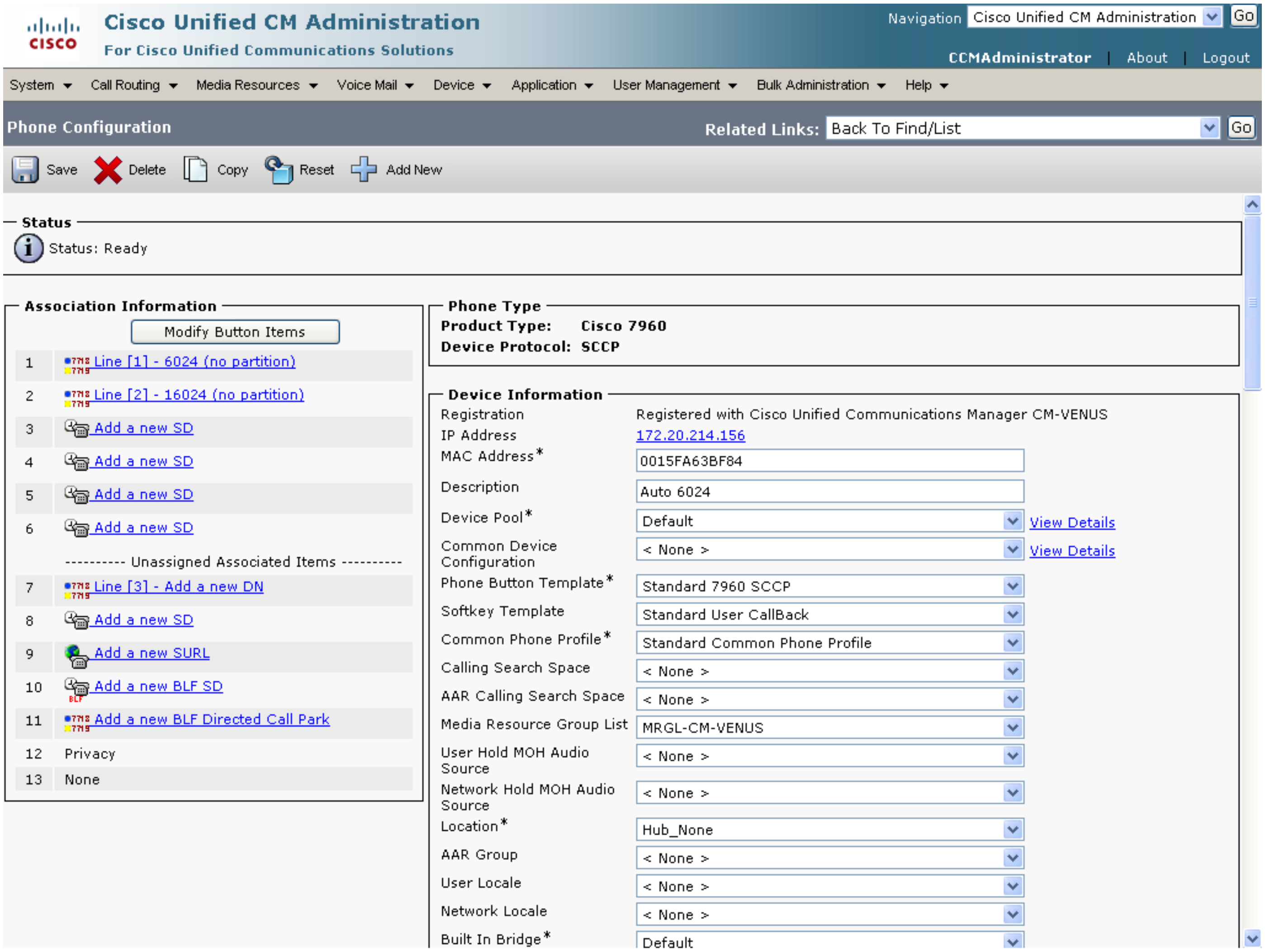
IP Phones

IP Phones – 1 of 1.

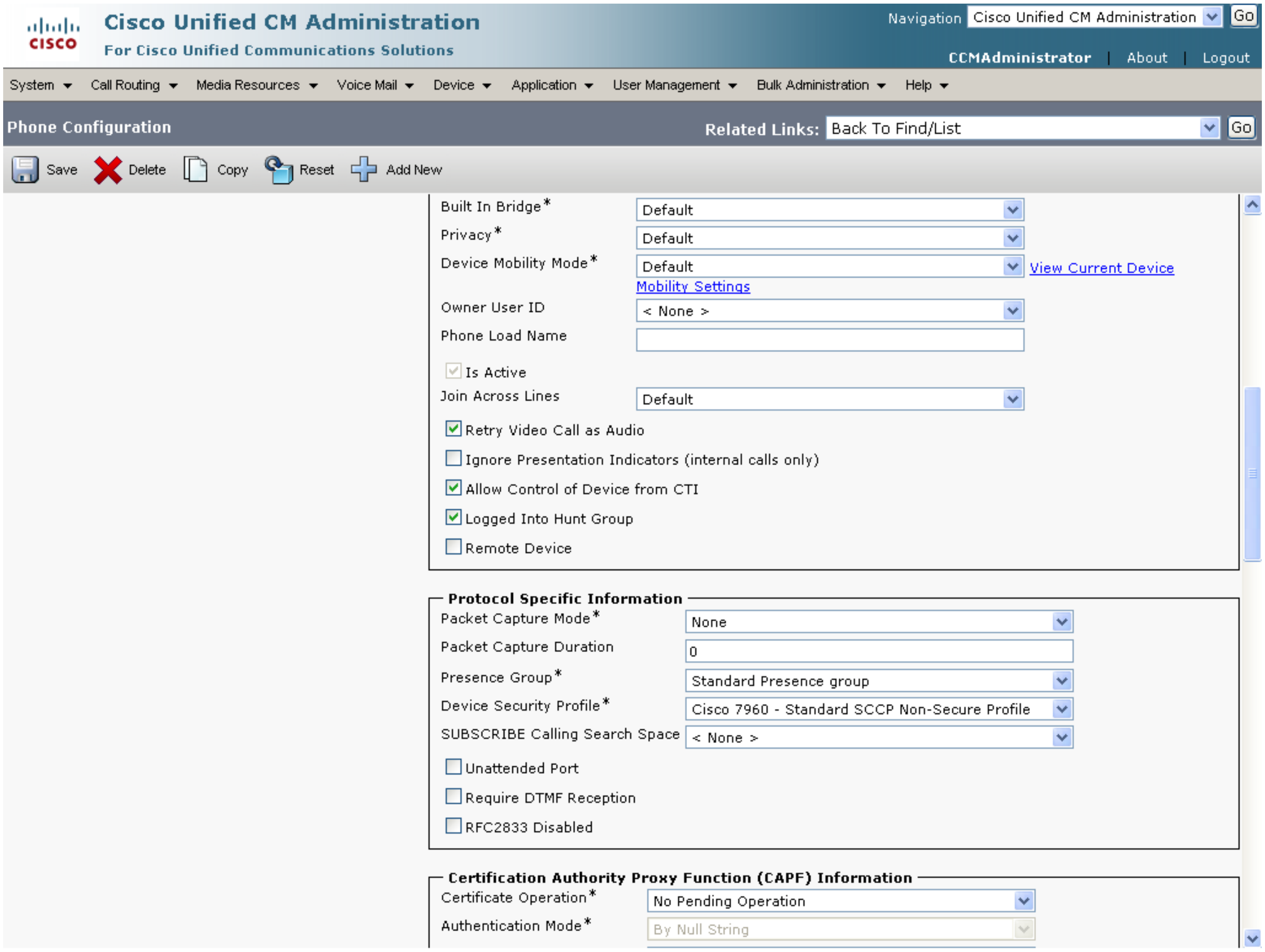


		Device Name(Line)	Description	Device Pool	Device Protocol	Status	IP Address	Copy	Super Copy
	7961	SEP00070E366289	Auto 6022	Default	SCCP	Registered with CM-VENUS	172.20.214.109		
	7960	SEP000F9079FAB5	6003 SIP	Default	SIP	Registered with CM-VENUS	172.20.214.103		
	7960	SEP0011219E5701	6002 SIP	Default	SIP	Registered with CM-VENUS	172.20.214.102		
	7960	SEP00127F3B9053	6001 SCCP	Default	SCCP	Registered with CM-VENUS	172.20.214.101		
	7960	SEP00127F3B9054	6000 SCCP	Default	SCCP	Registered with CM-VENUS	172.20.214.100		
	7971	SEP00127FAE961B	6005 SIP	Default	SIP	Registered with CM-VENUS	172.20.214.105		
	7961	SEP00152B8F3527	6014 SIP	Default	SIP	Registered with CM-VENUS	172.20.214.114		
	7961	SEP00152B8F3767	6013 SIP	Default	SIP	Registered with CM-VENUS	172.20.214.113		
	7960	SEP0015FA0CB1DC	Auto 6023	Default	SCCP	Registered with CM-VENUS	172.20.214.147		
	7960	SEP0015FA63BF84	Auto 6024	Default	SCCP	Registered with CM-VENUS	172.20.214.156		

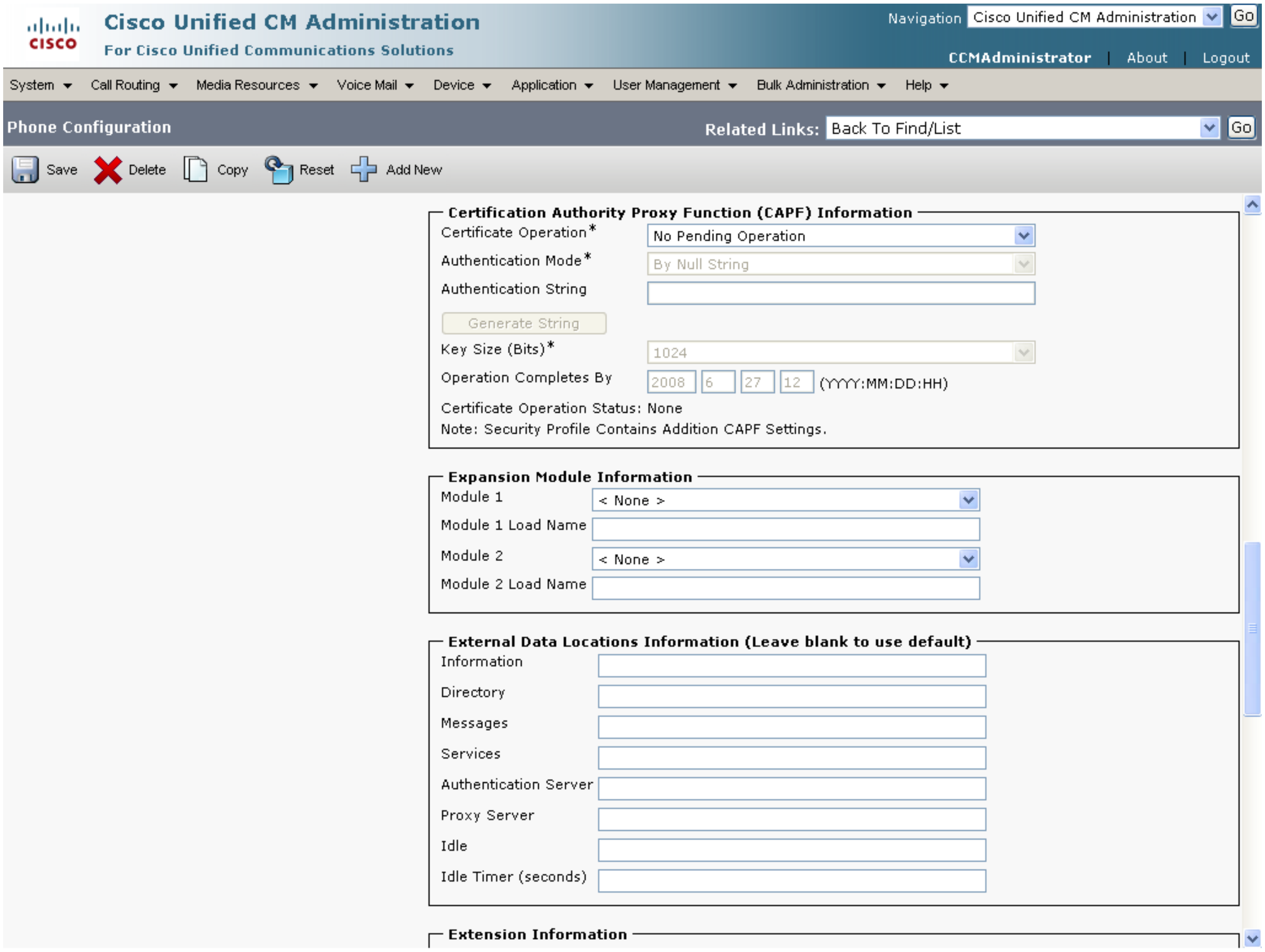
Cisco Unified IP Phone 7960, x6024, SCCP – 1 of 4.



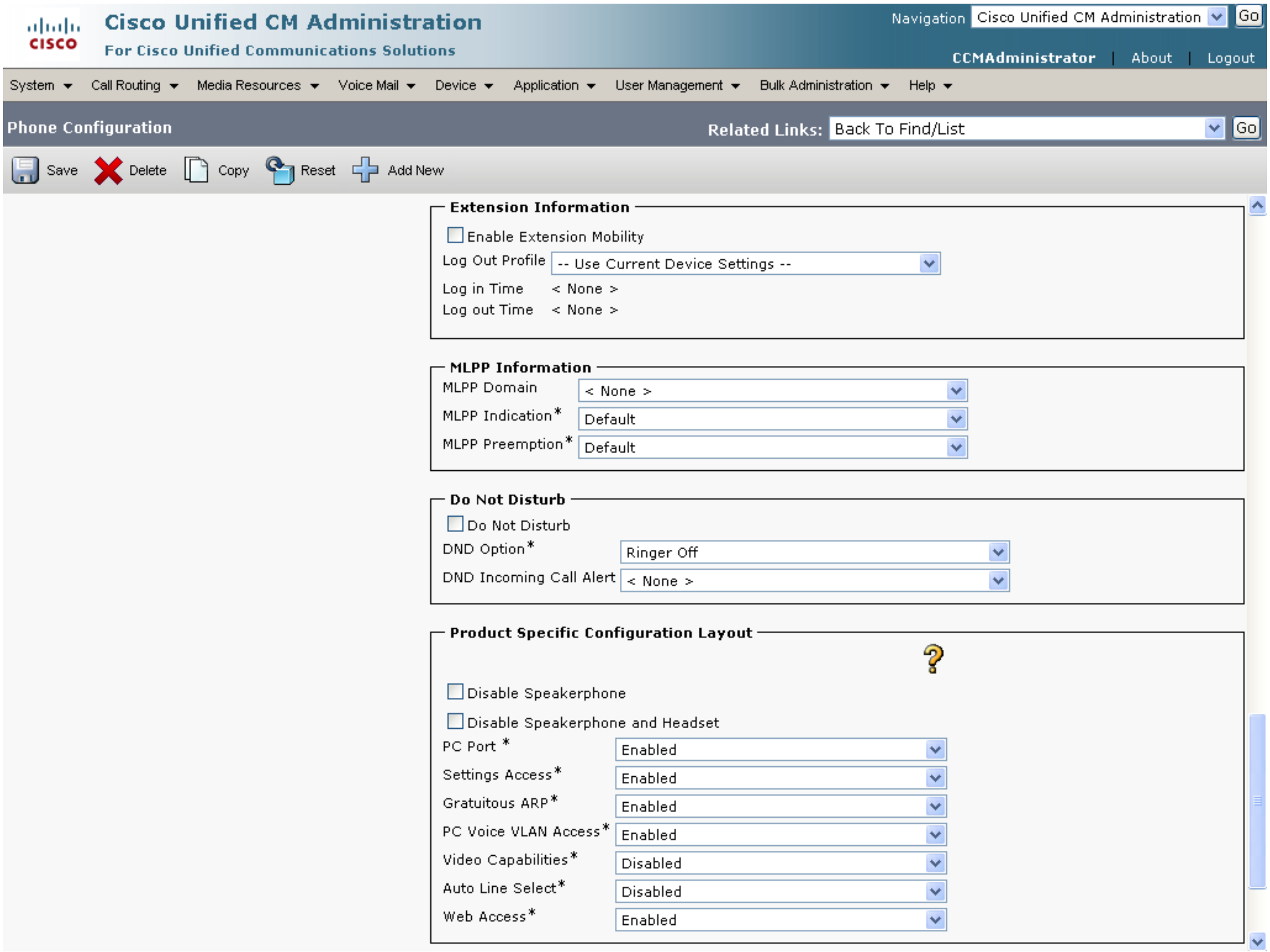
Cisco Unified IP Phone 7960, x6024, SCCP – 2 of 4.



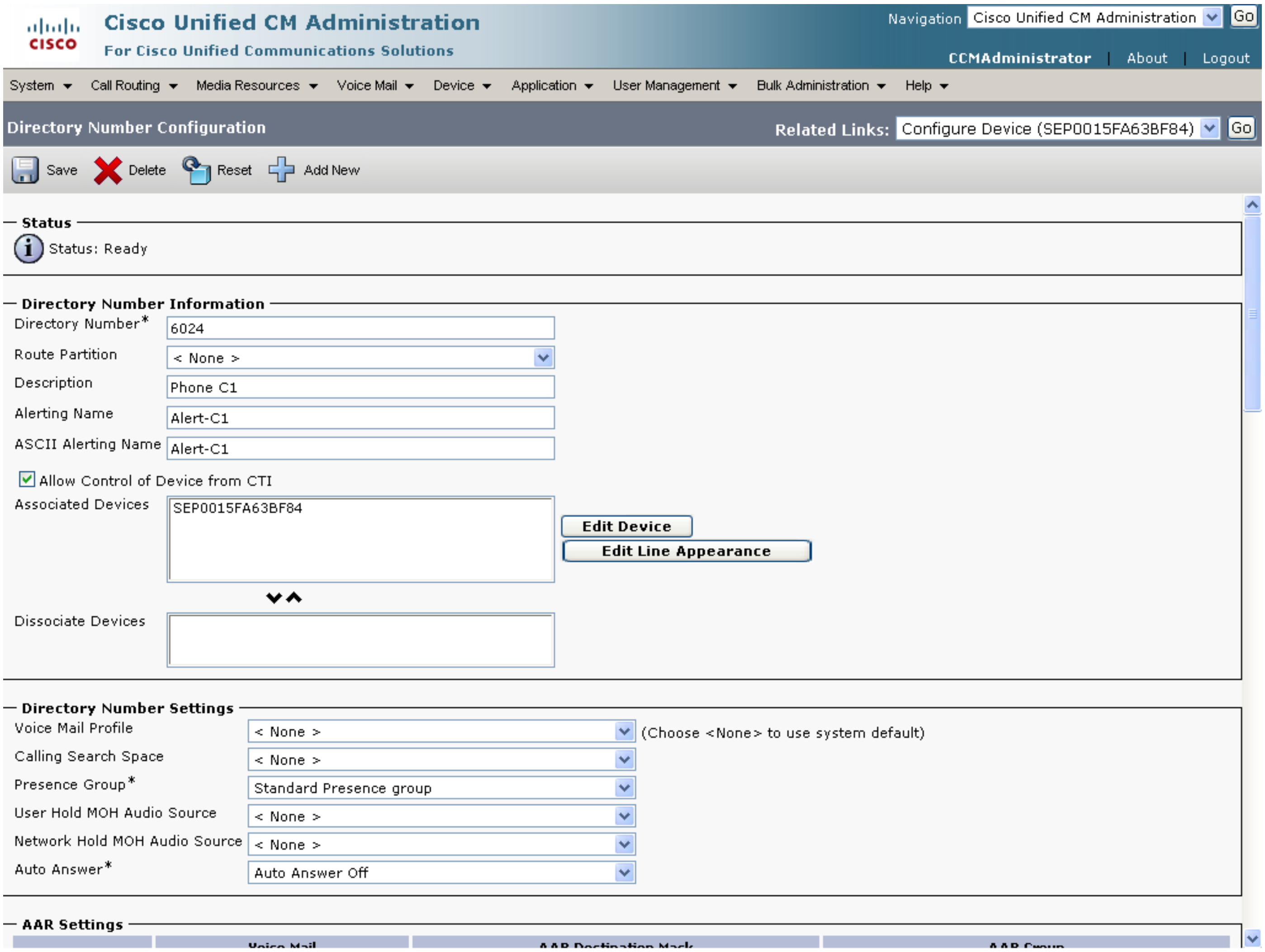
Cisco Unified IP Phone 7960, x6024, SCCP – 3 of 4.



Cisco Unified IP Phone 7960, x6024, SCCP – 4 of 4.

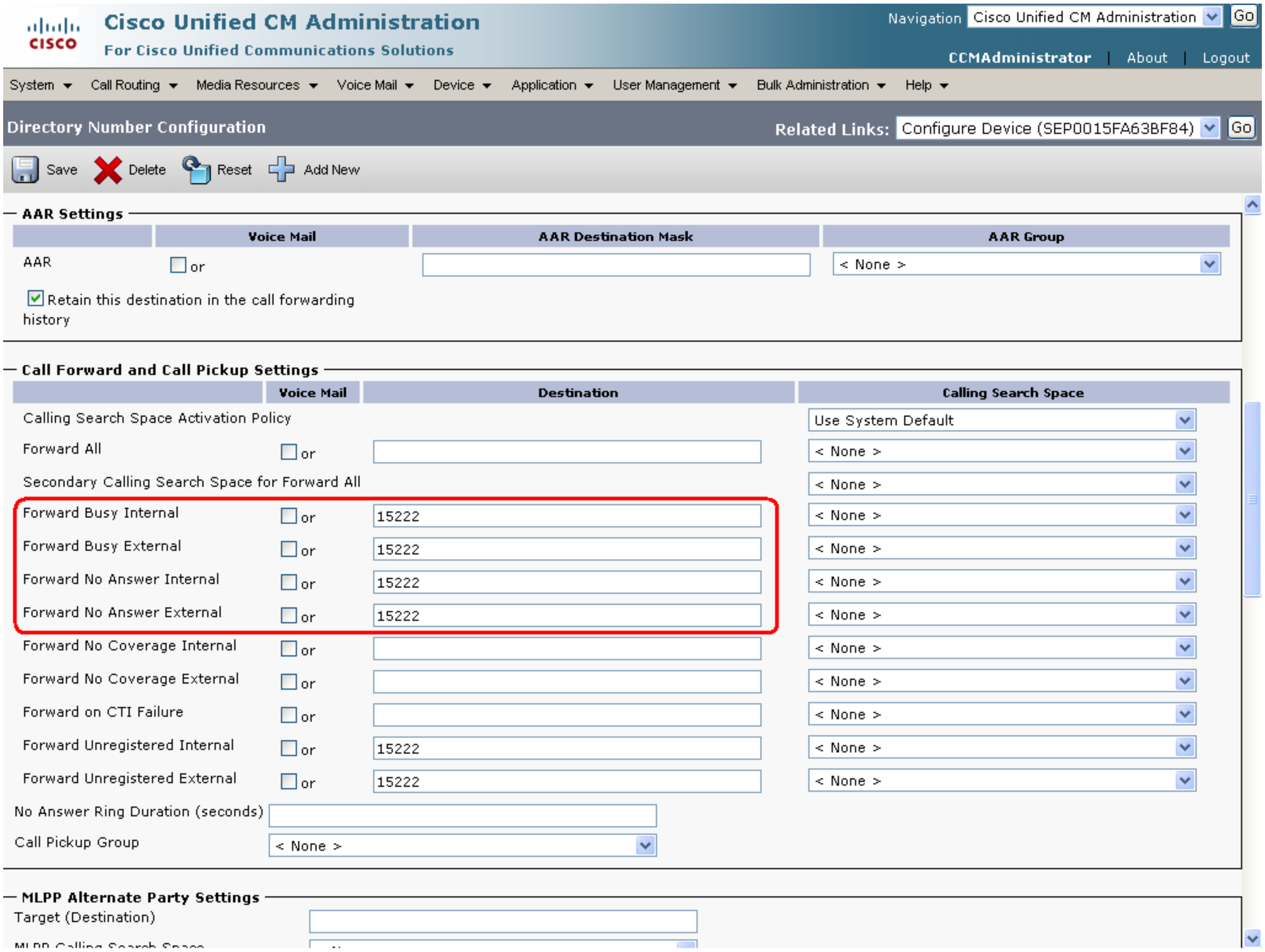


Line 6024 – 1 of 4.

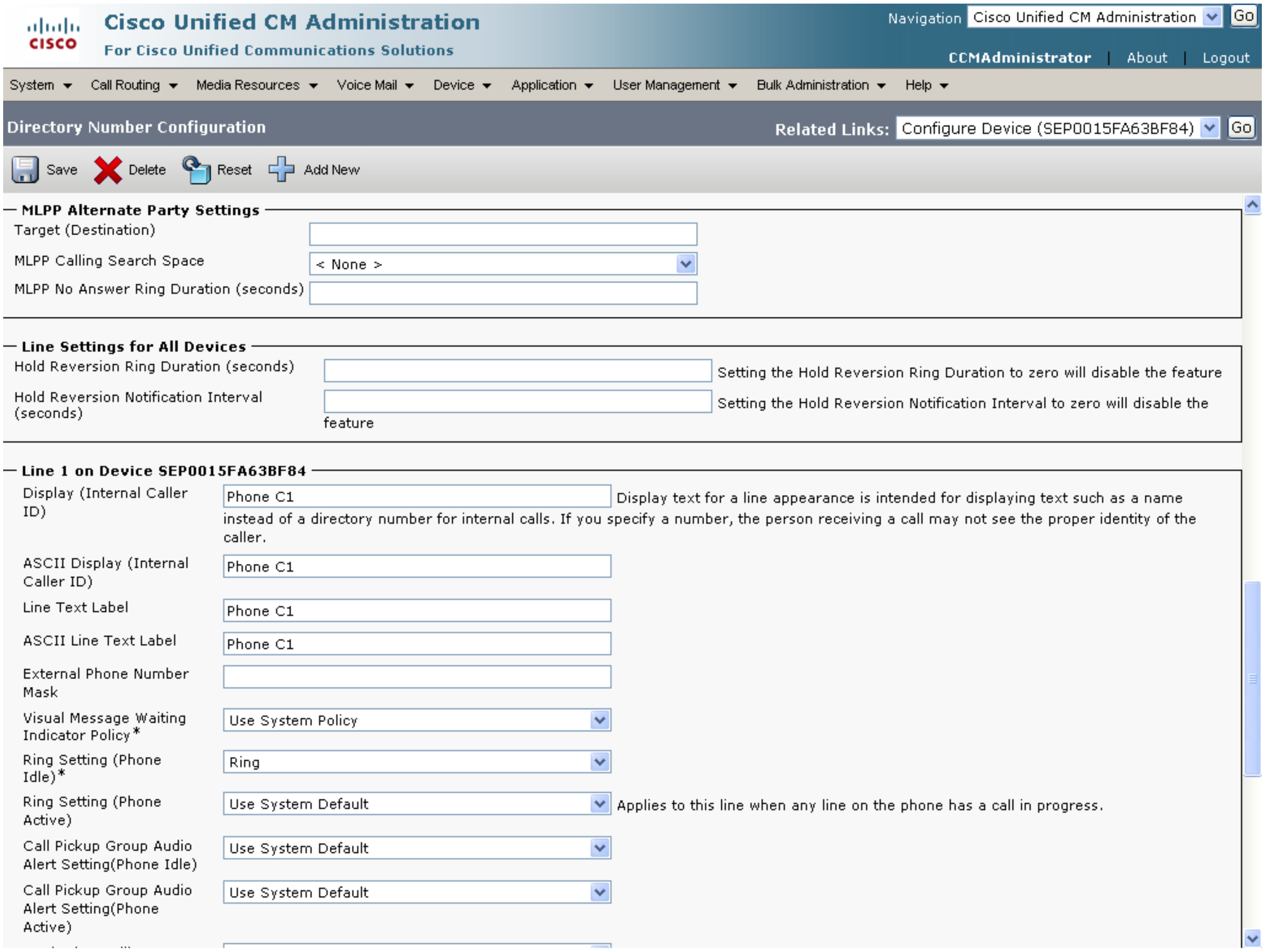


Line 6024 – 2 of 4.

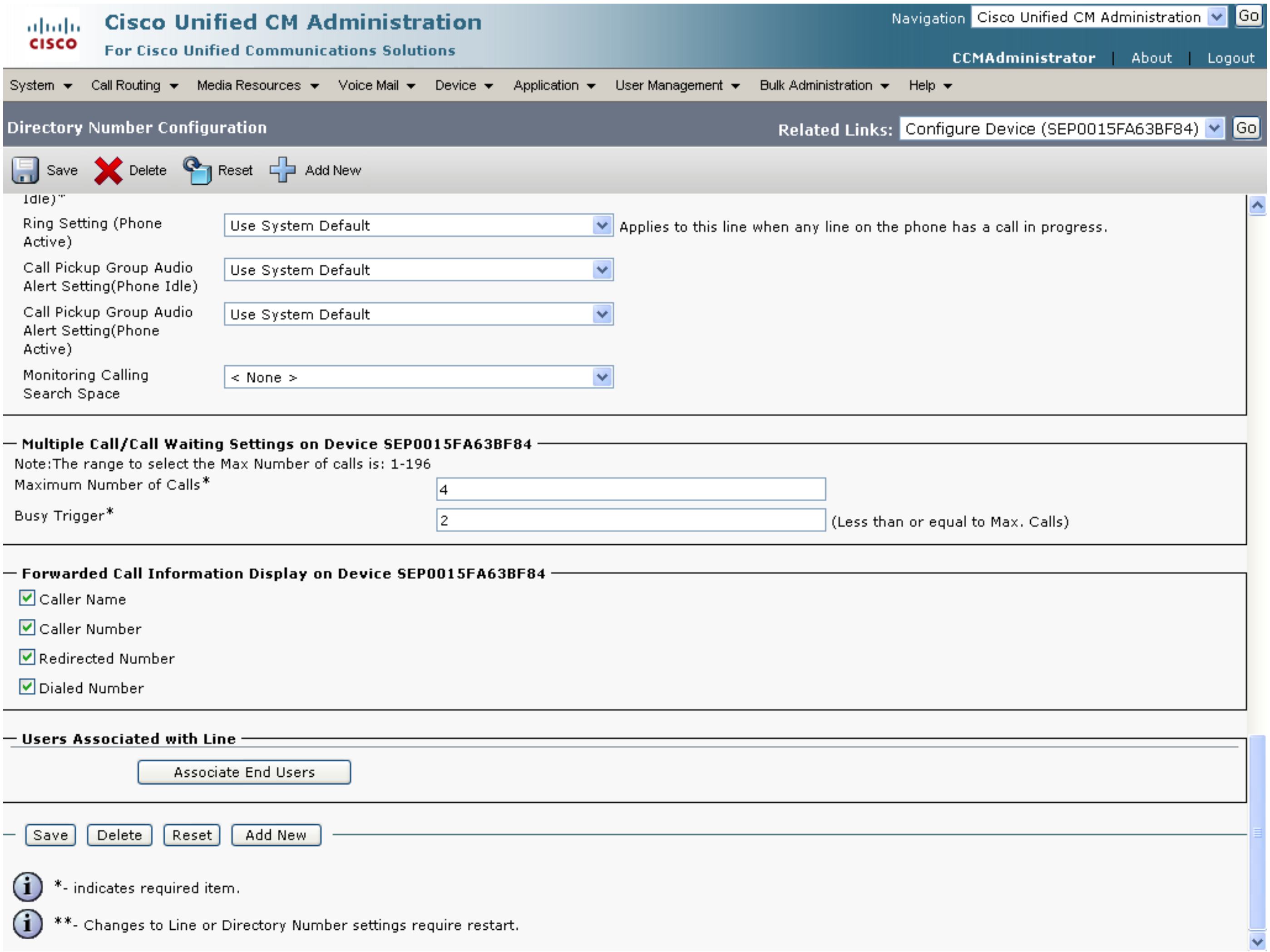
Line is currently forwarded to x15222 on Busy or No Answer.



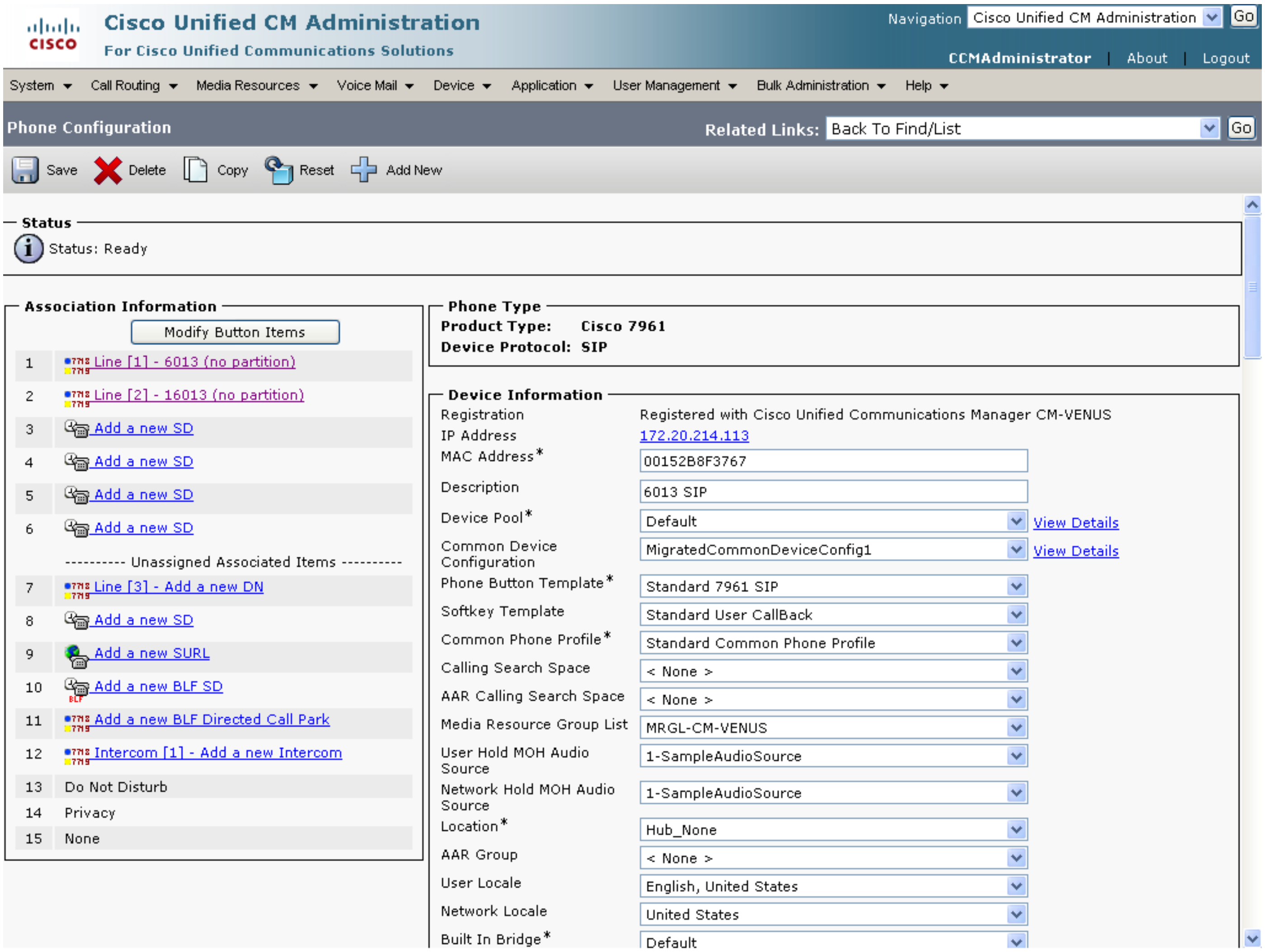
Line 6024 – 3 of 4.



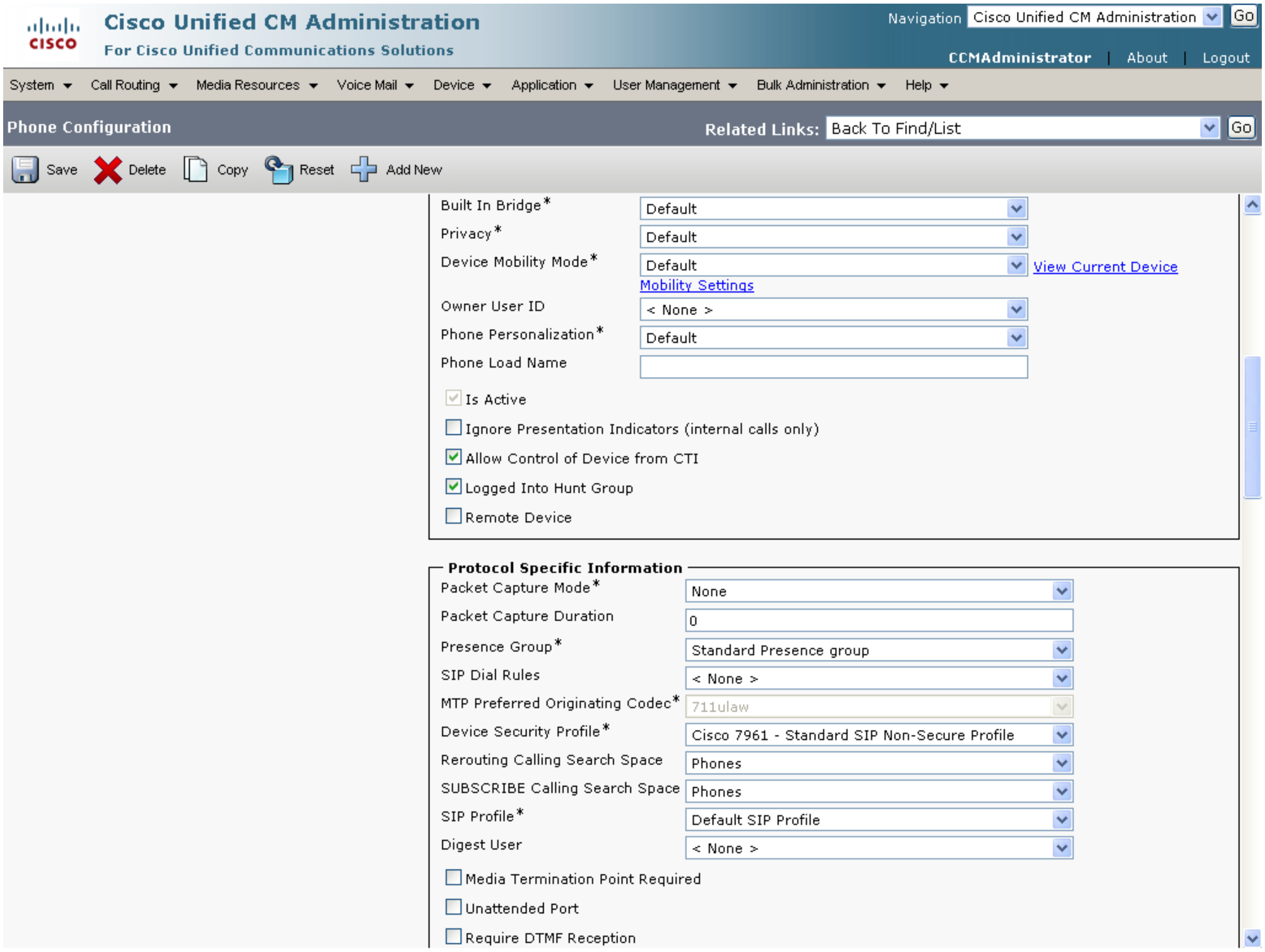
Line 6024 – 4 of 4.



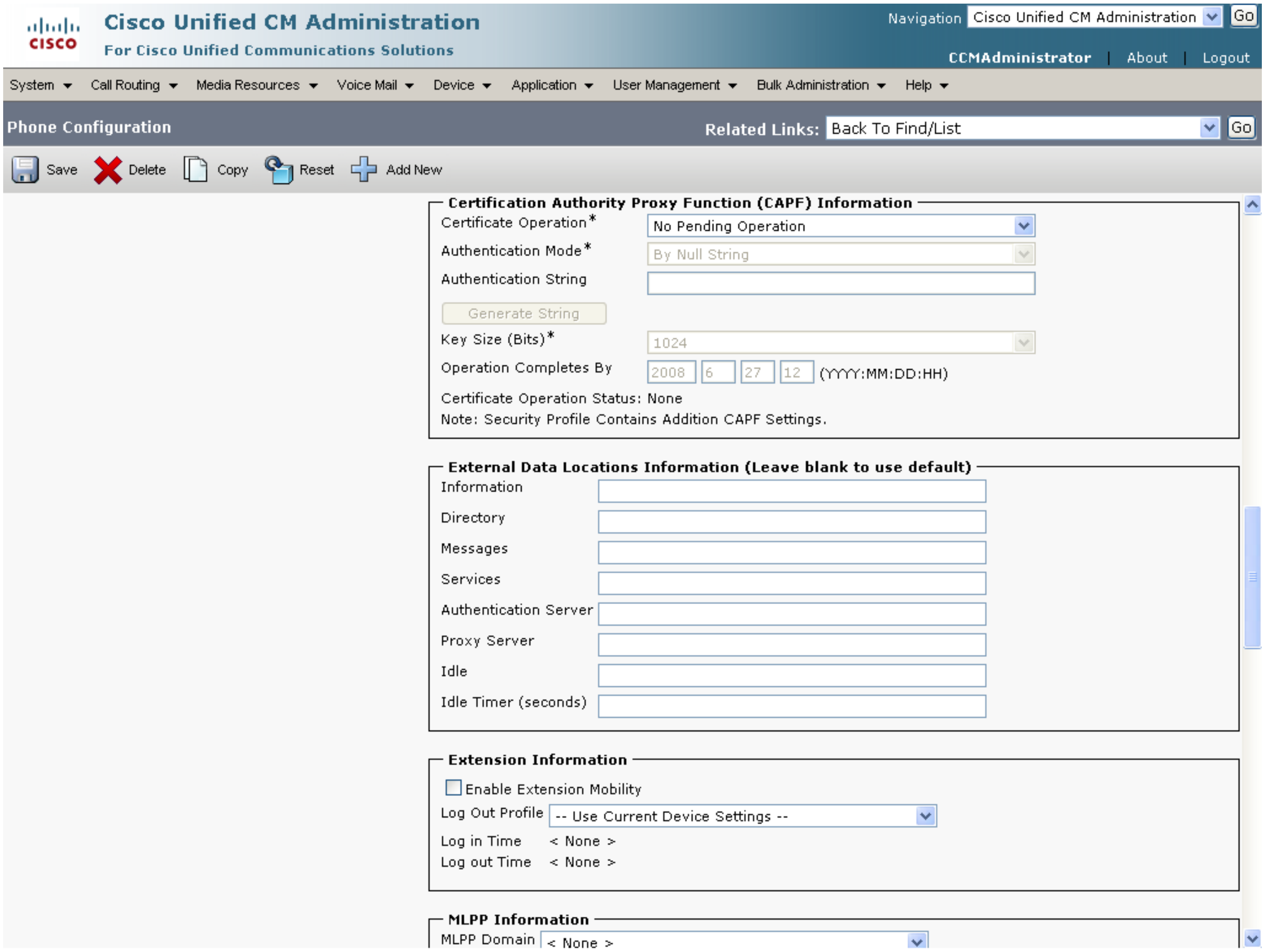
Cisco Unified IP Phone 7961, x6013, SIP – 1 of 5.



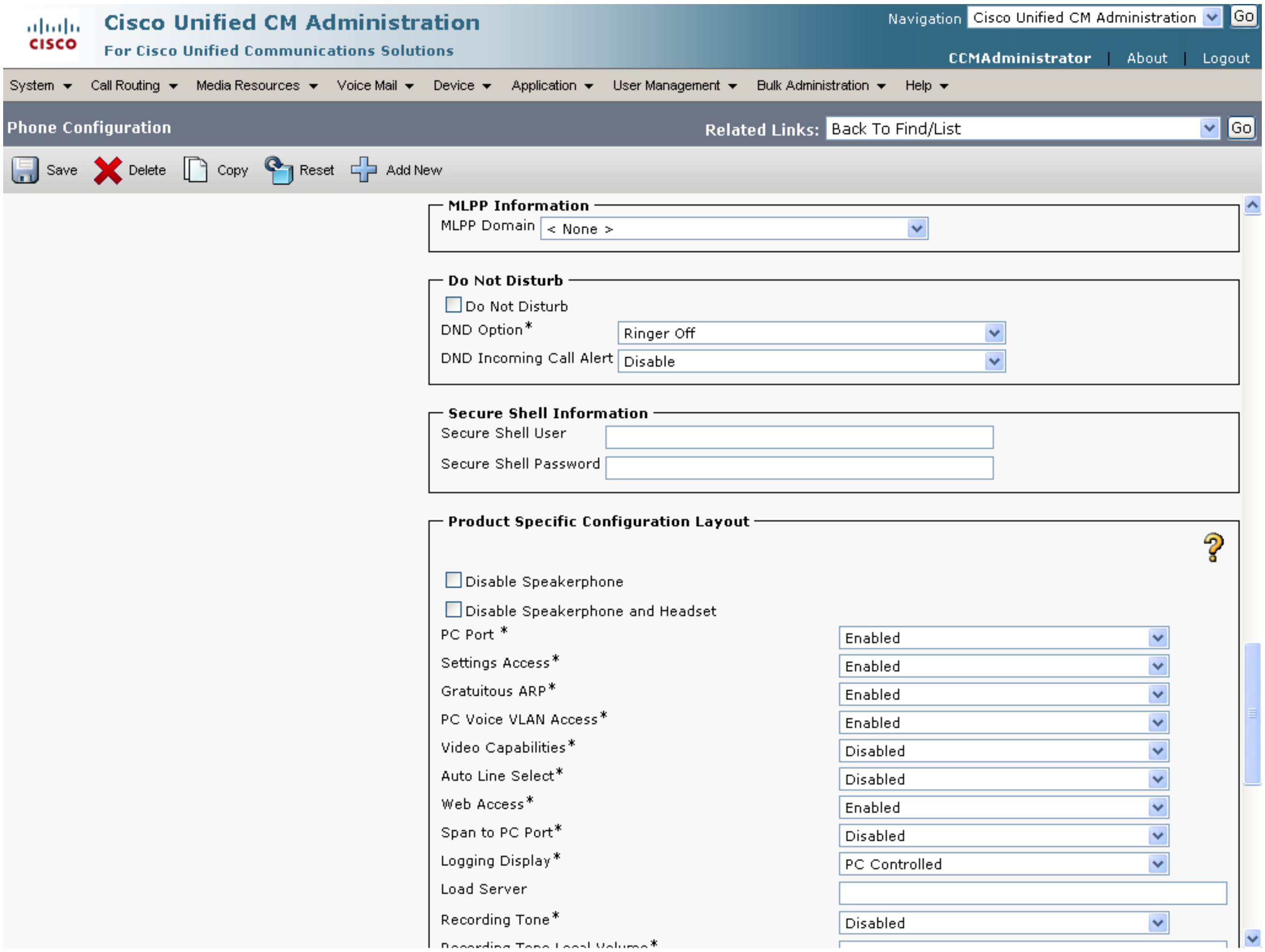
Cisco Unified IP Phone 7961, x6013, SIP – 2 of 5.



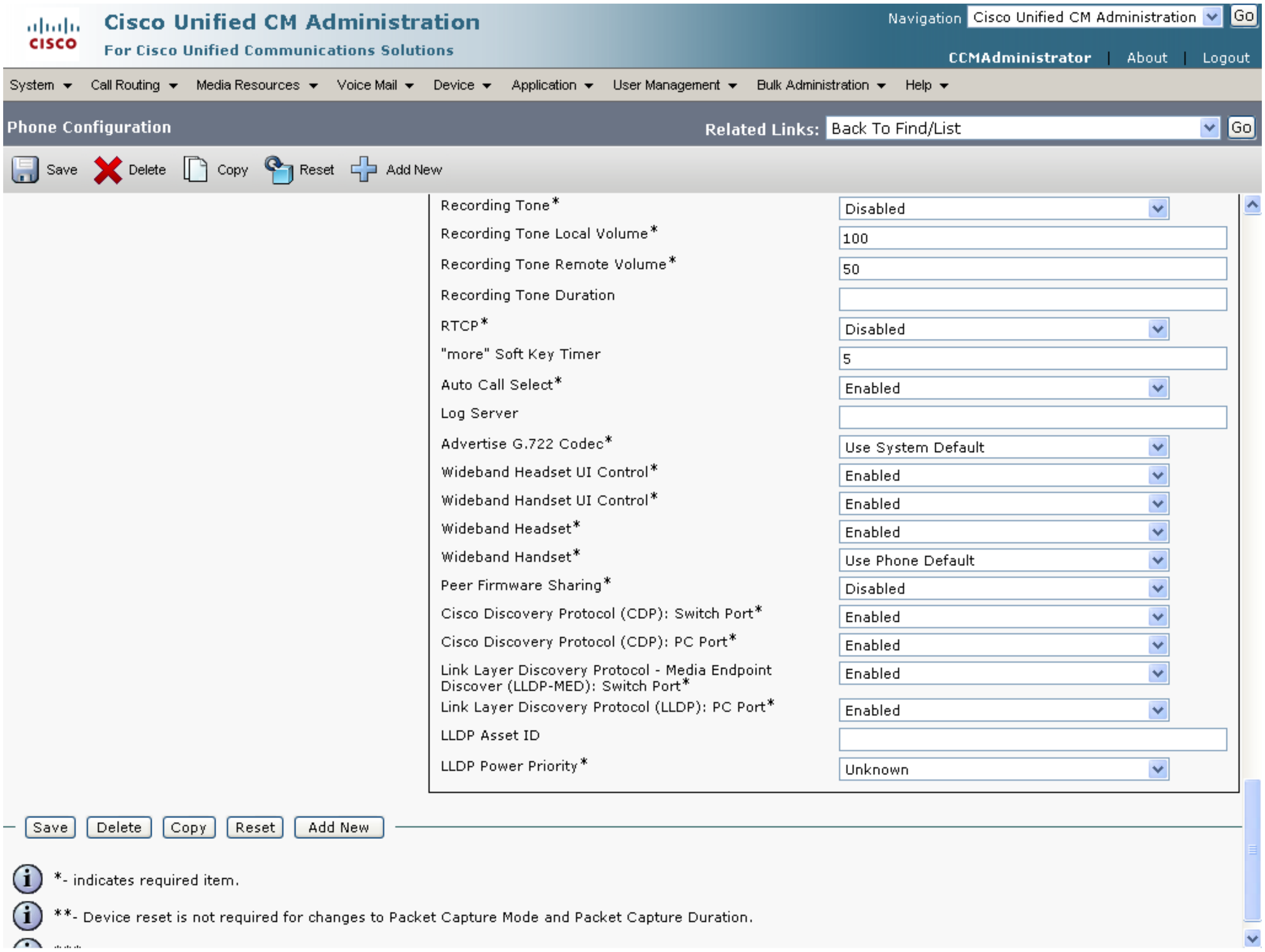
Cisco Unified IP Phone 7961, x6013, SIP – 3 of 5.



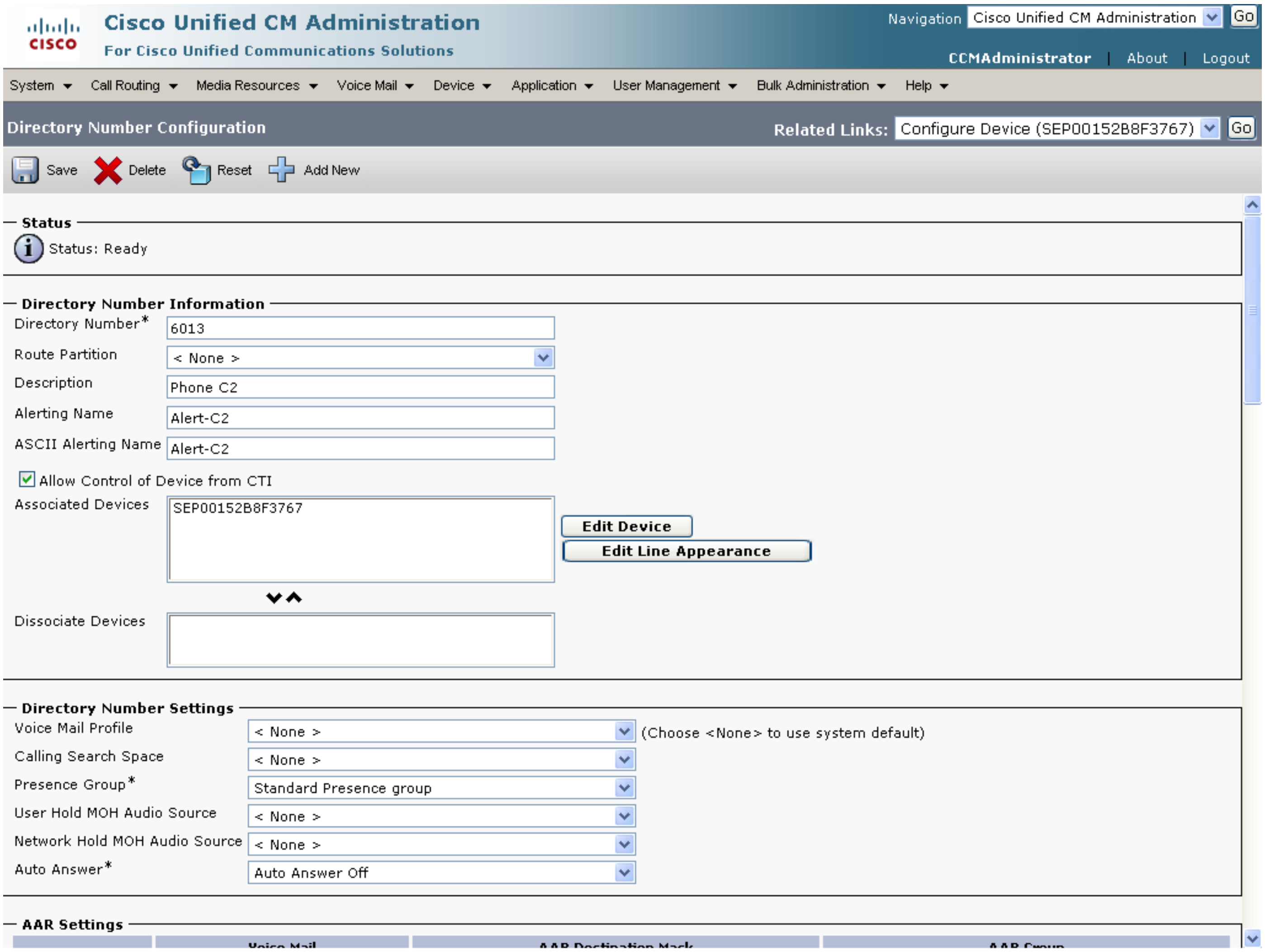
Cisco Unified IP Phone 7961, x6013, SIP – 4 of 5.



Cisco Unified IP Phone 7961, x6013, SIP – 5 of 5.

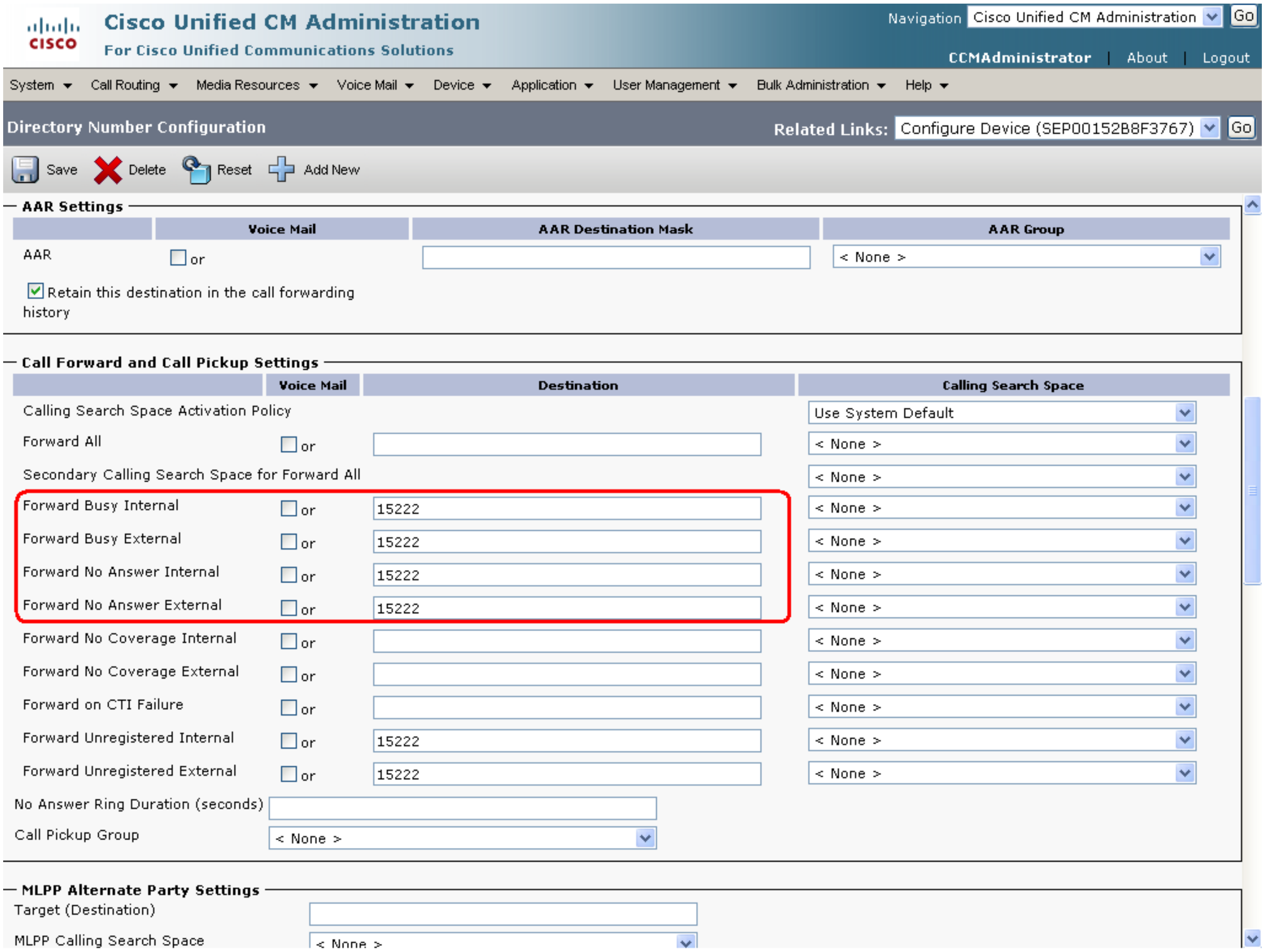


Line 6013 – 1 of 4.

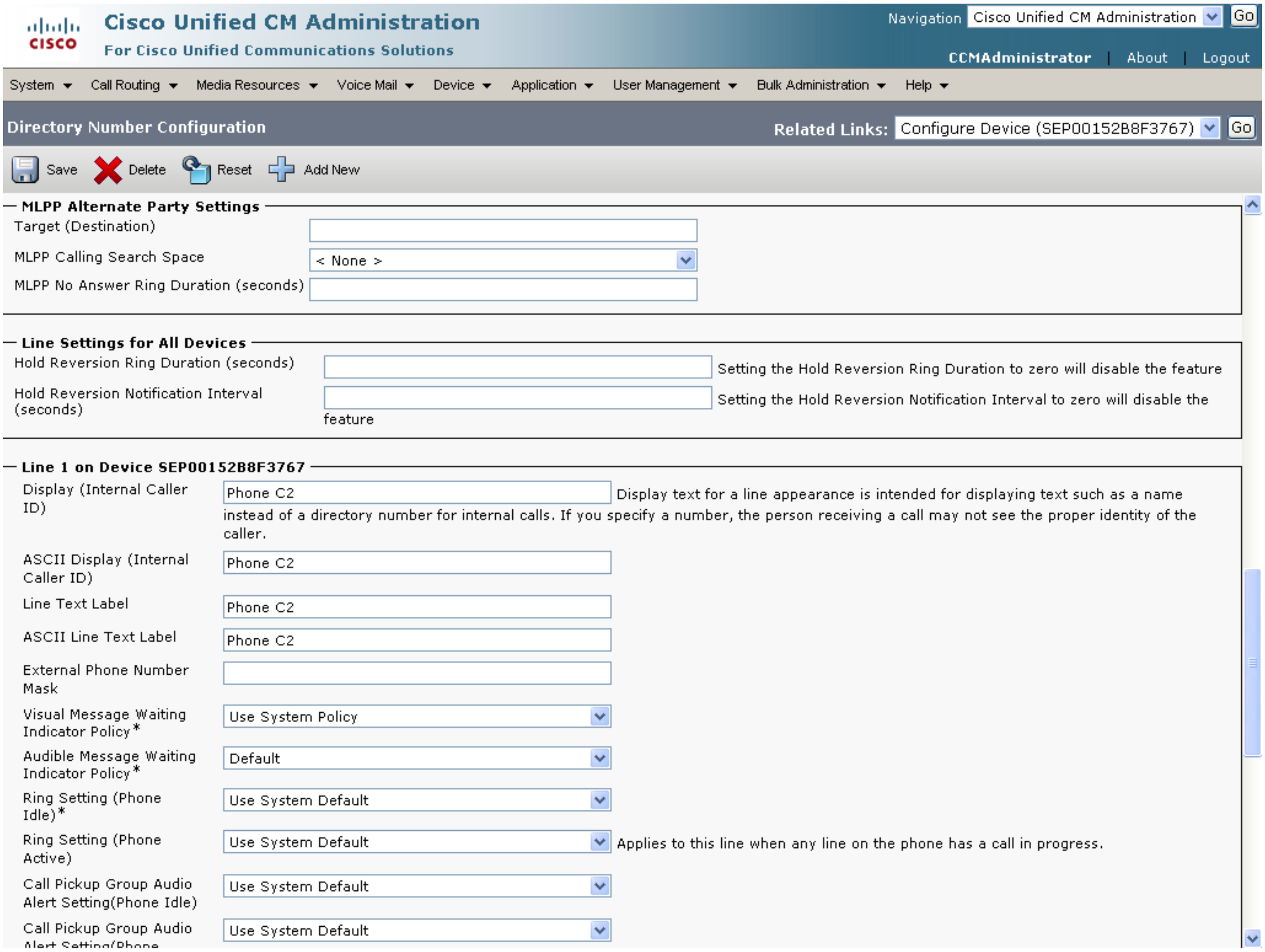


Line 6013 – 2 of 4.

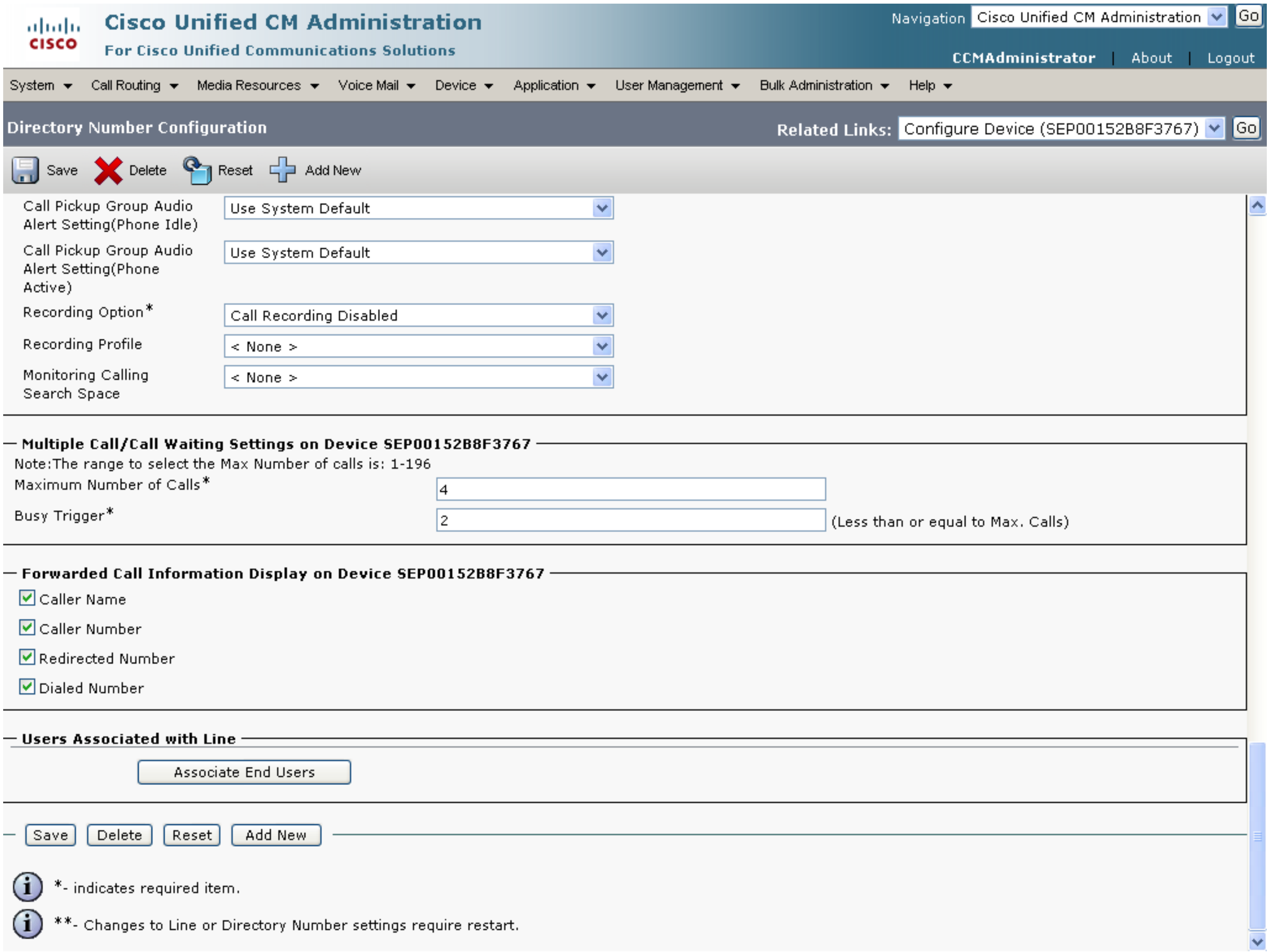
Line is currently forwarded to x15222 on Busy or No Answer.



Line 6013 – 3 of 4.

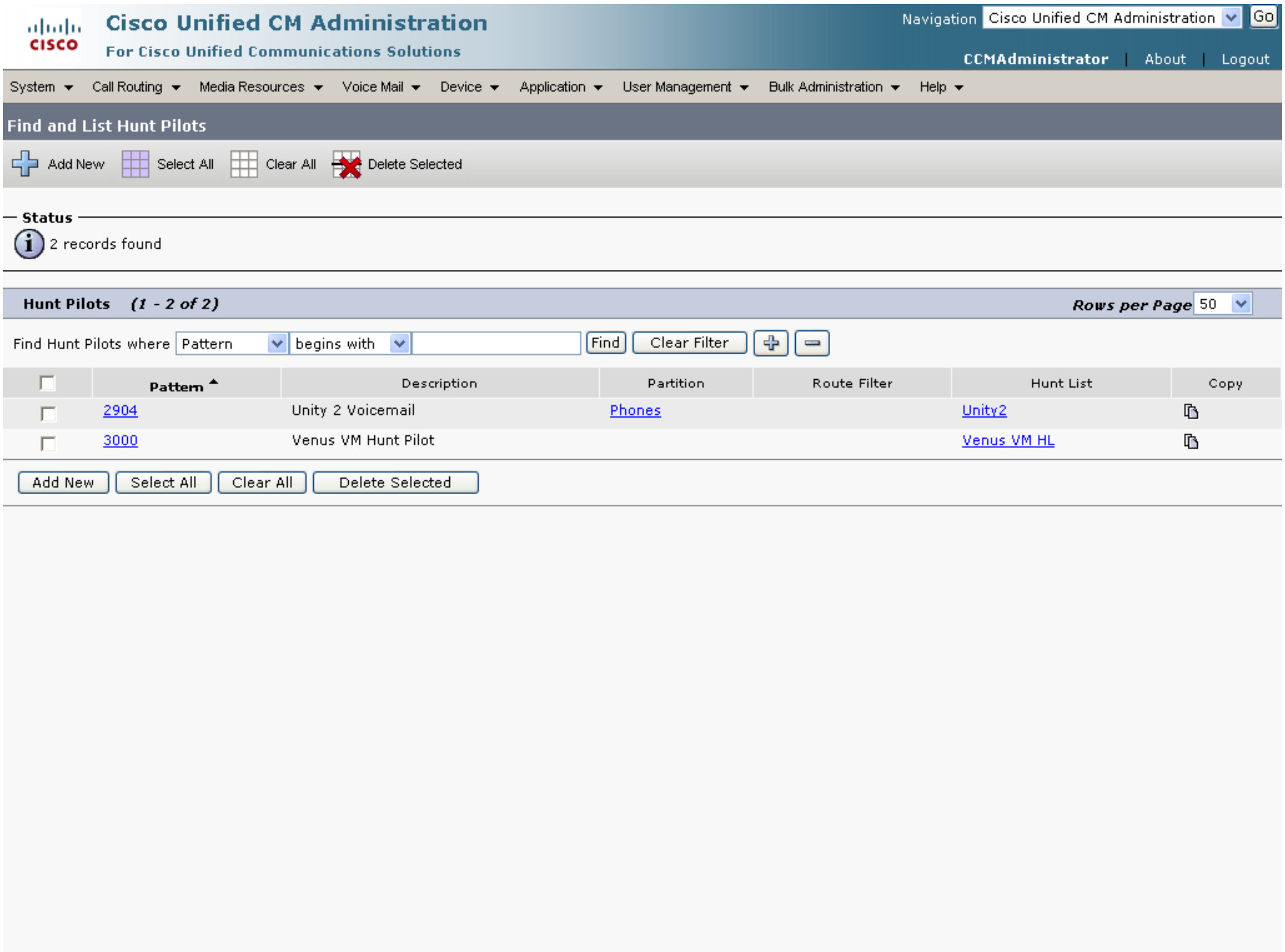


Line 6013 – 4 of 4.

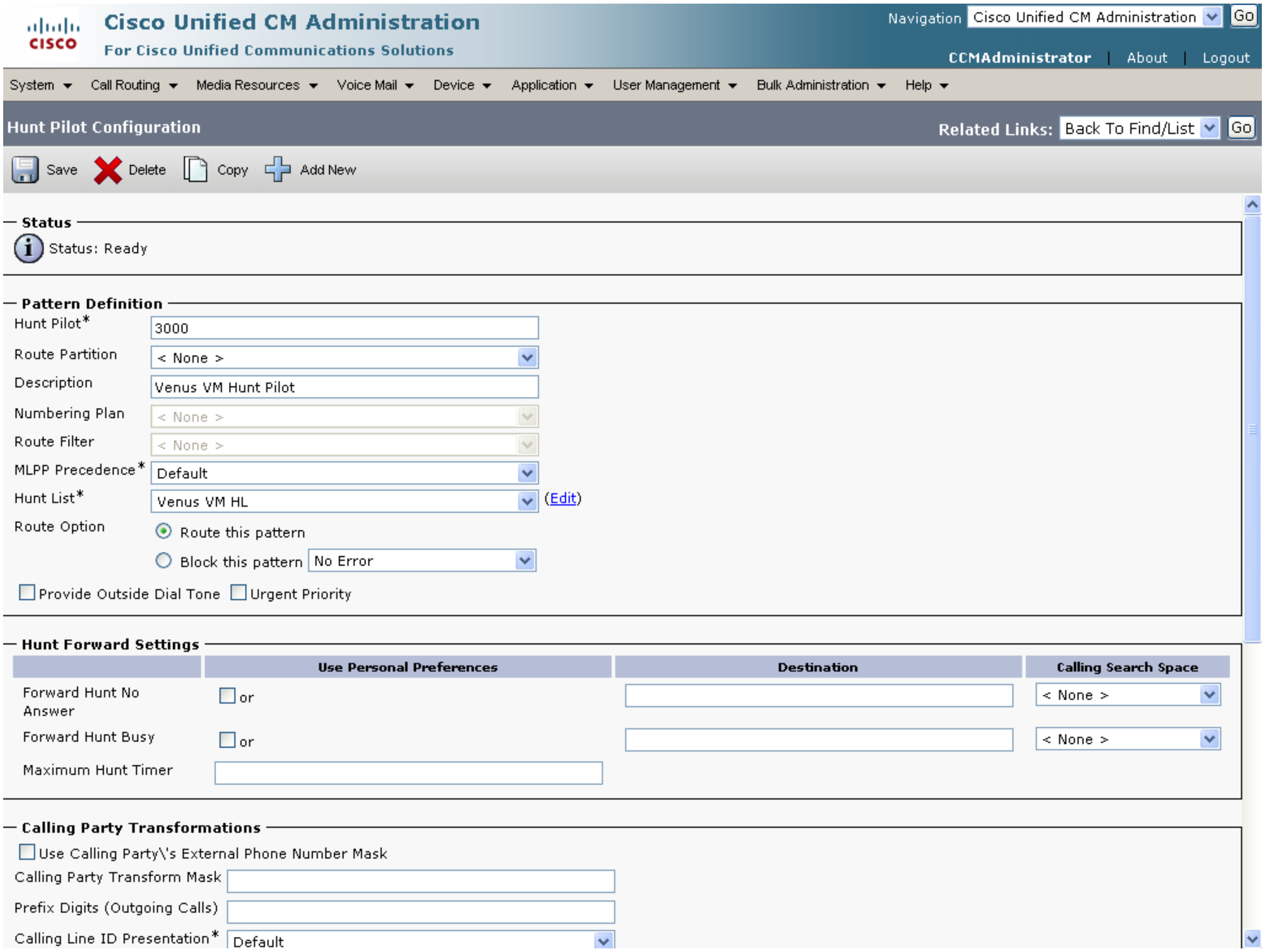


Voice Mail Integration (with Cisco Unity)

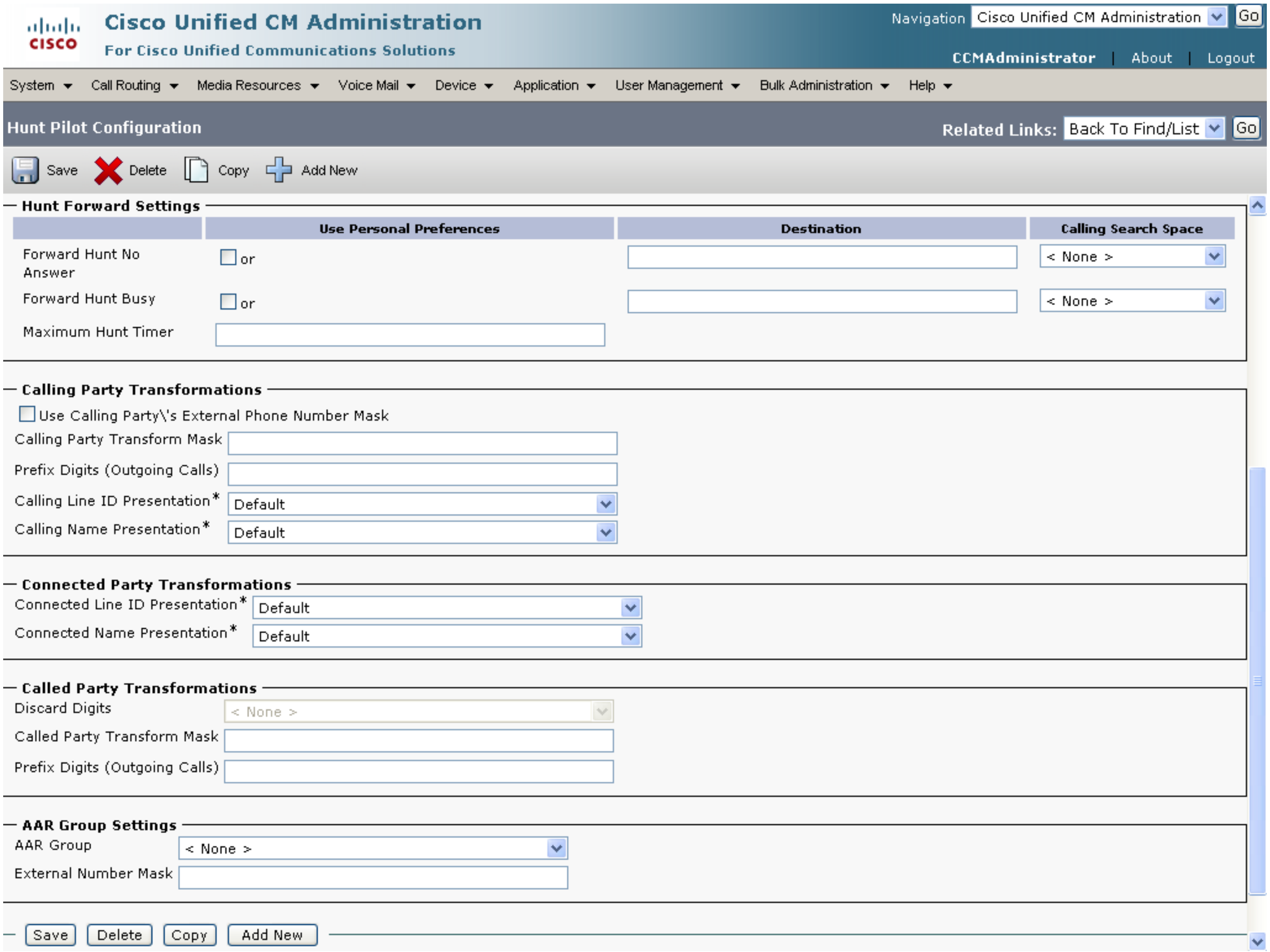
List Hunt Pilots – 1 of 1.



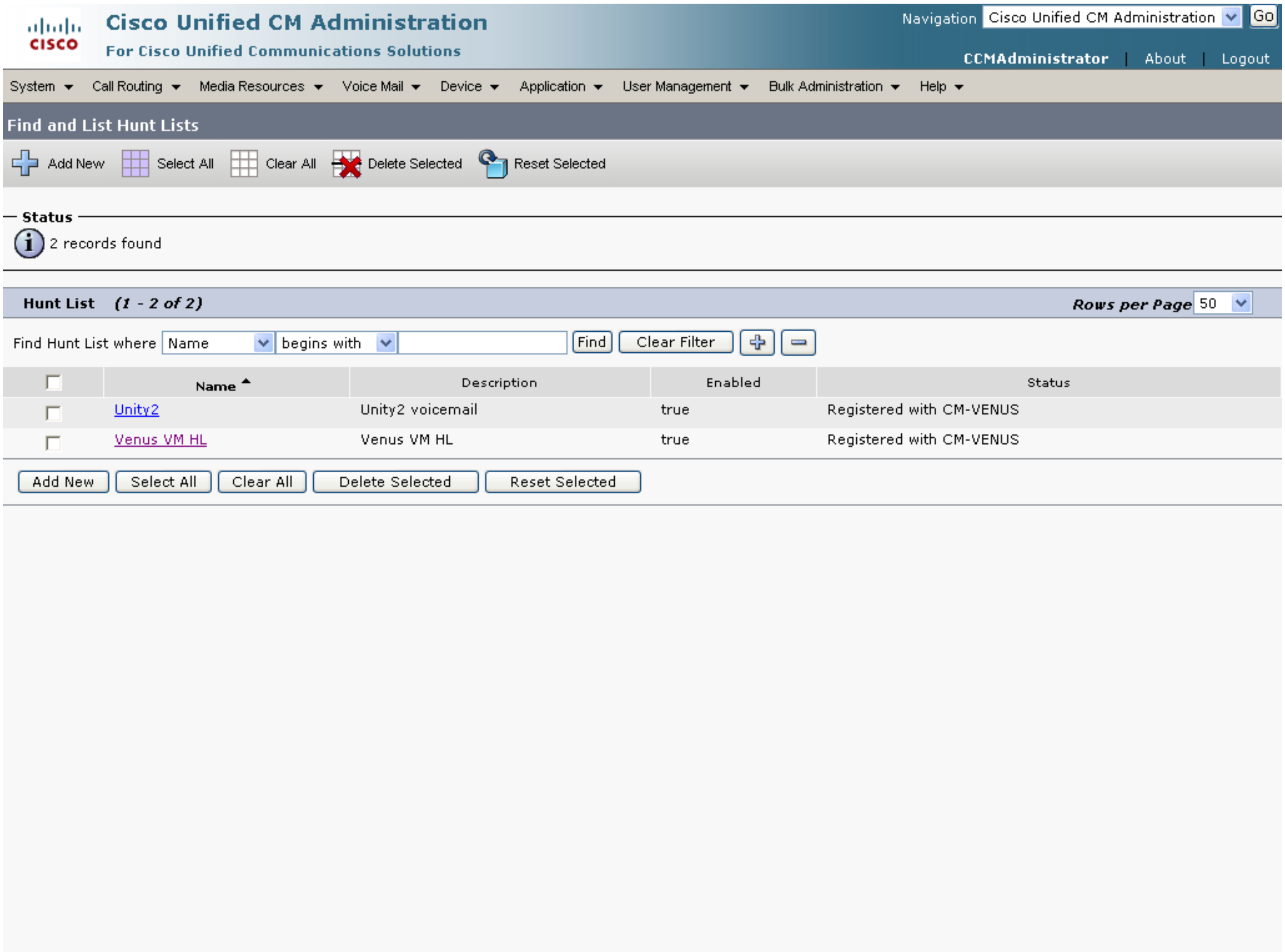
Hunt Pilot 3000 – 1 of 2.



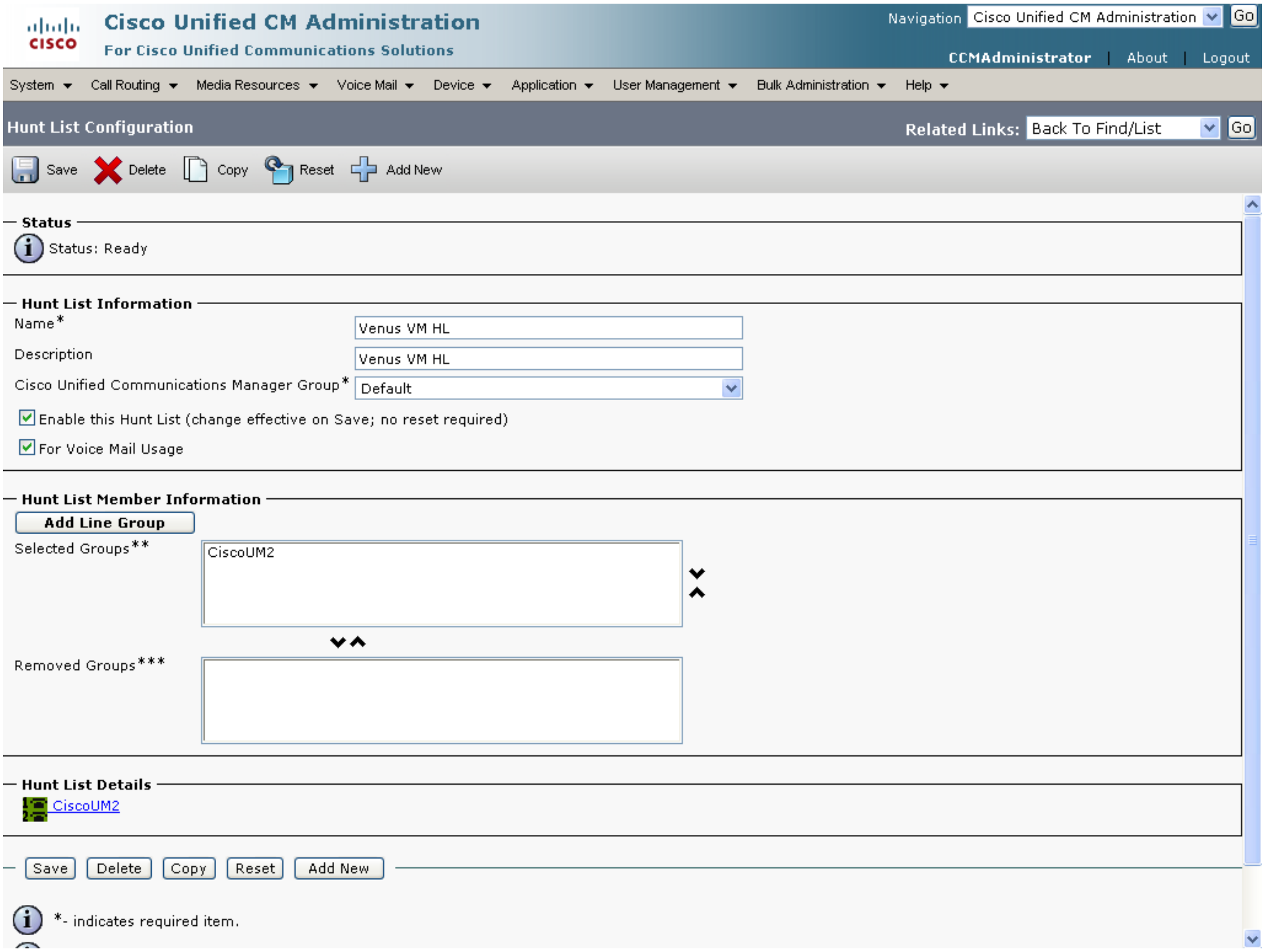
Hunt Pilot 3000 – 2 of 2.



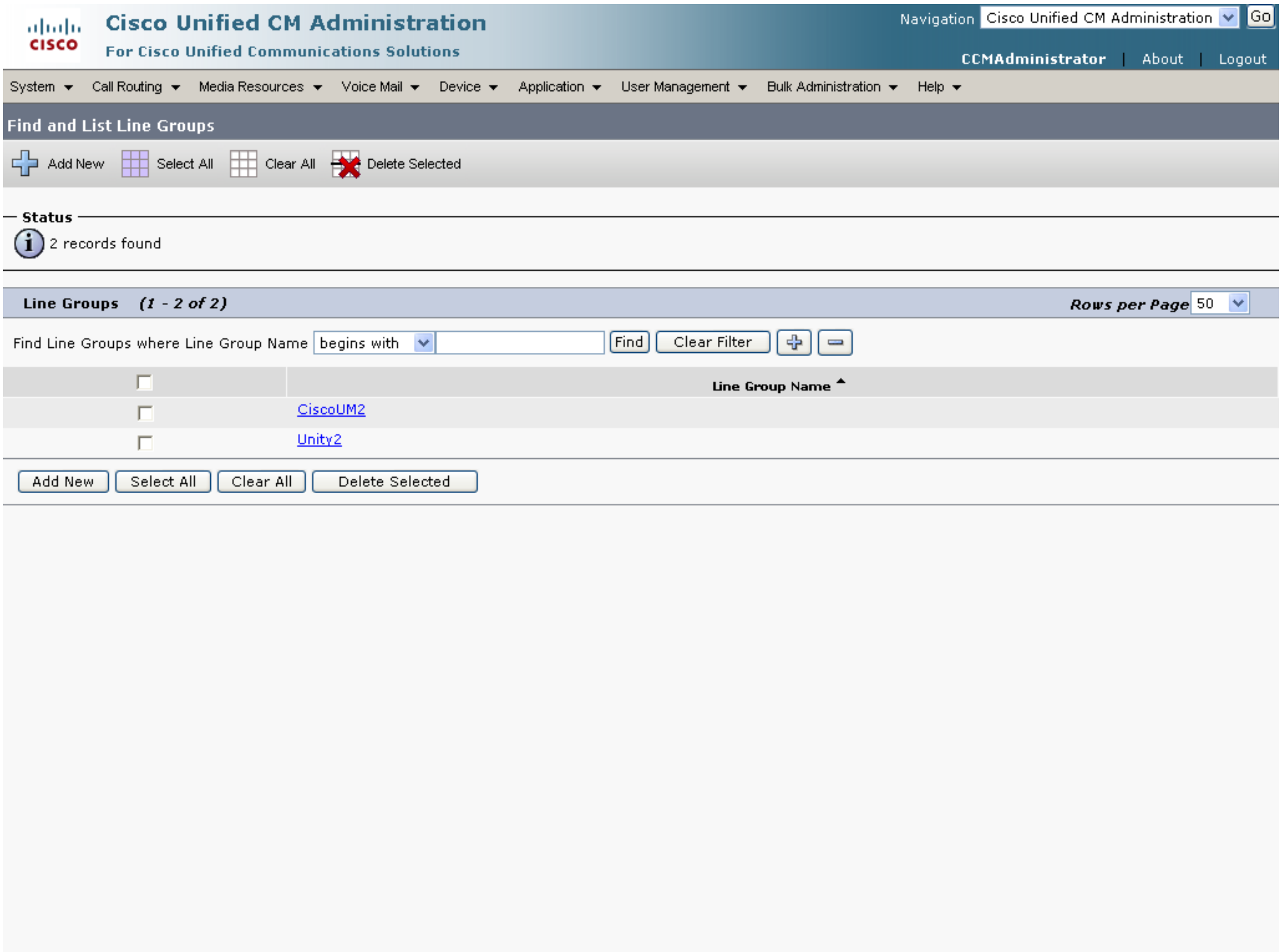
Hunt Lists – 1 of 1.



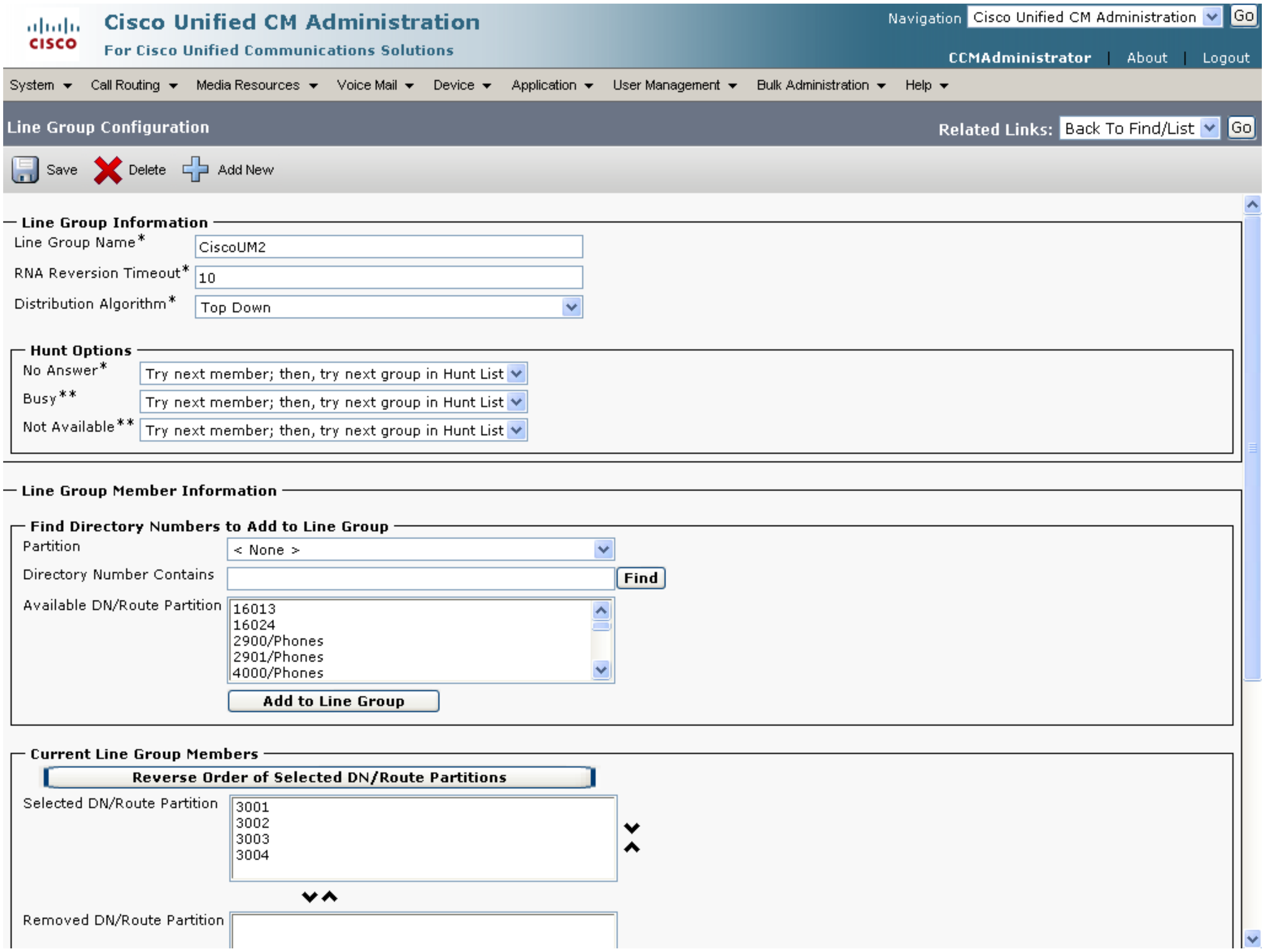
Hunt List for Cisco Unity – 1 of 1.



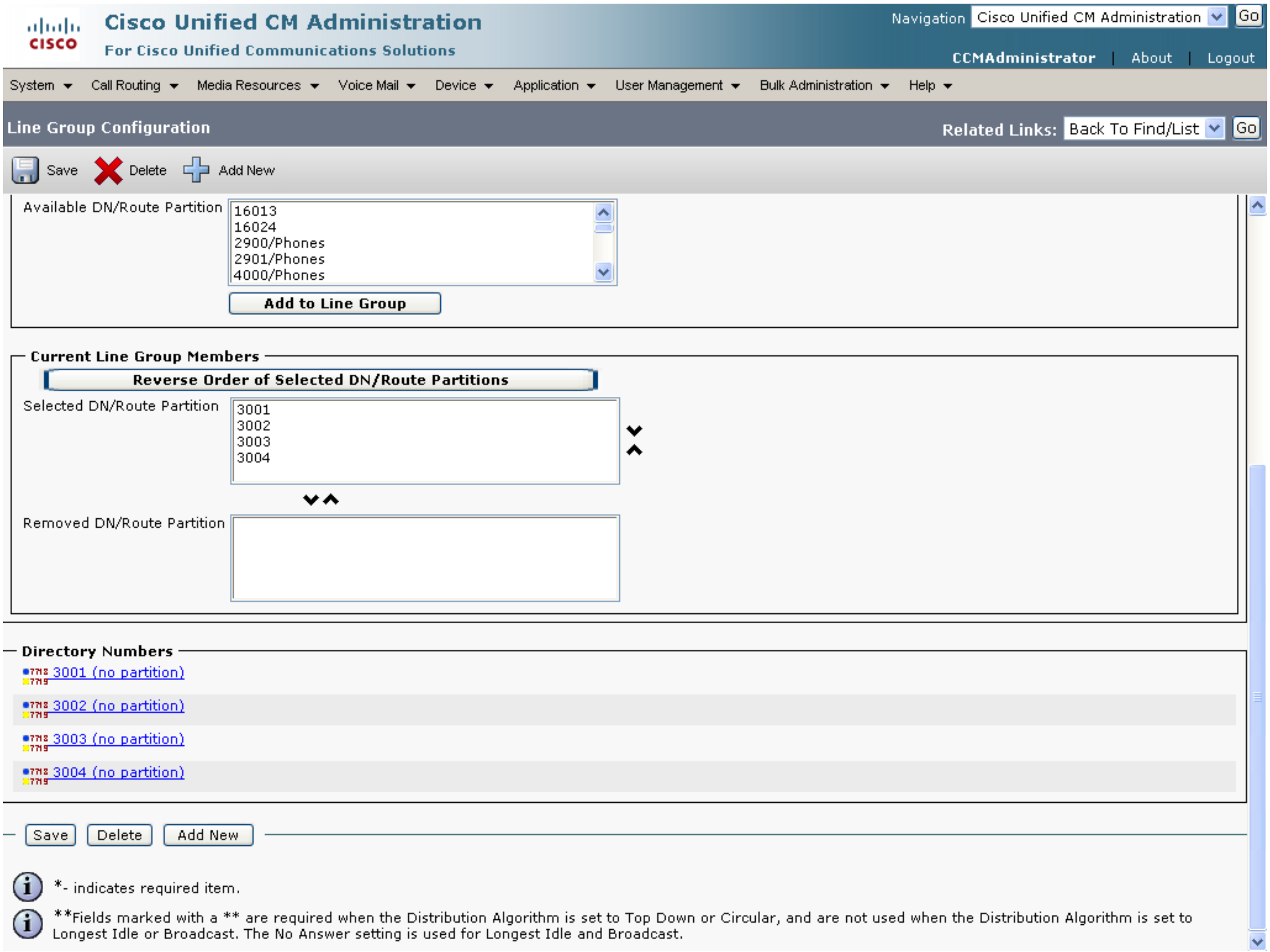
Line Groups – 1 of 1.



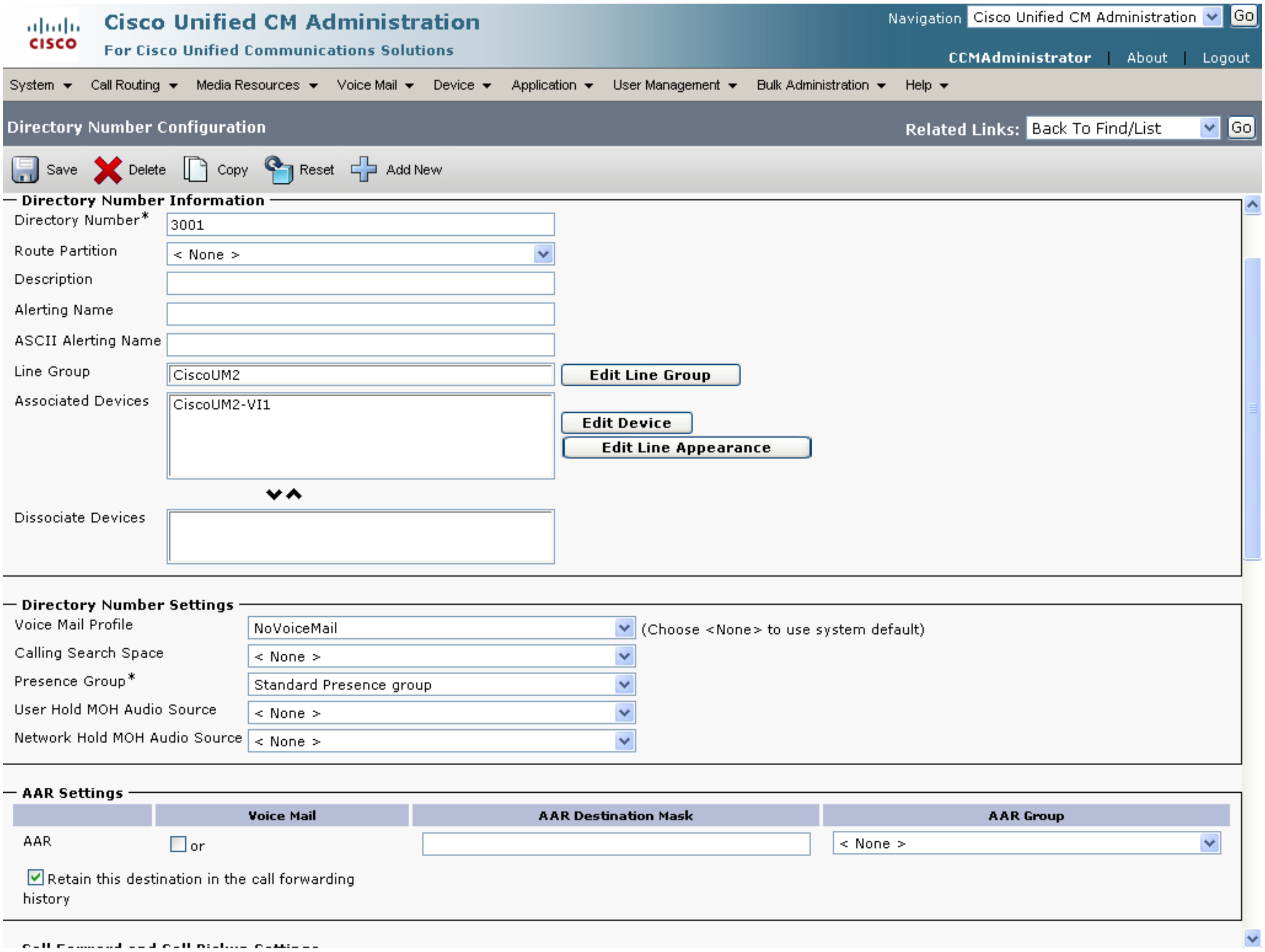
Line Group for Cisco Unity – 1 of 2.



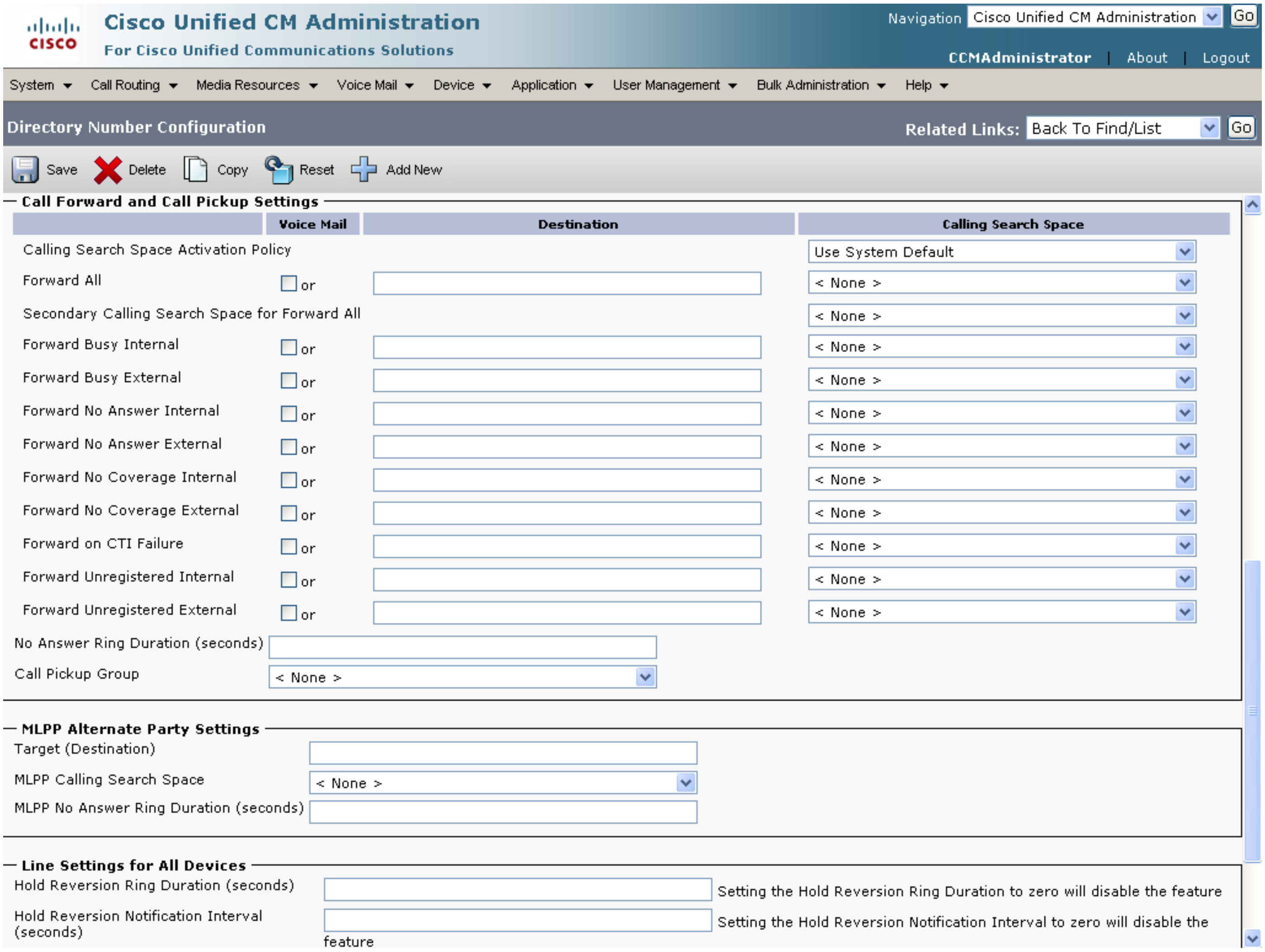
Line Group for Cisco Unity – 2 of 2.



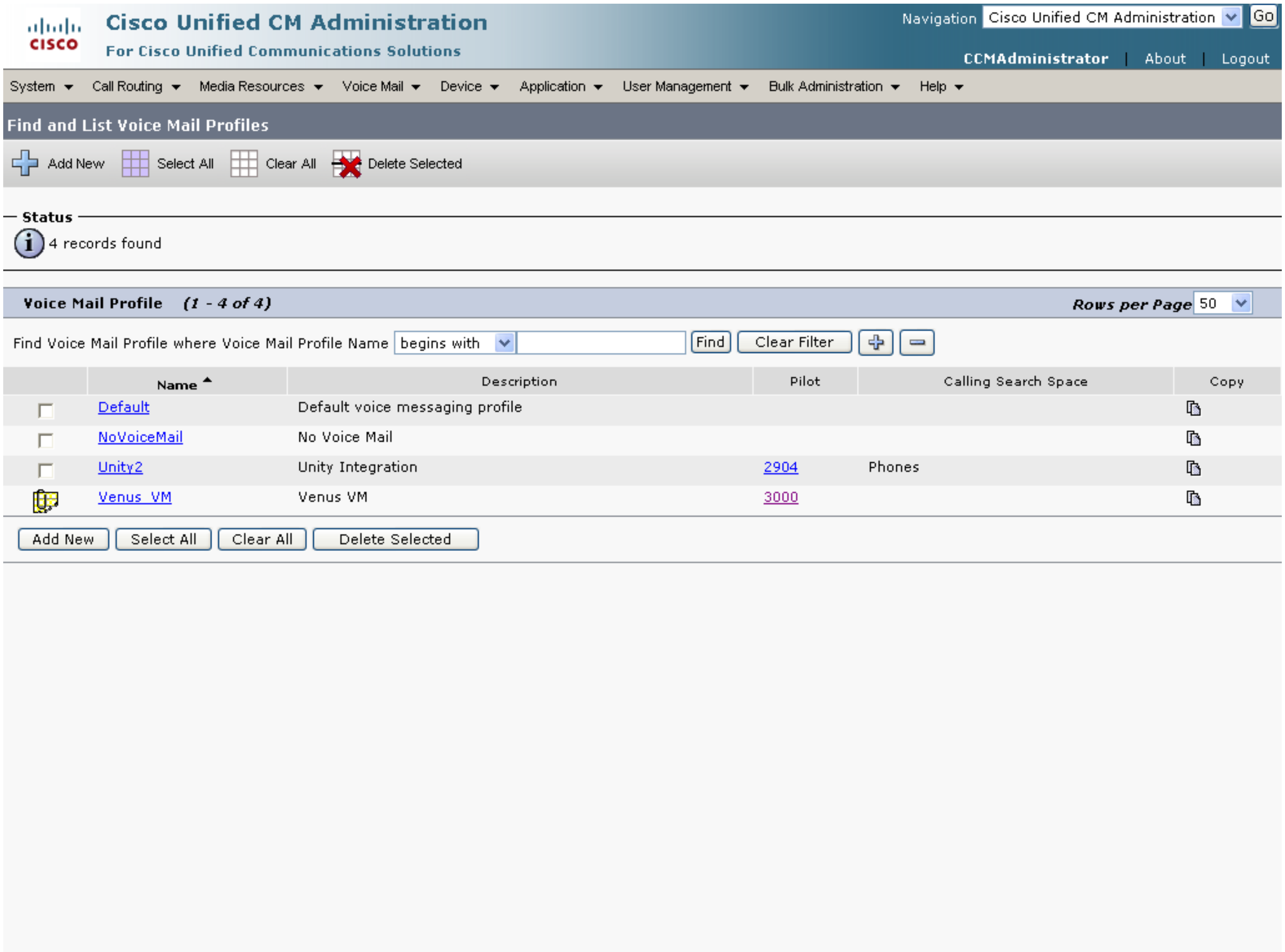
Line Group Member 3001 – 1 of 2.



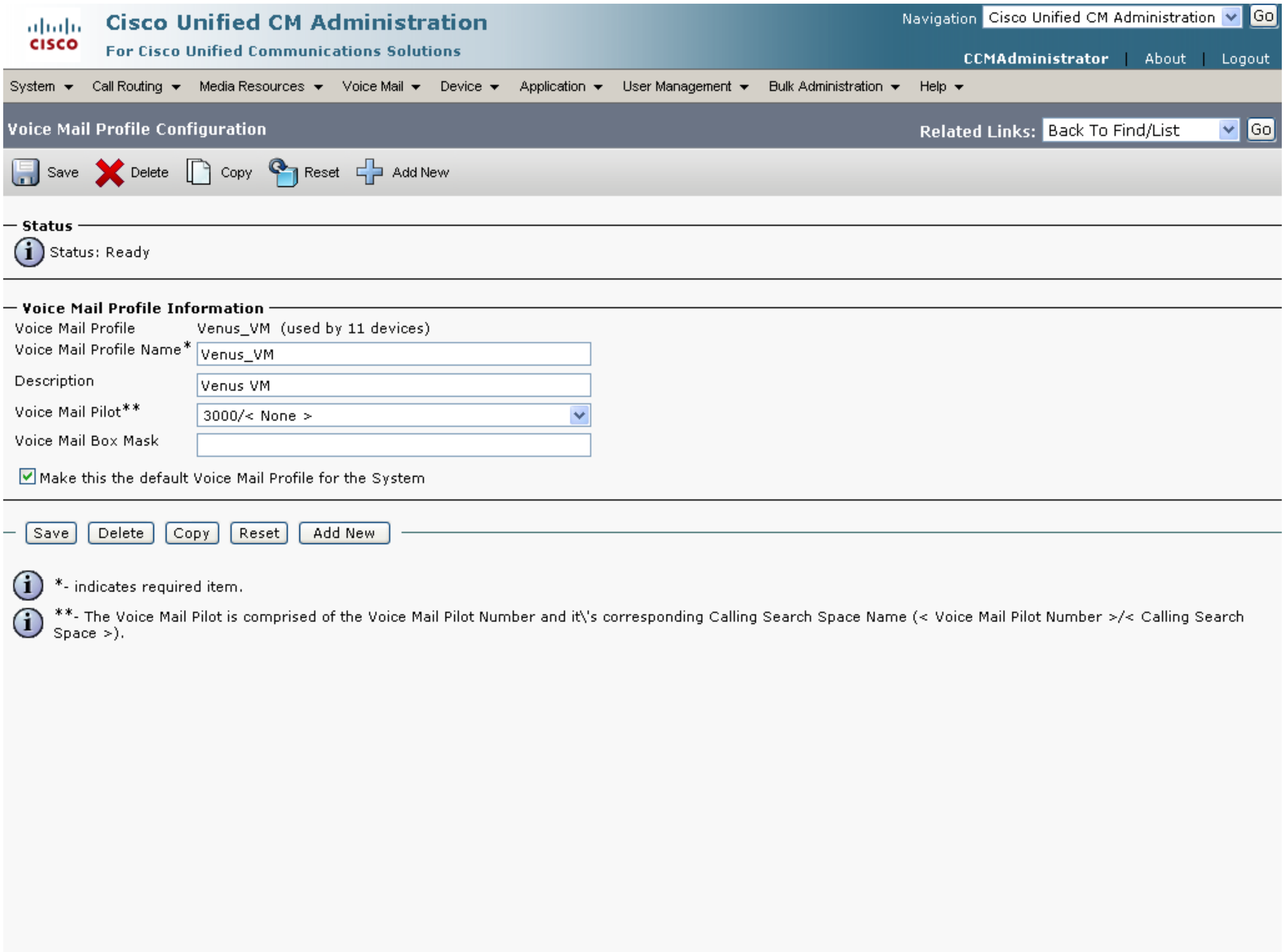
Line Group Member 3001 – 2 of 2.



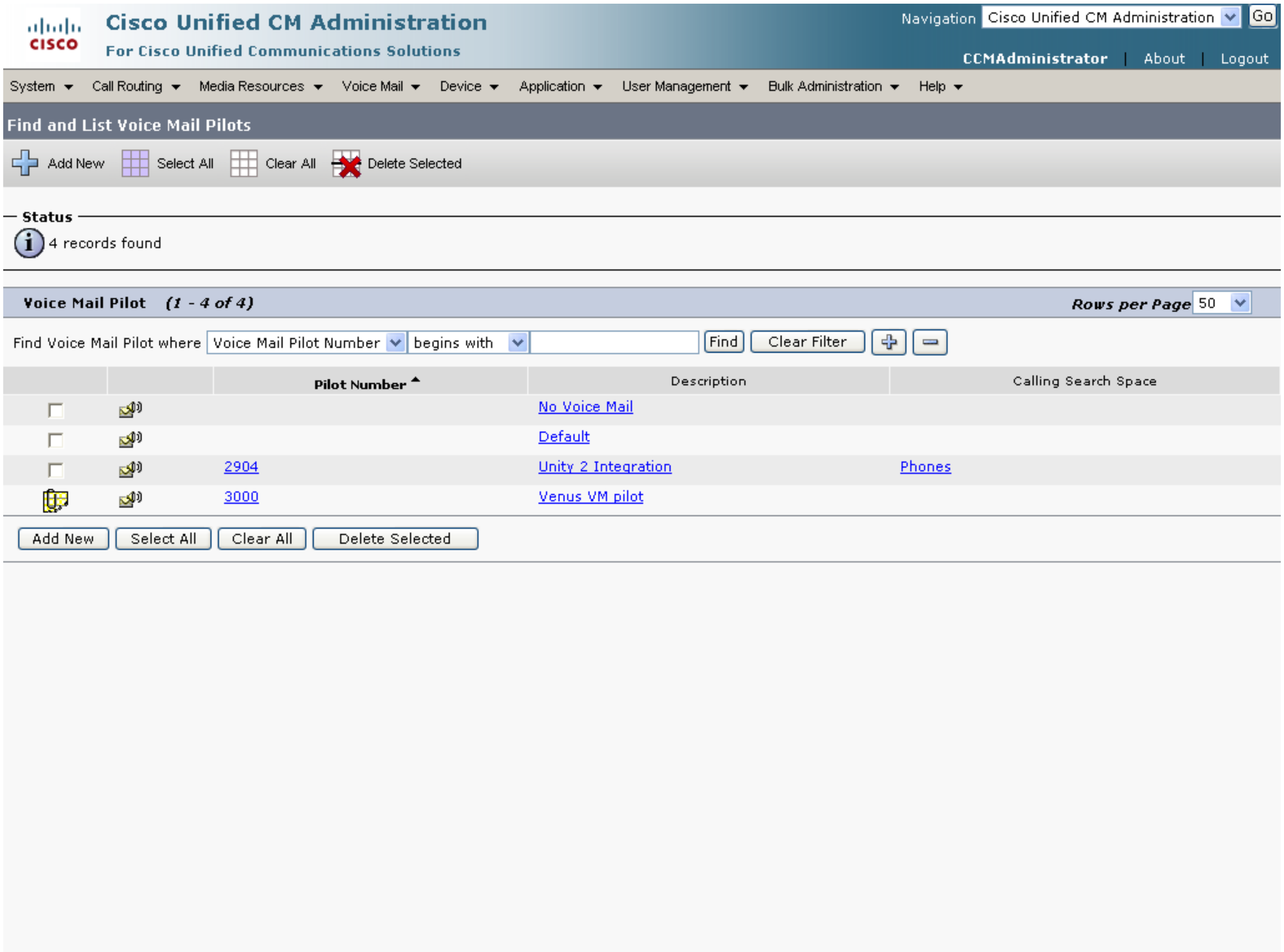
Voice Mail Profiles – 1 of 1.



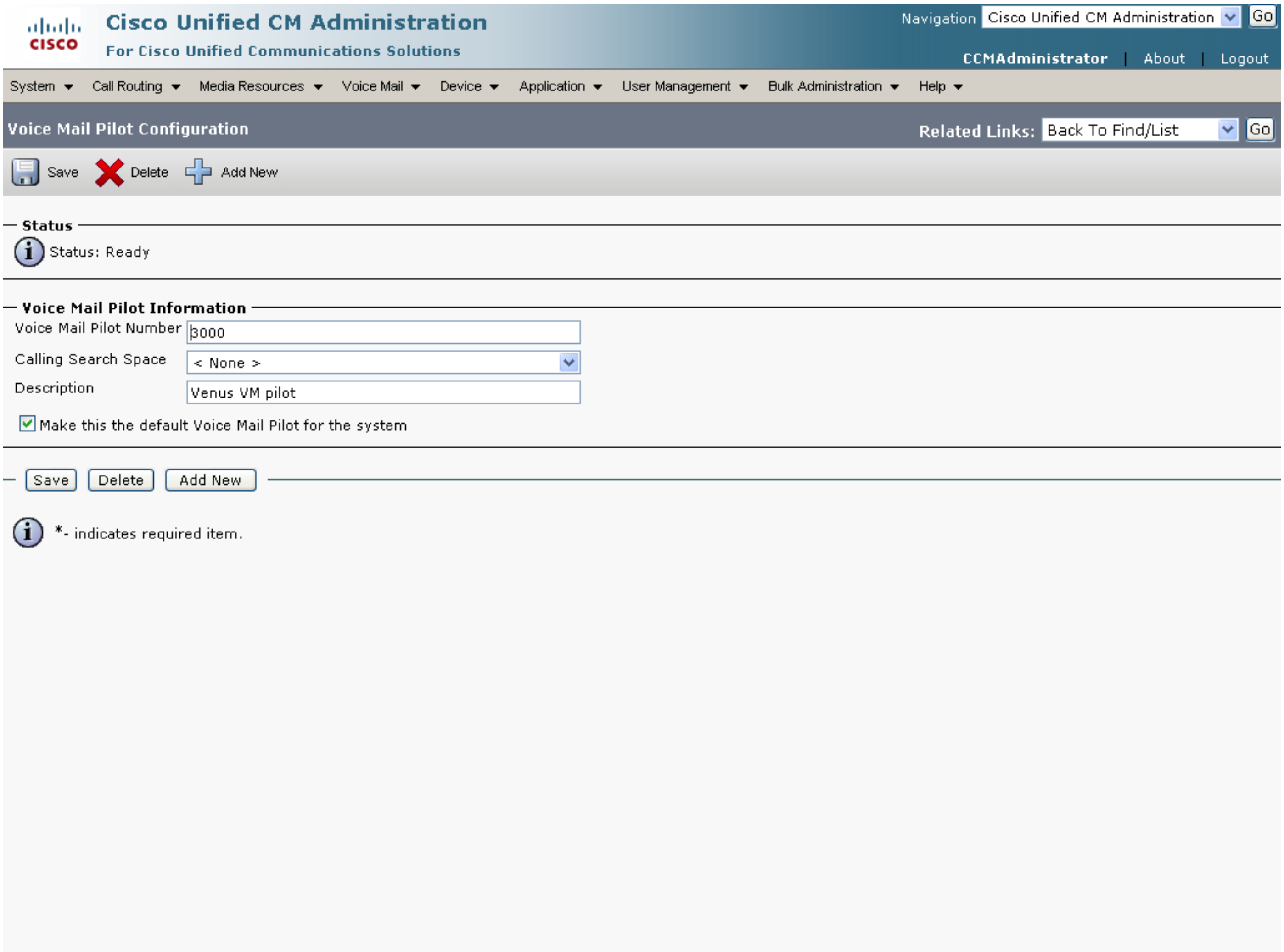
Unity Voice Mail Profile – 1 of 1.



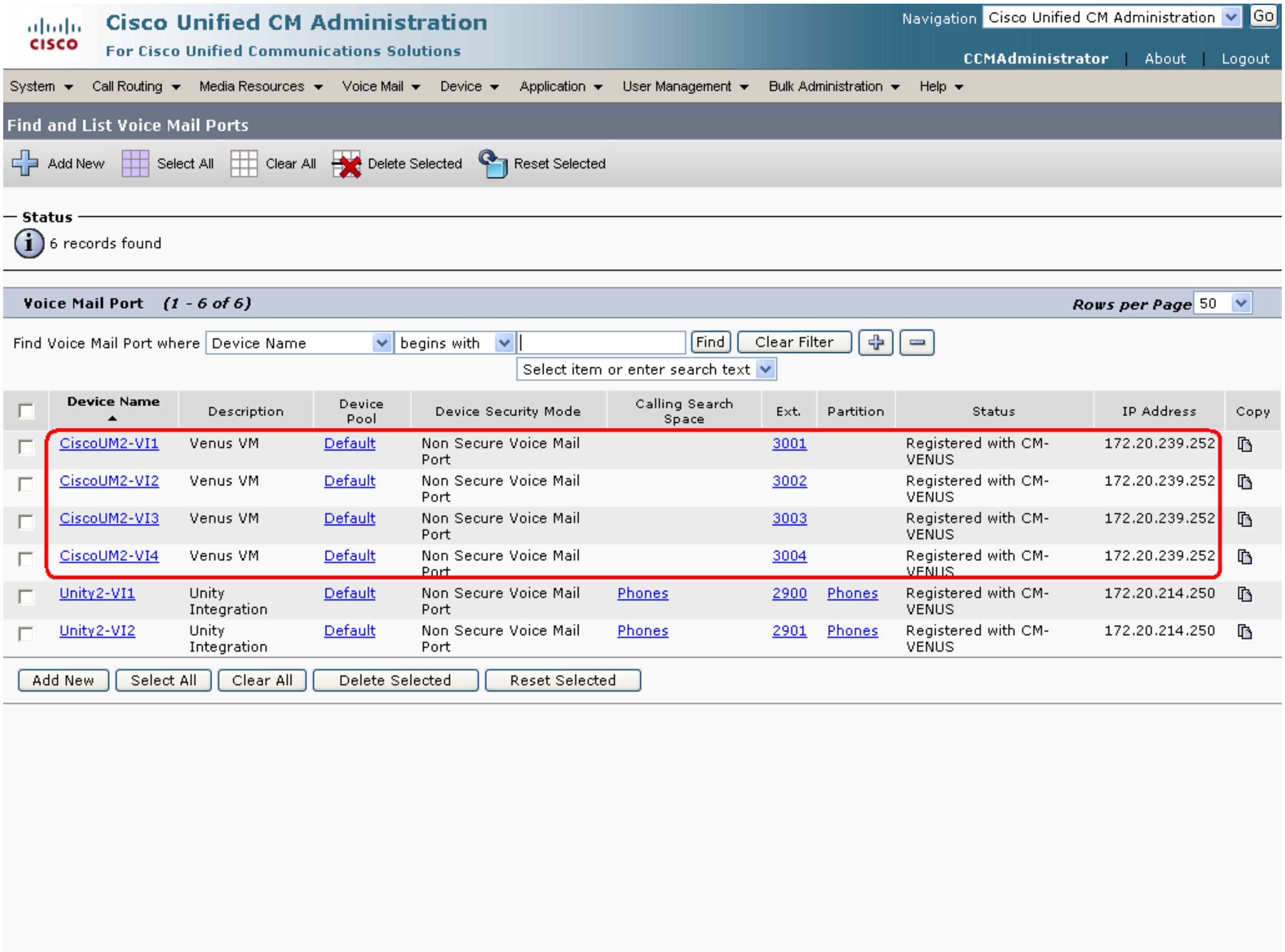
Voice Mail Pilots – 1 of 1.



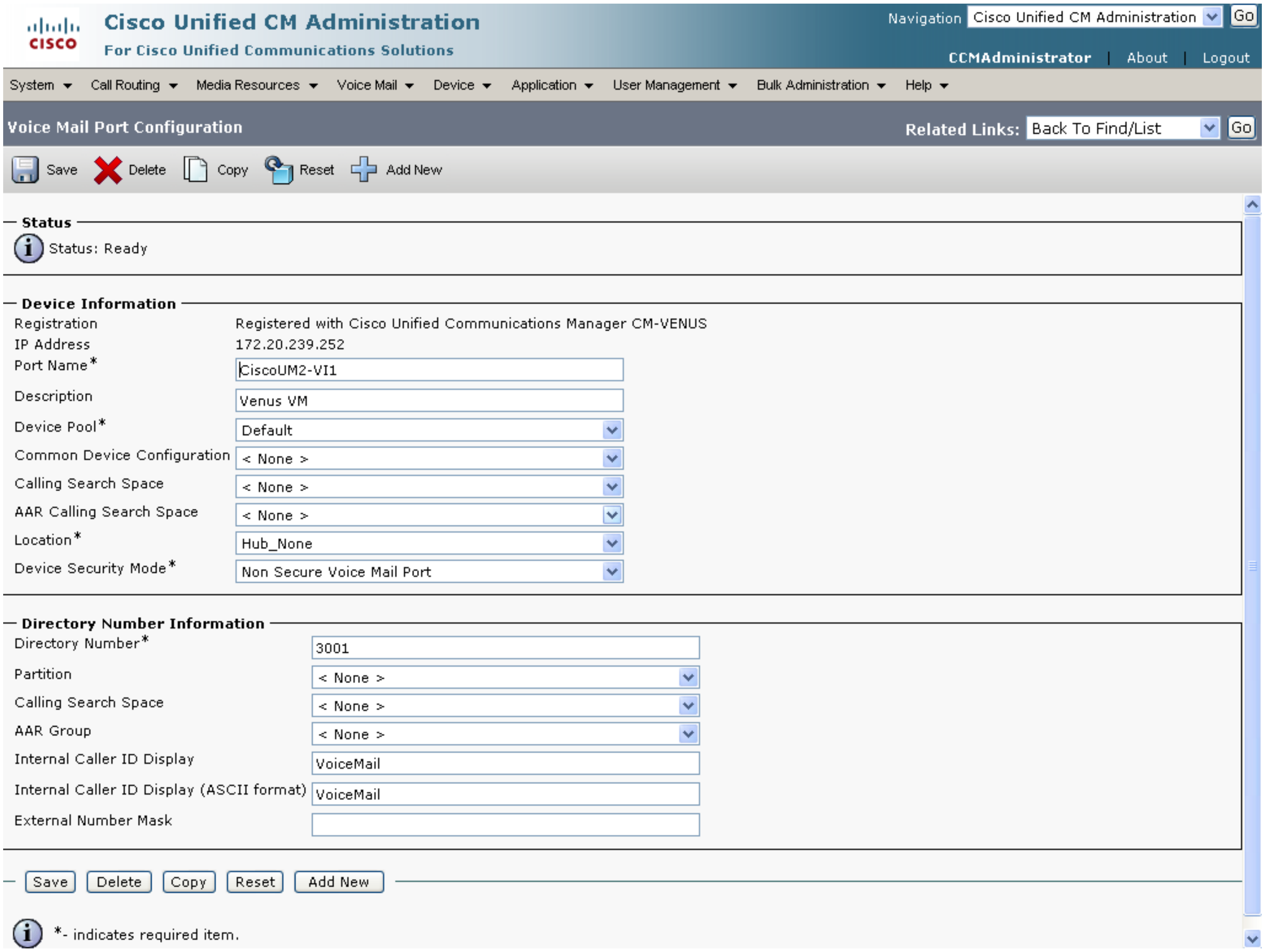
Voice Mail Pilot 3000 (for Cisco Unity) – 1 of 1.



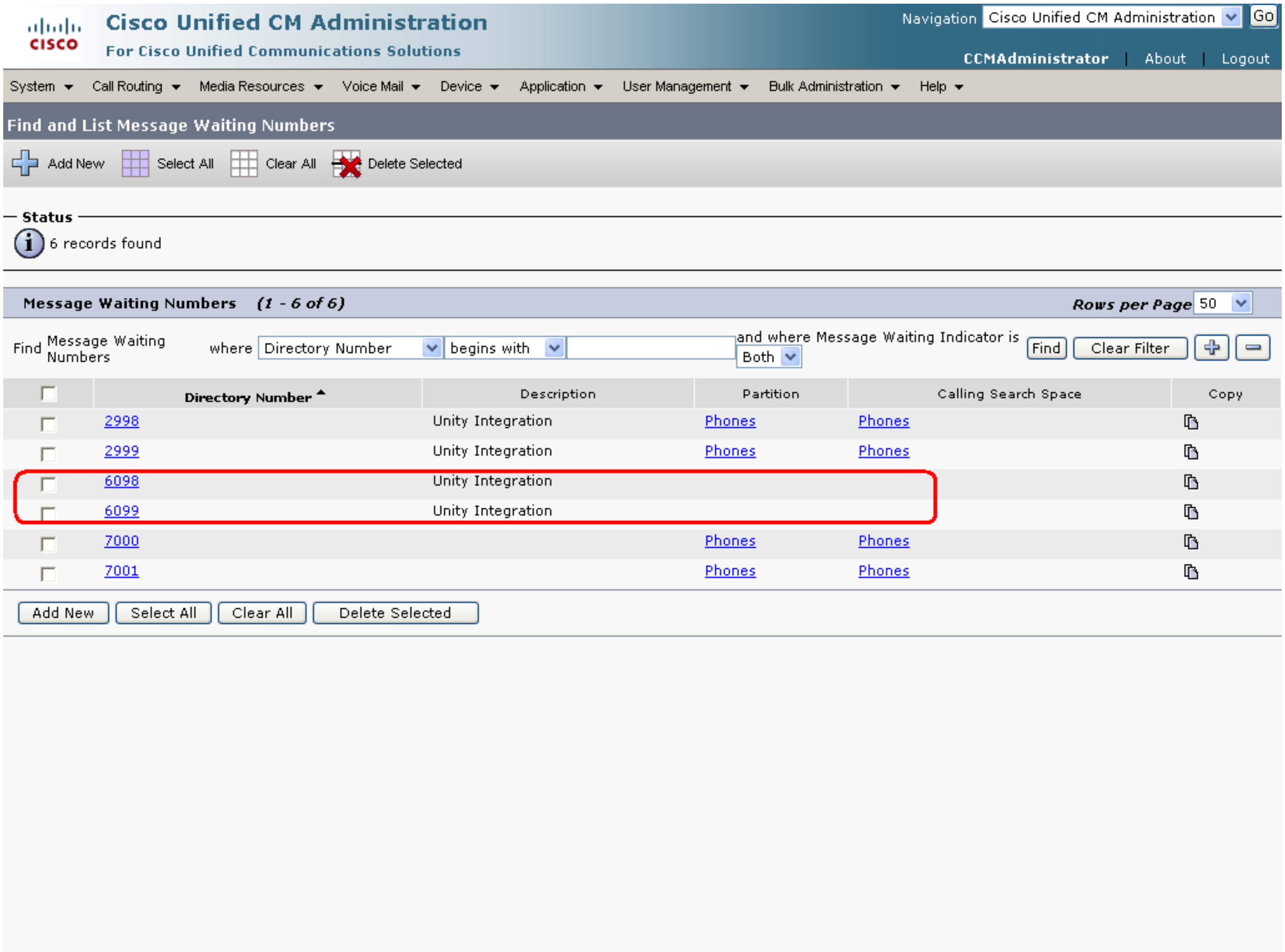
Voice Mail Ports – 1 of 1.



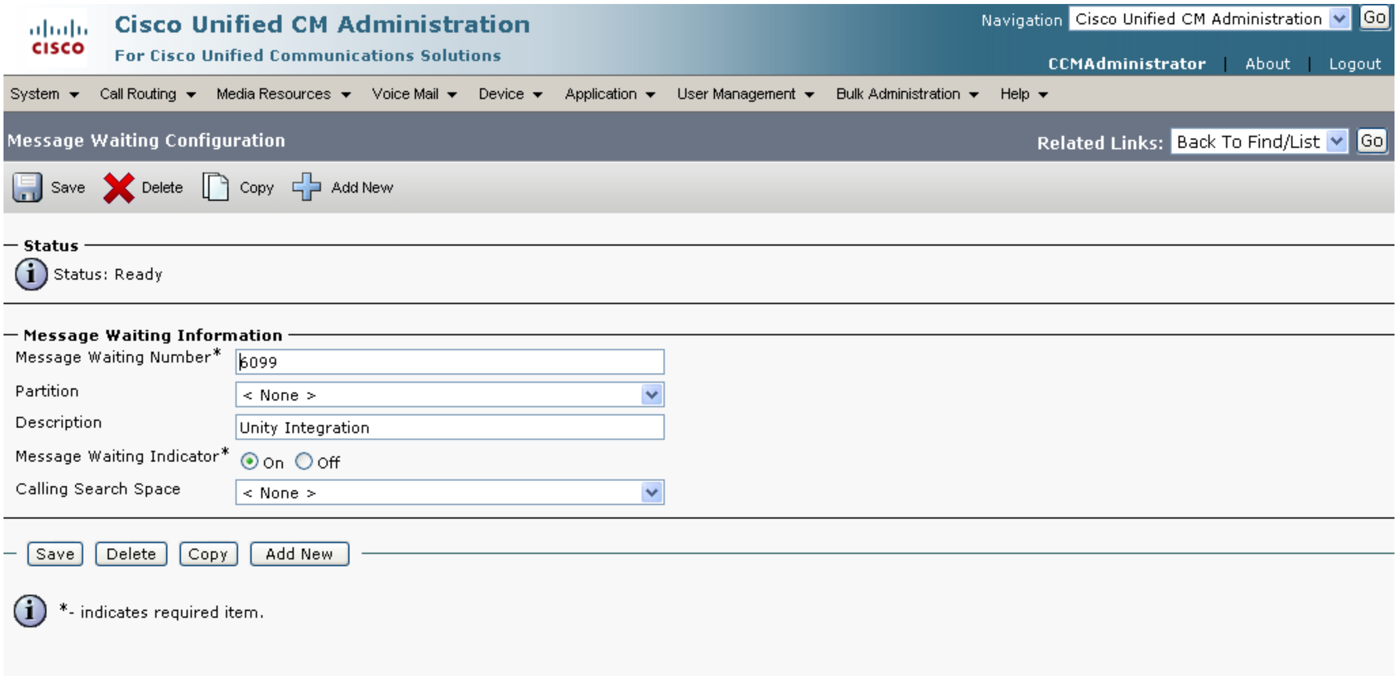
Voice Mail Port 3001 – 1 of 1.



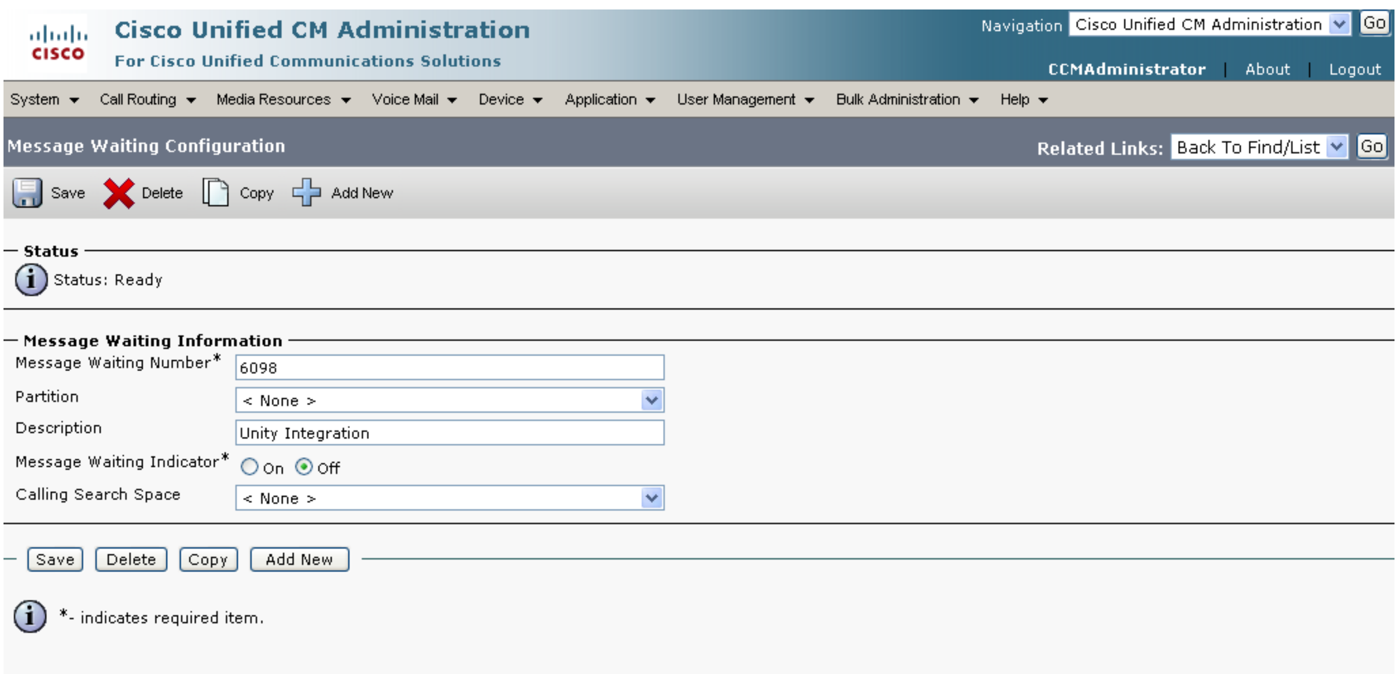
MWI Numbers – 1 of 1.



MWI On Number – 1 of 1.

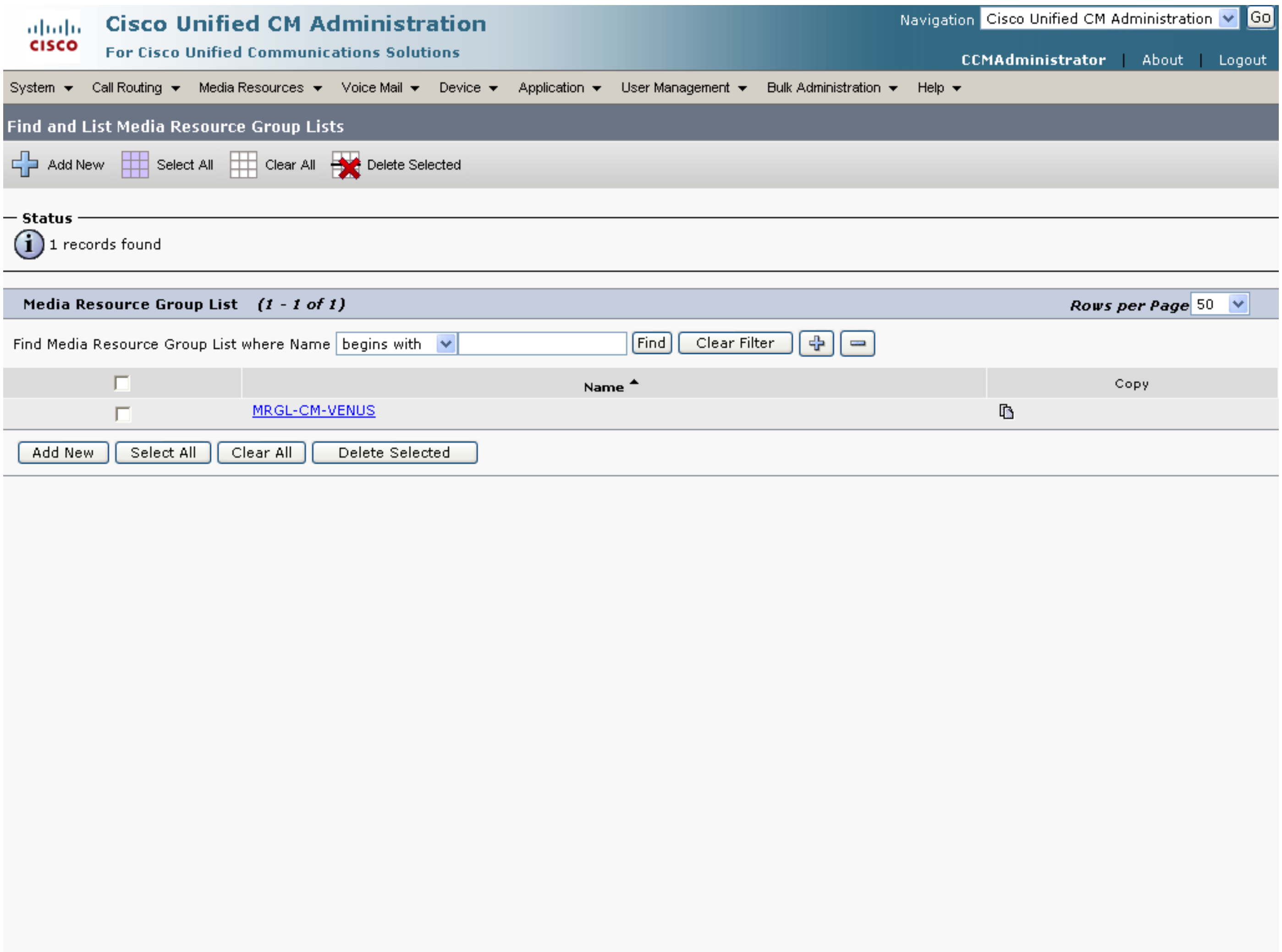


MWI Off Number – 1 of 1.

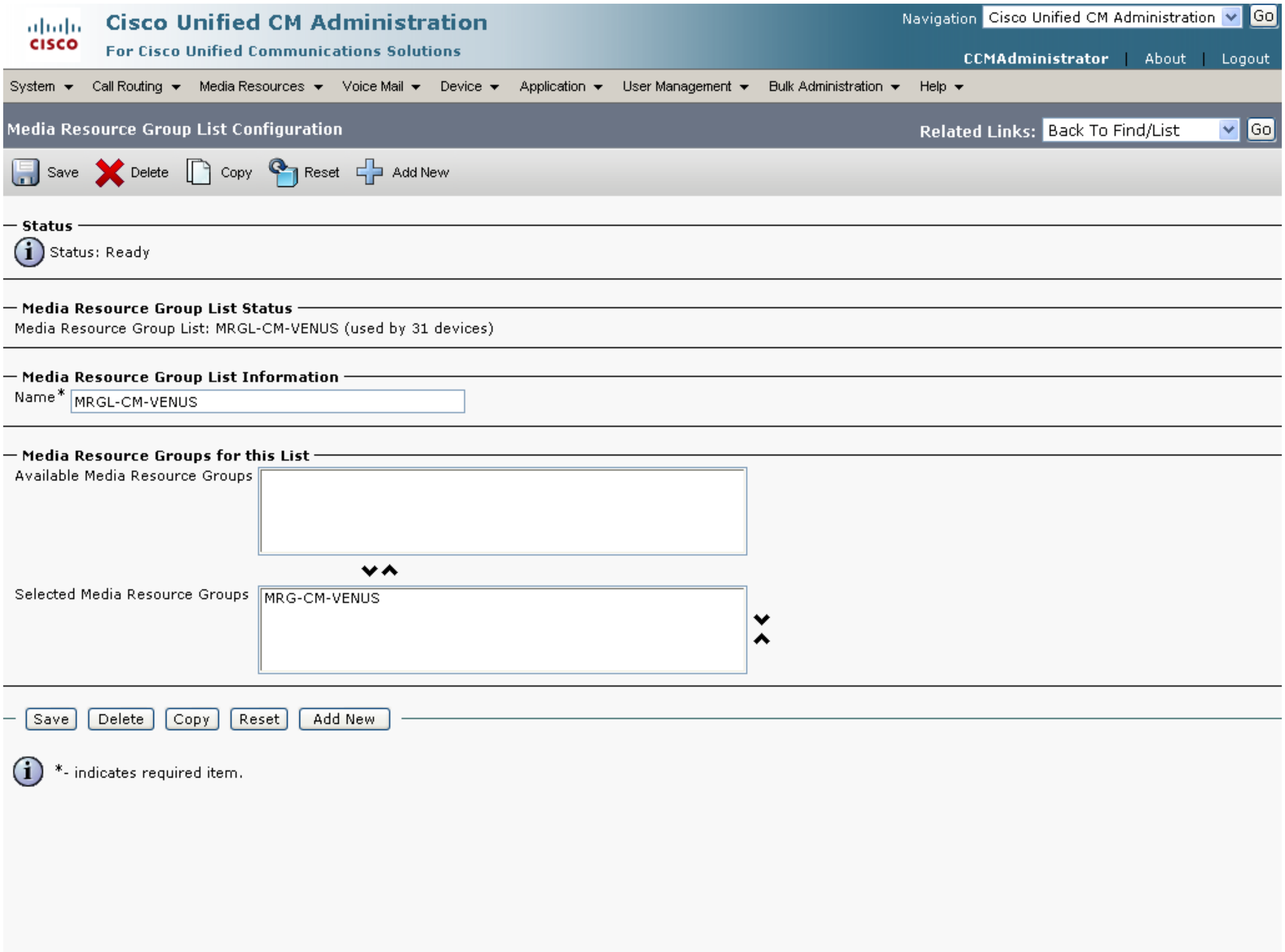


Resources

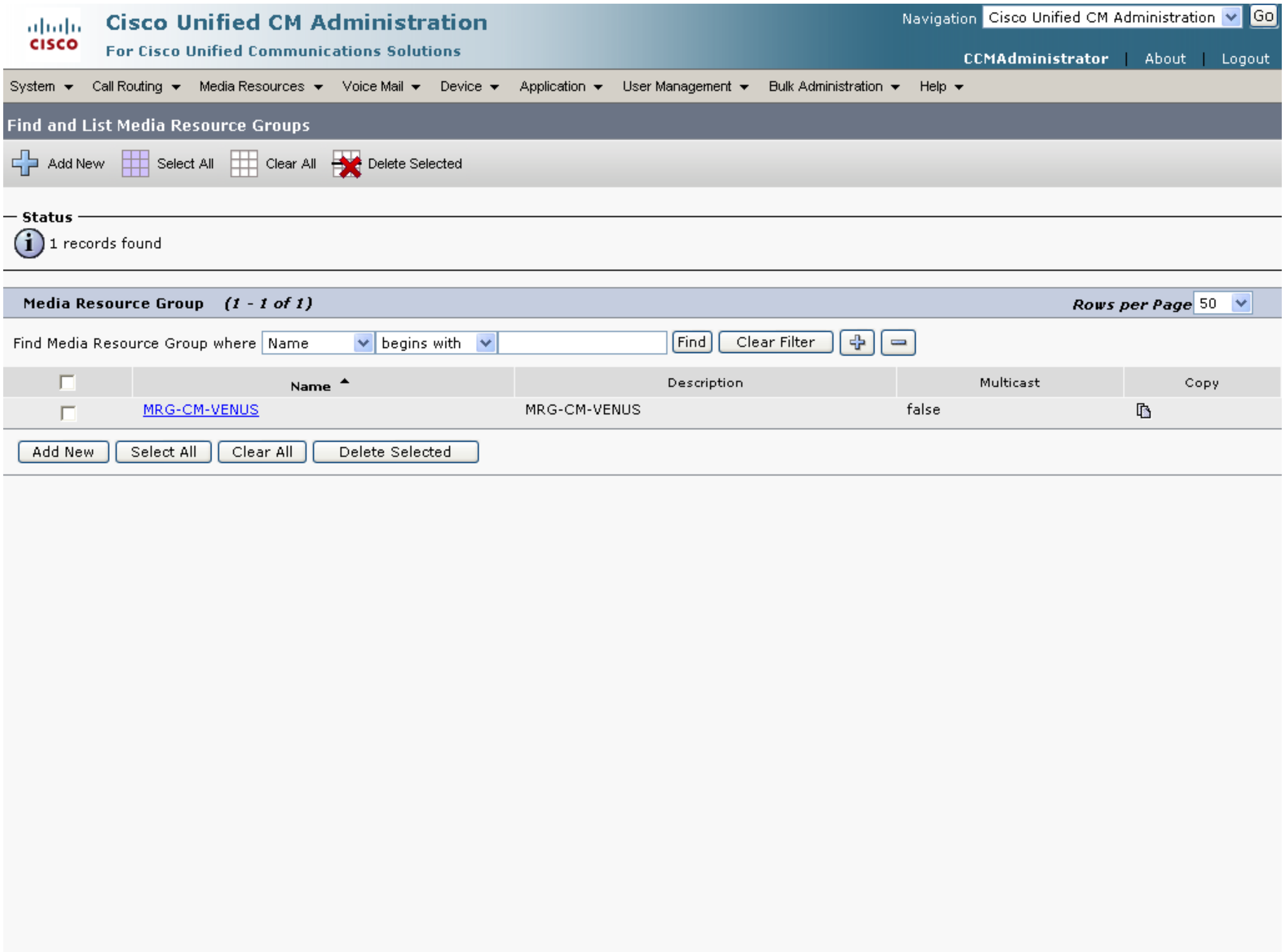
Media Resource Group Lists – 1 of 1.



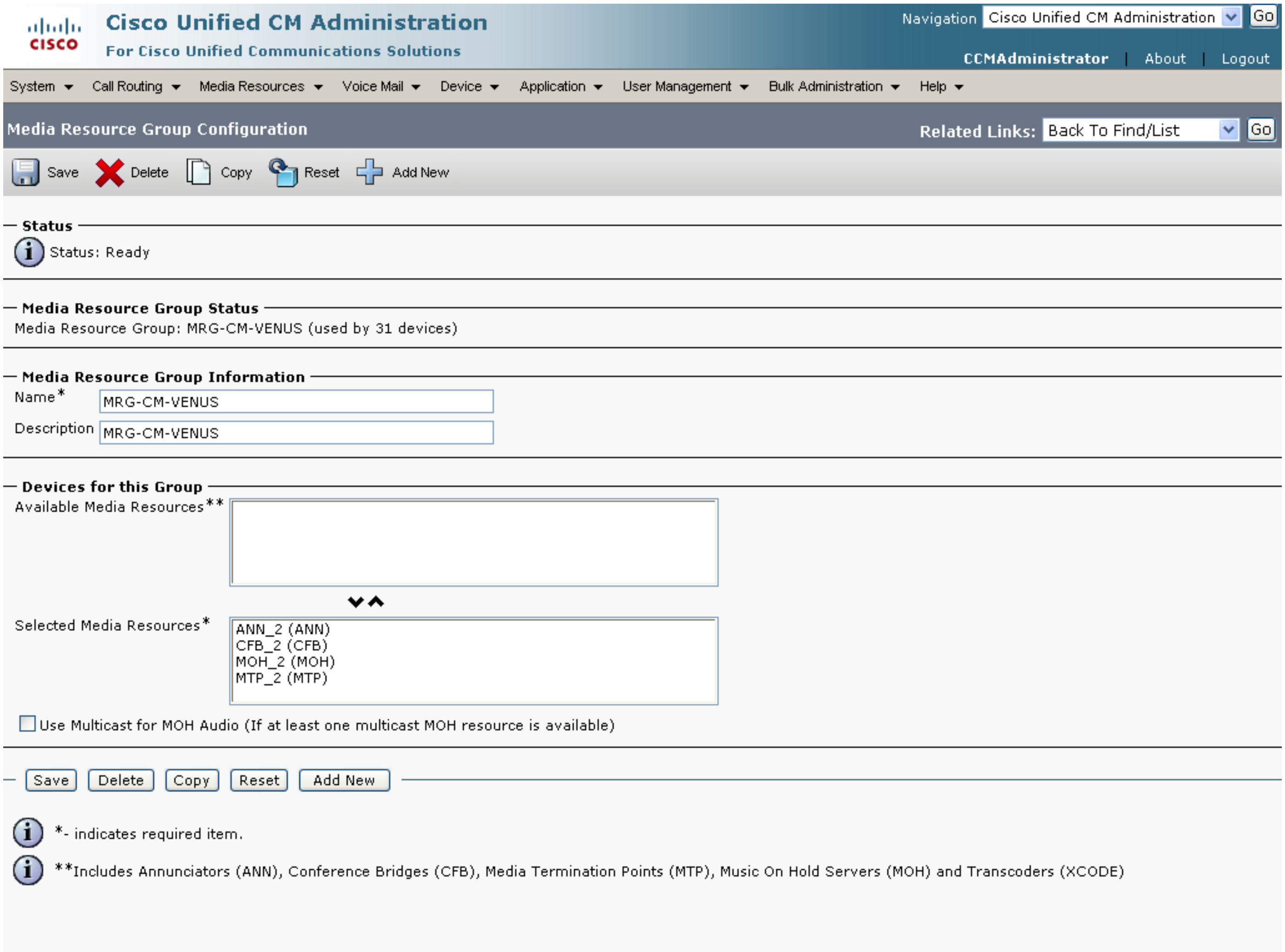
Media Resource Group List Details – 1 of 1.



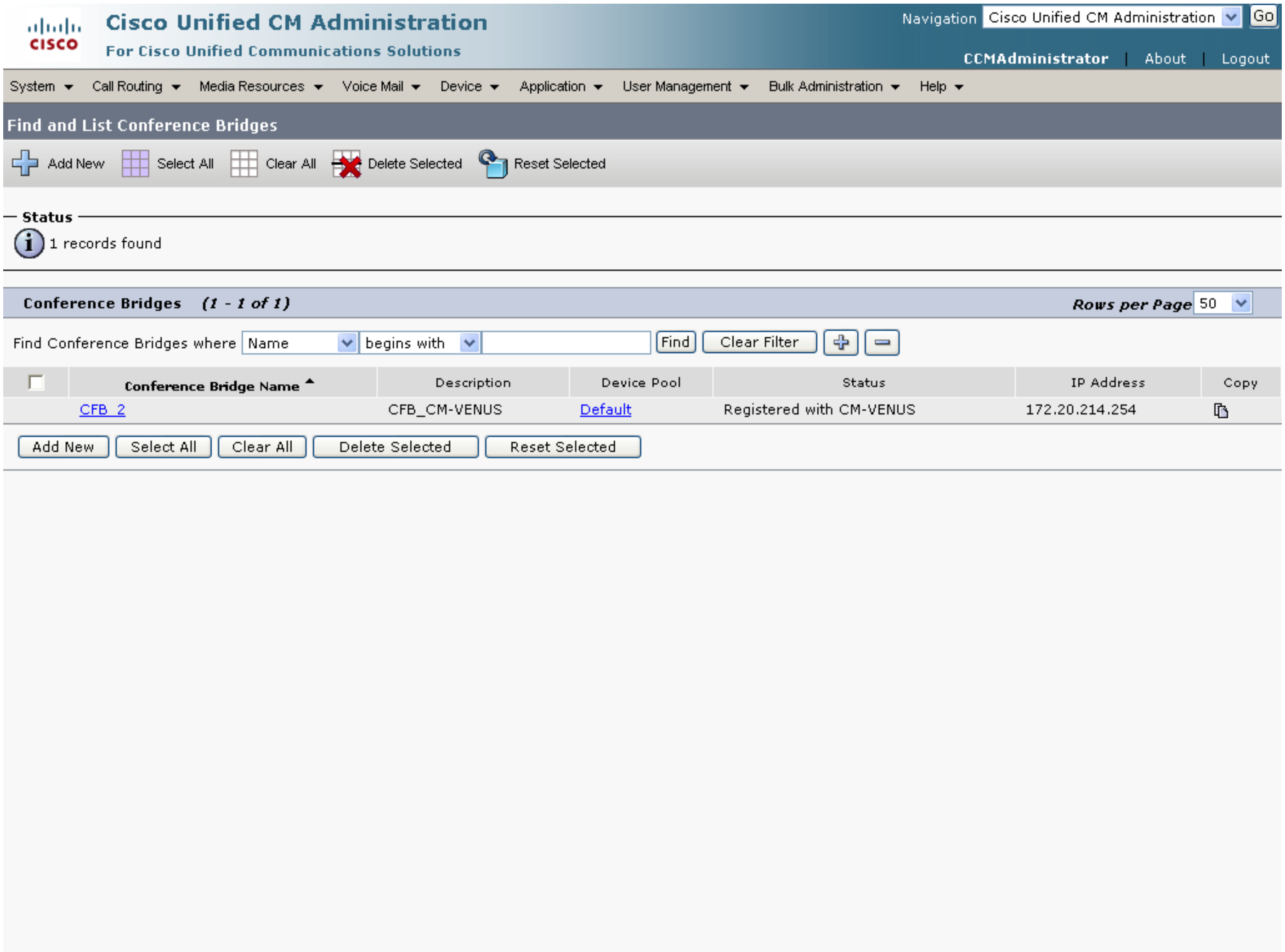
Media Resource Groups – 1 of 1.



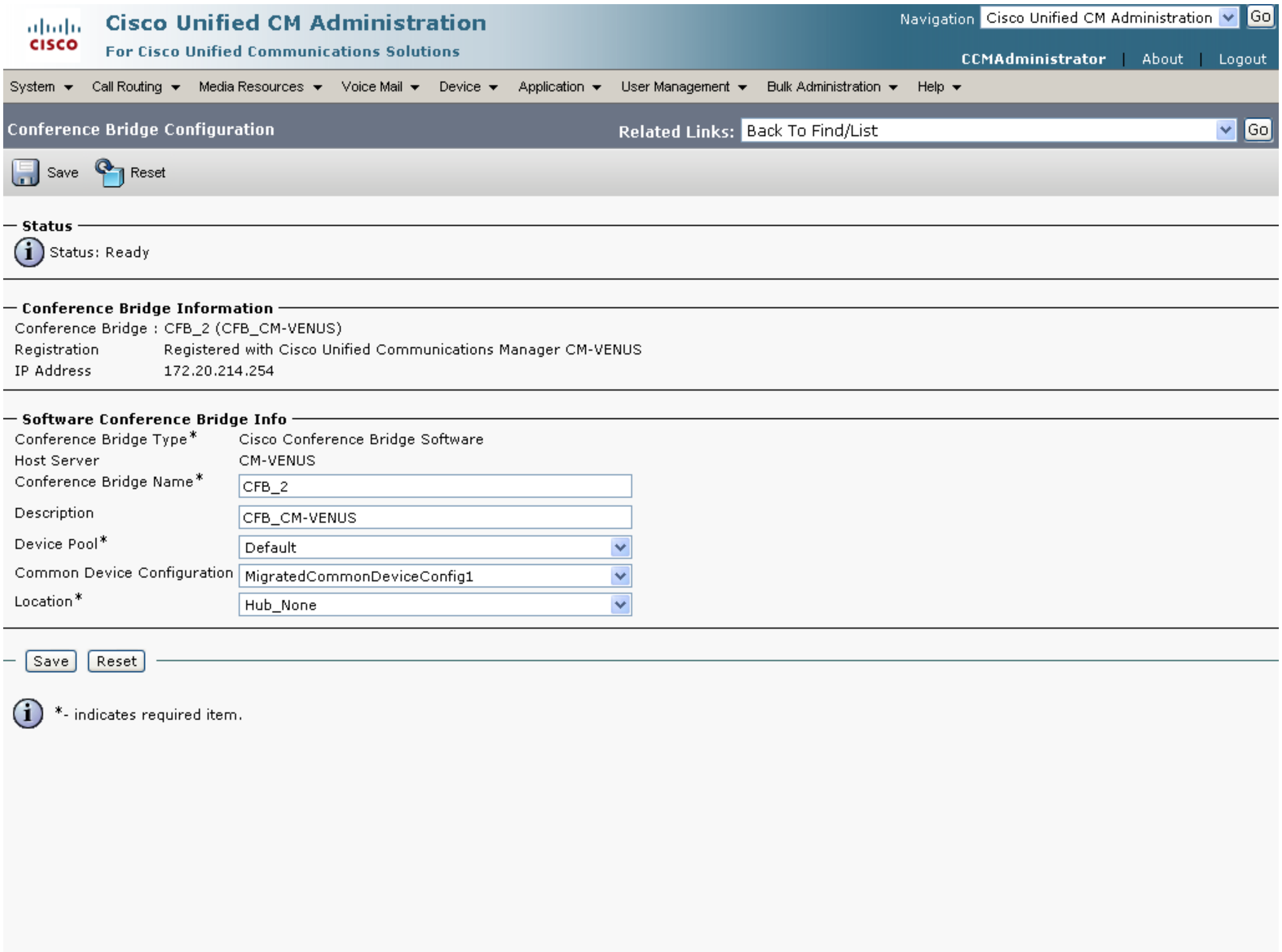
Media Resource Group Details– 1 of 1.



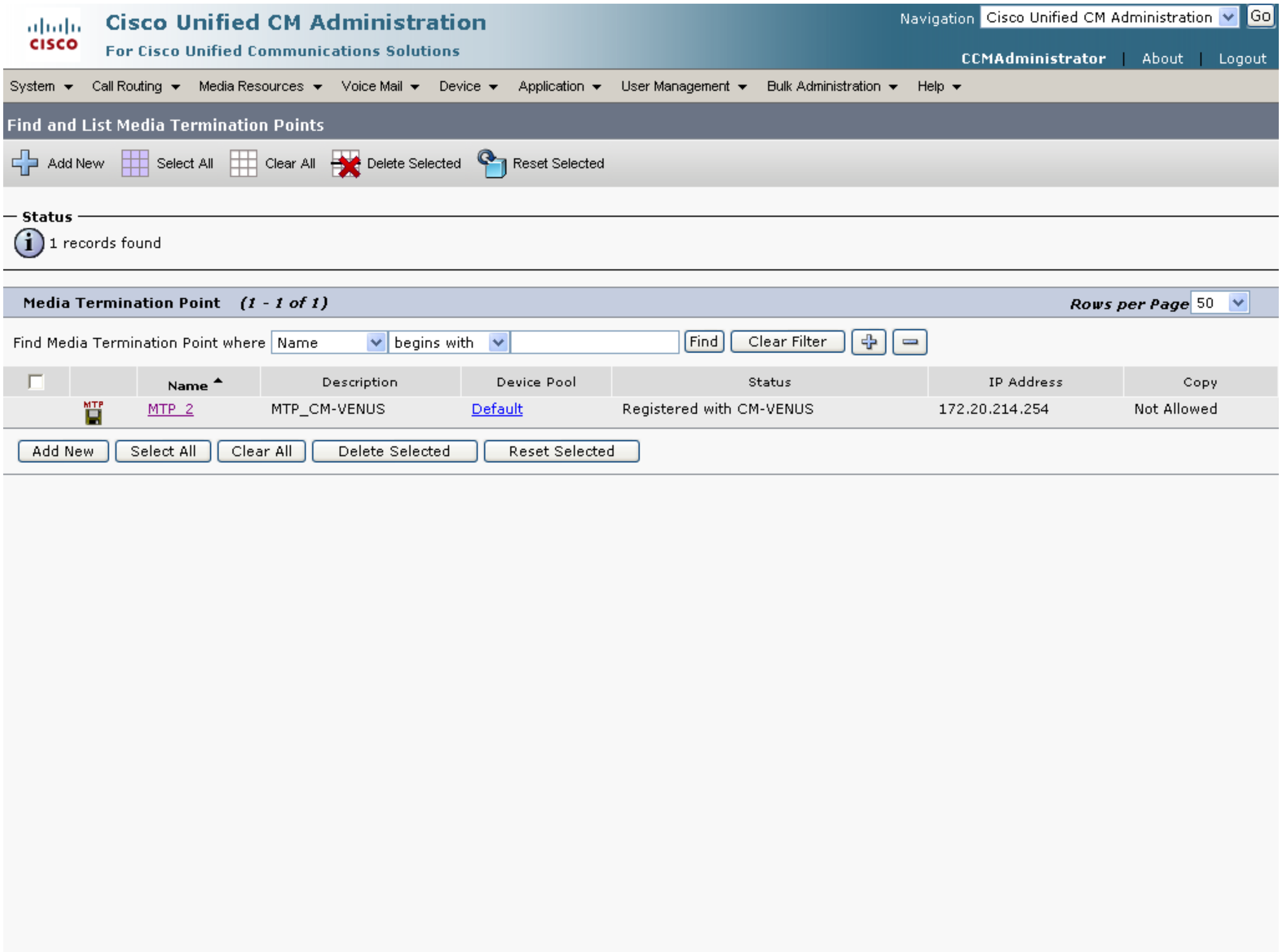
Conference Bridges – 1 of 1.



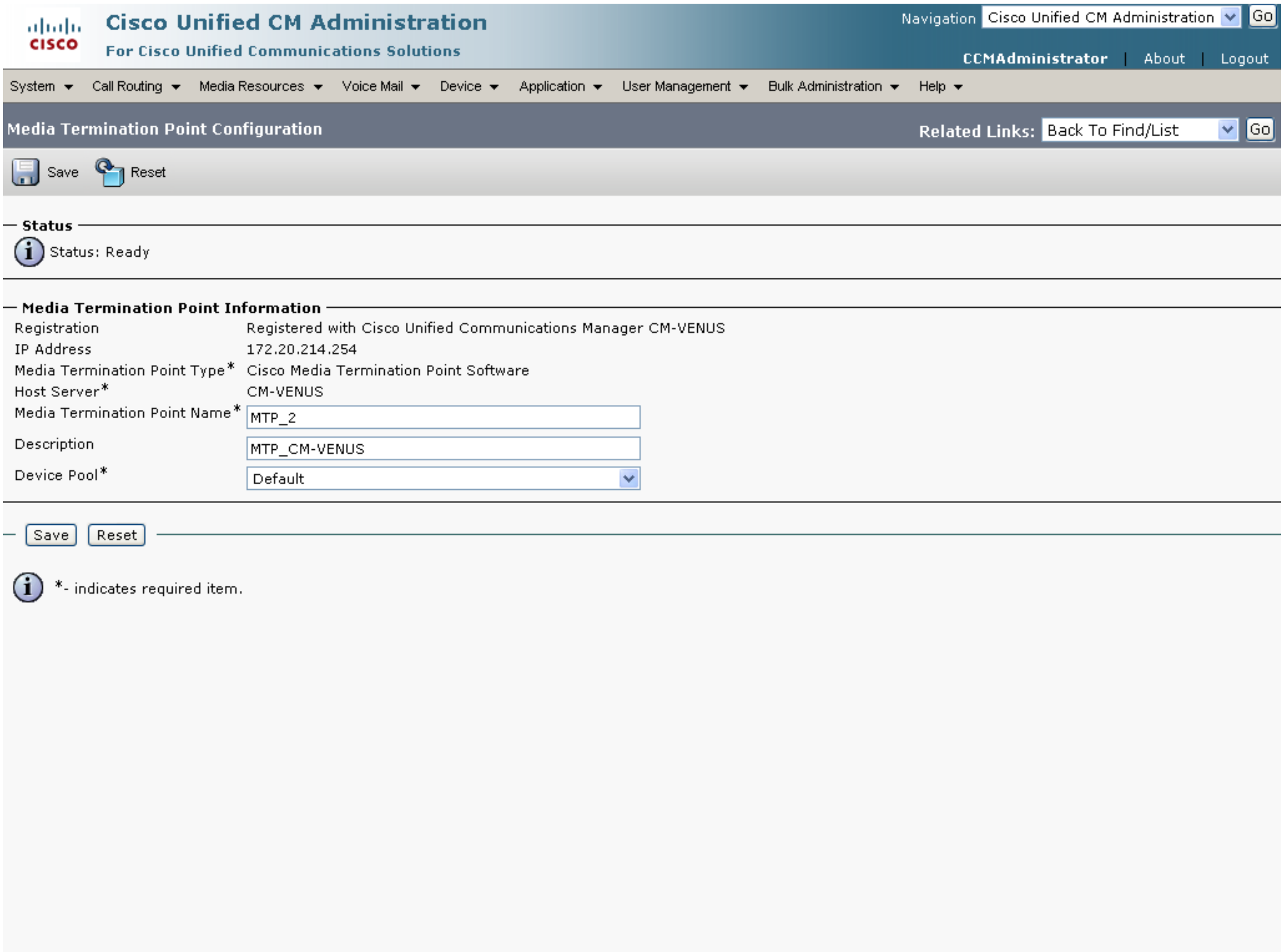
Conference Bridge Details – 1 of 1.



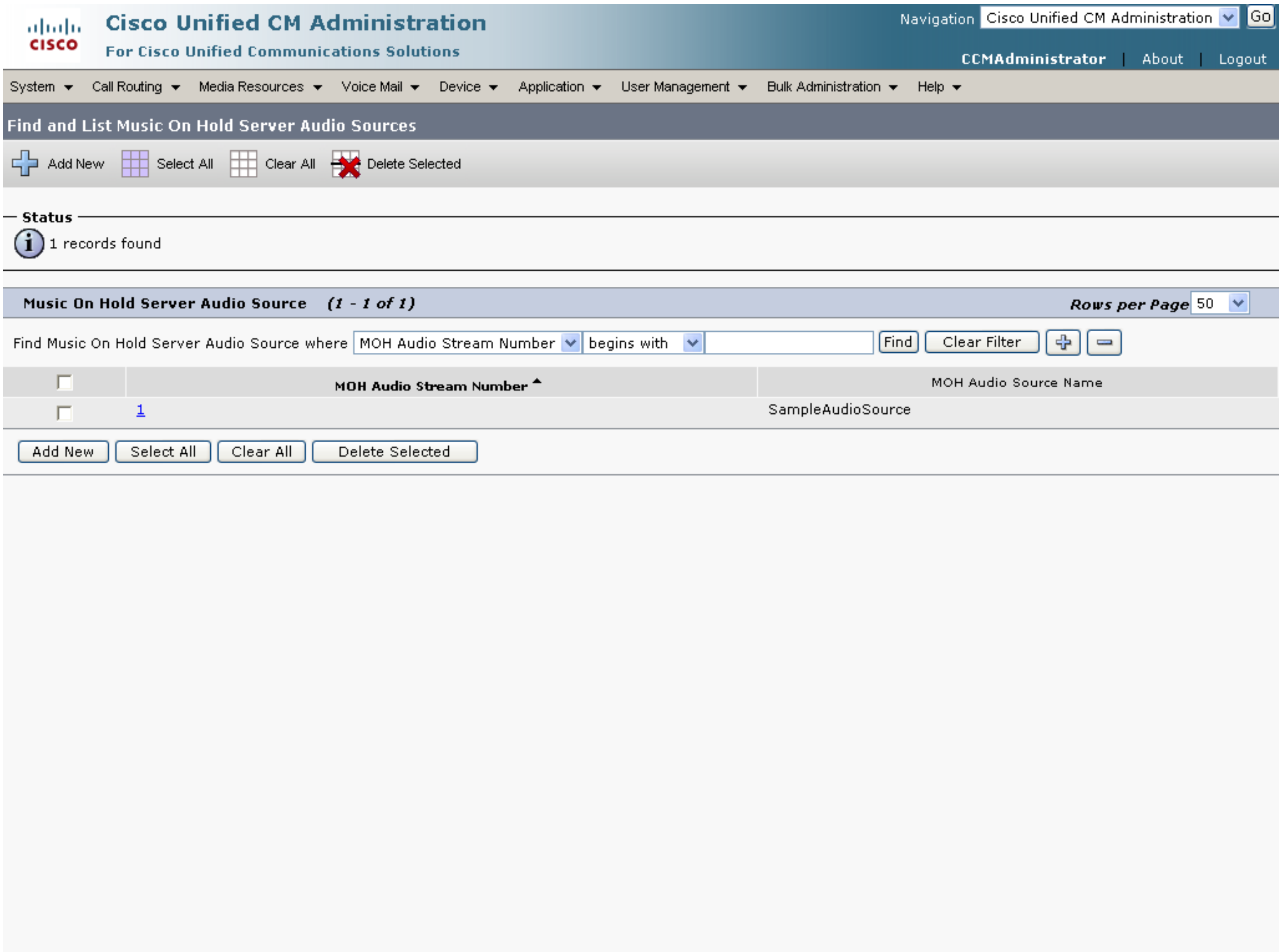
Media Termination Points – 1 of 1.



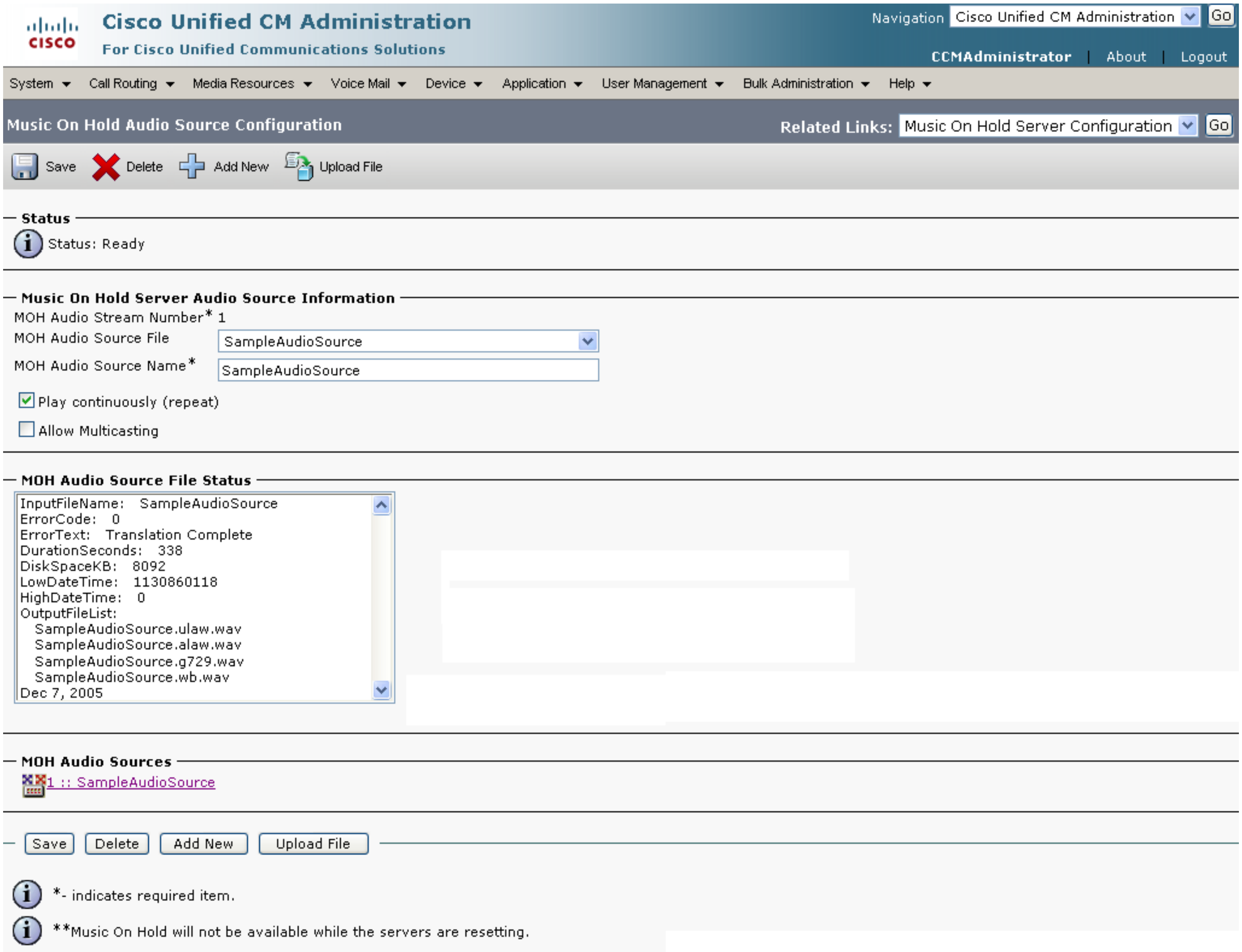
Media Termination Point Details – 1 of 1.



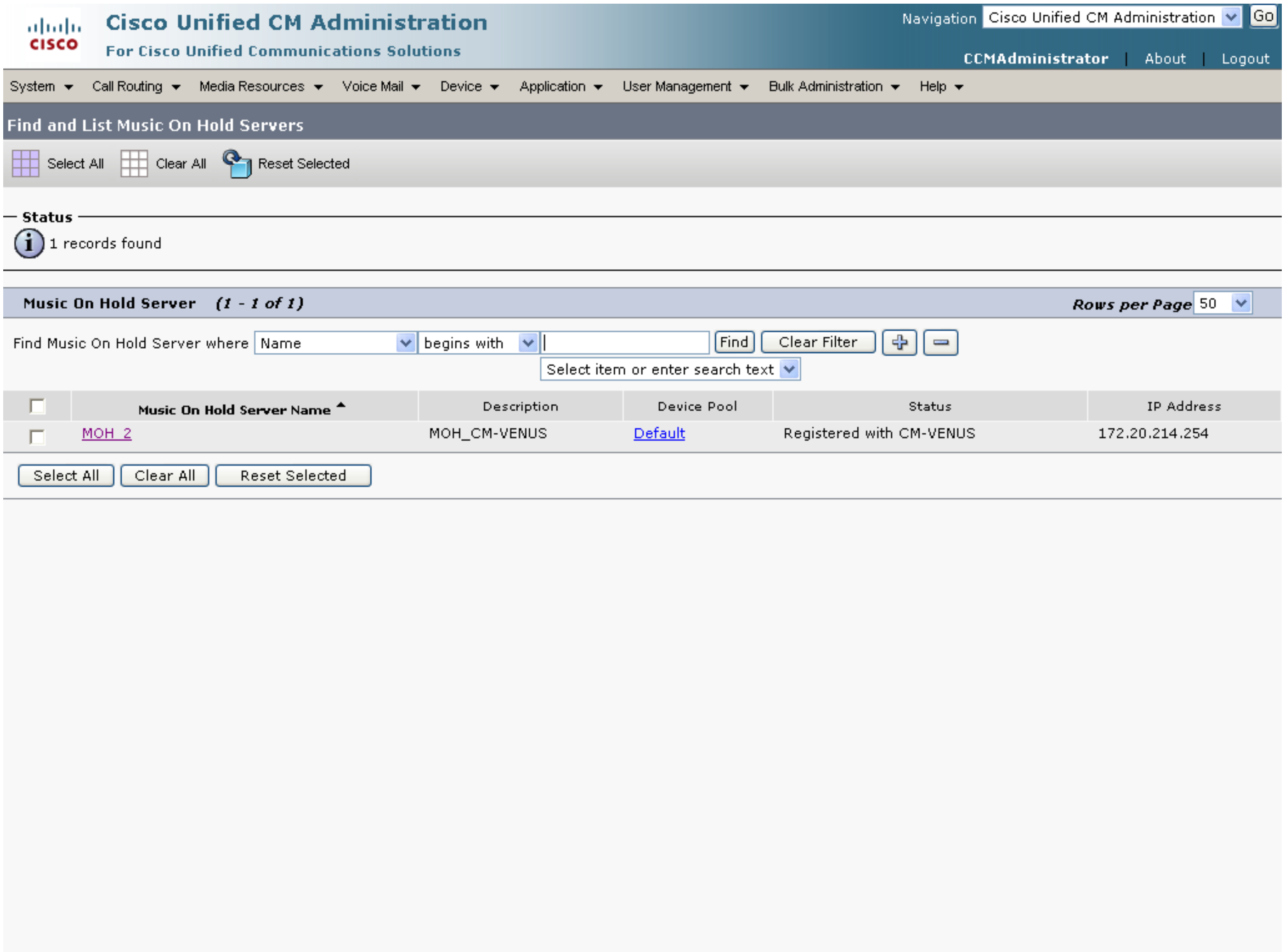
MoH Audio Sources – 1 of 1.



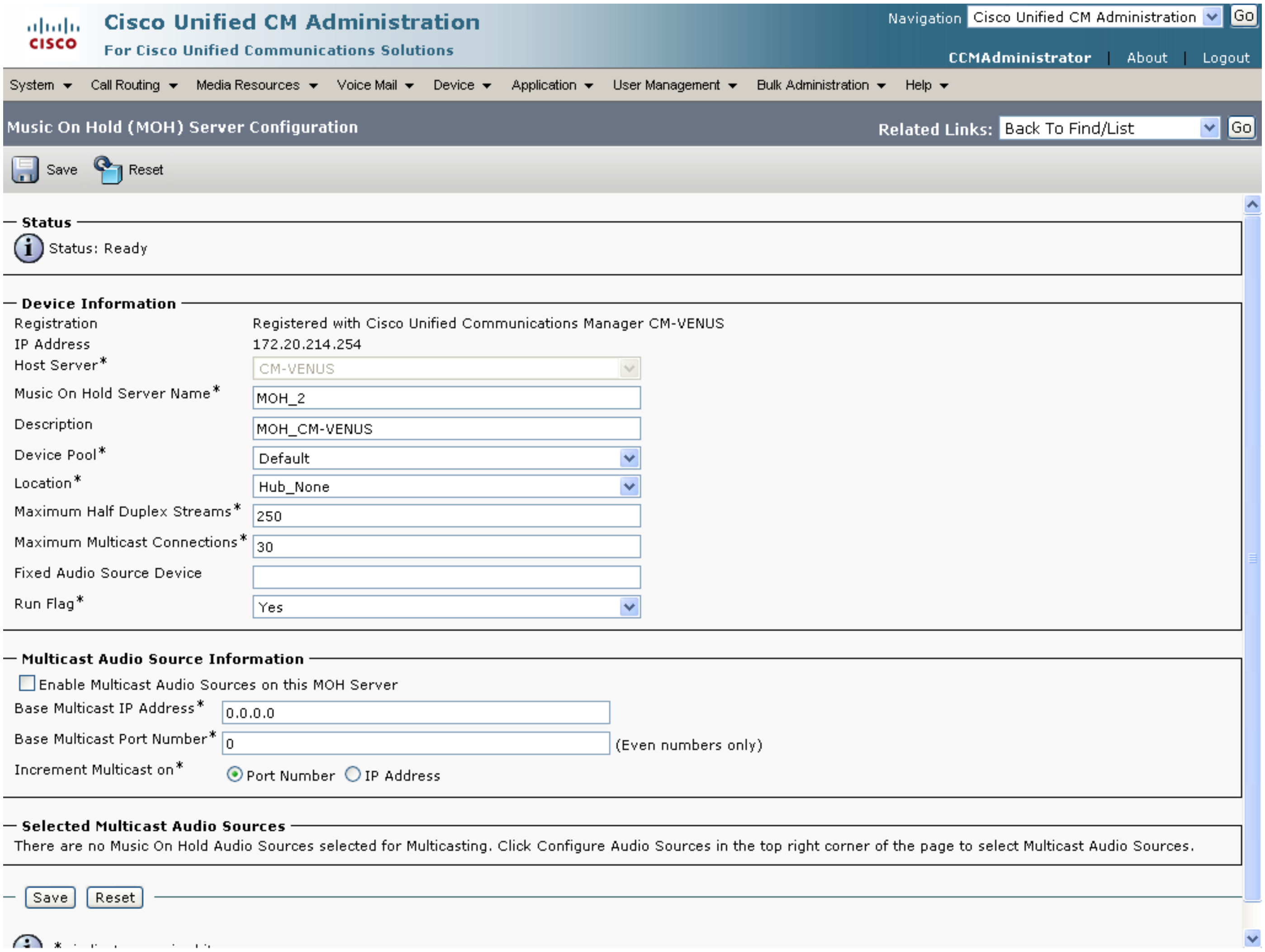
MoH Audio Source Details– 1 of 1.



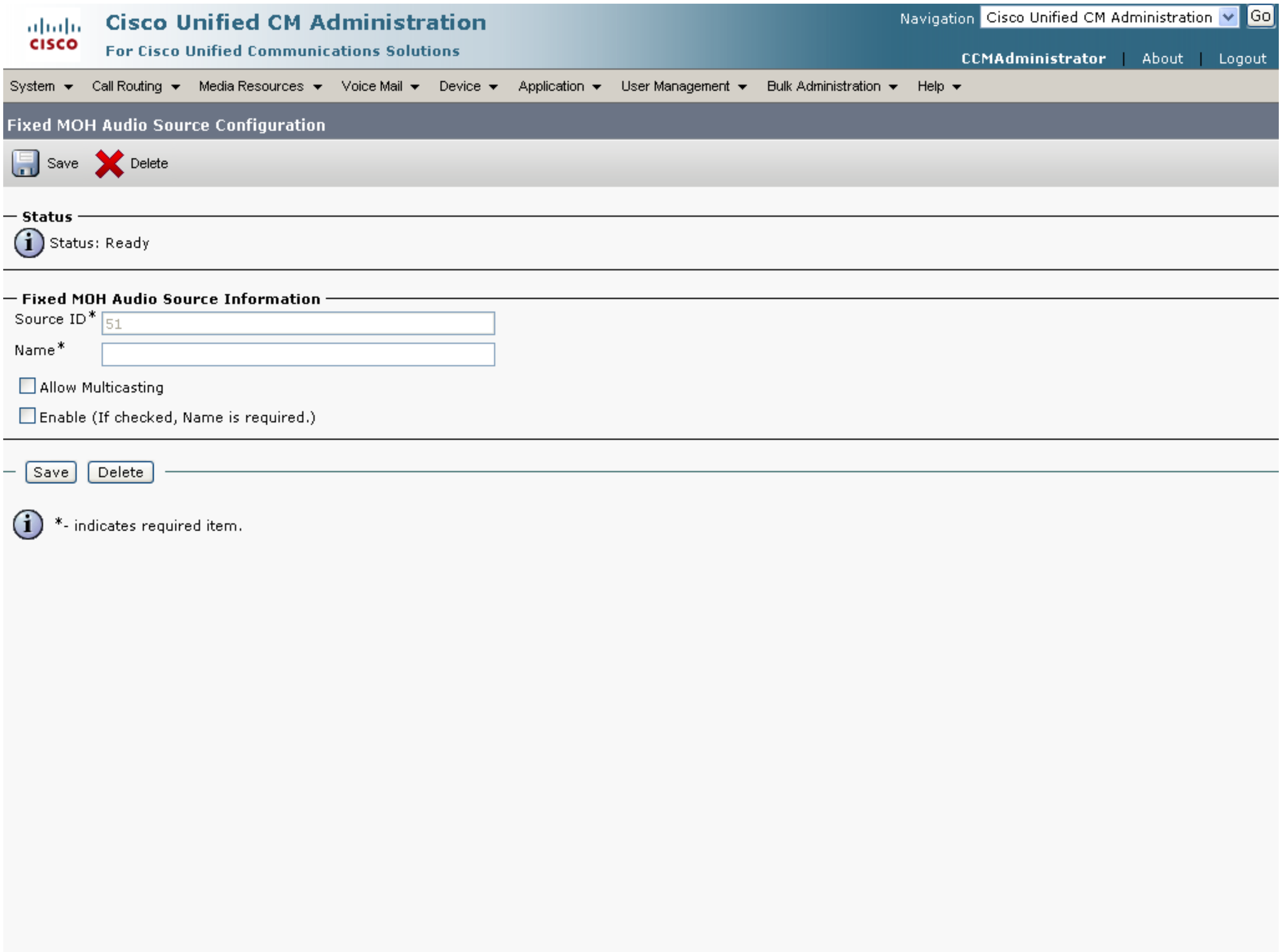
MoH Servers – 1 of 1.



MoH Server Details– 1 of 1.

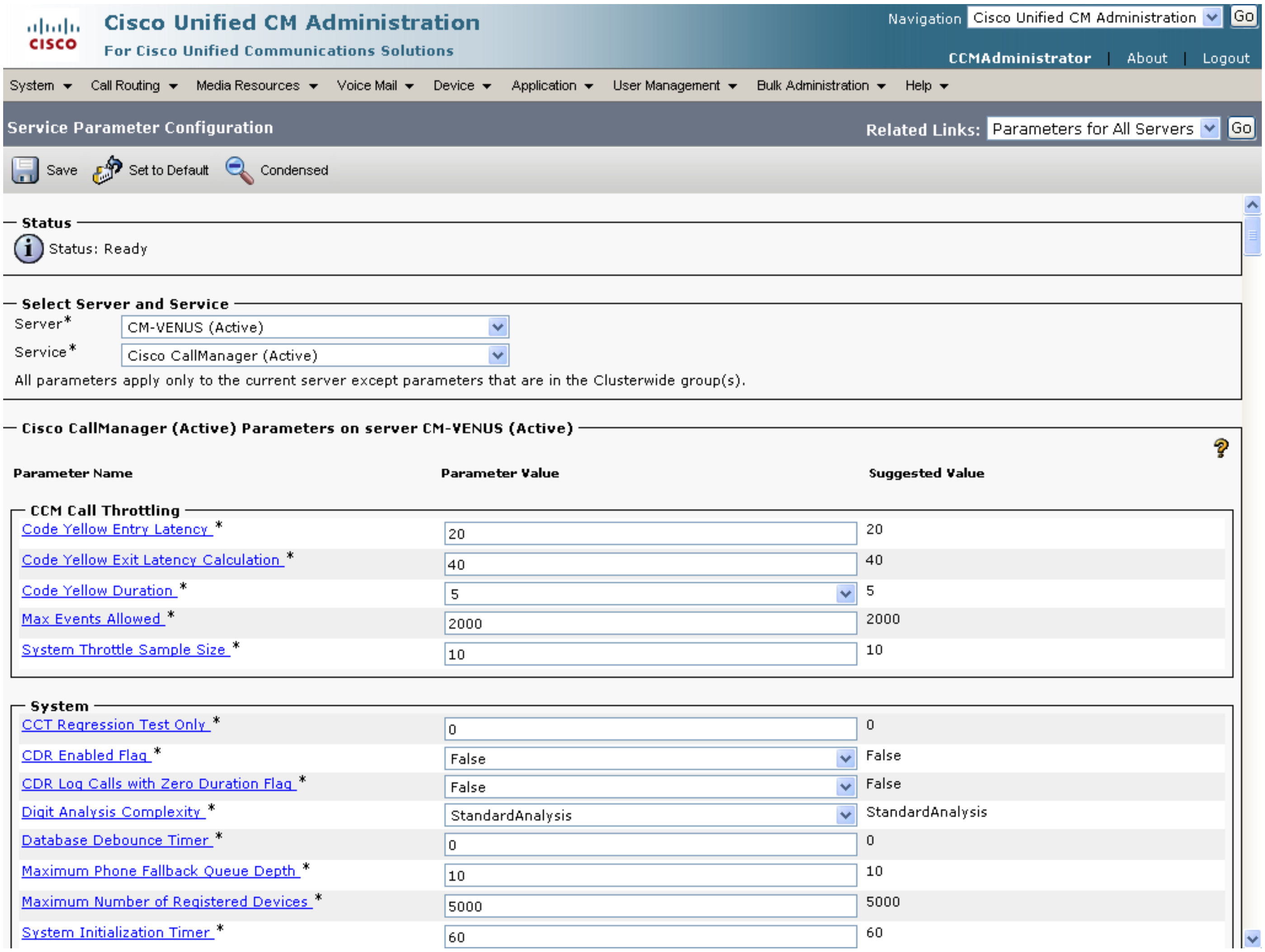


Fixed MoH Audio Source Configuration – 1 of 1.



Service Parameters

Service Parameters – 1 of 19.



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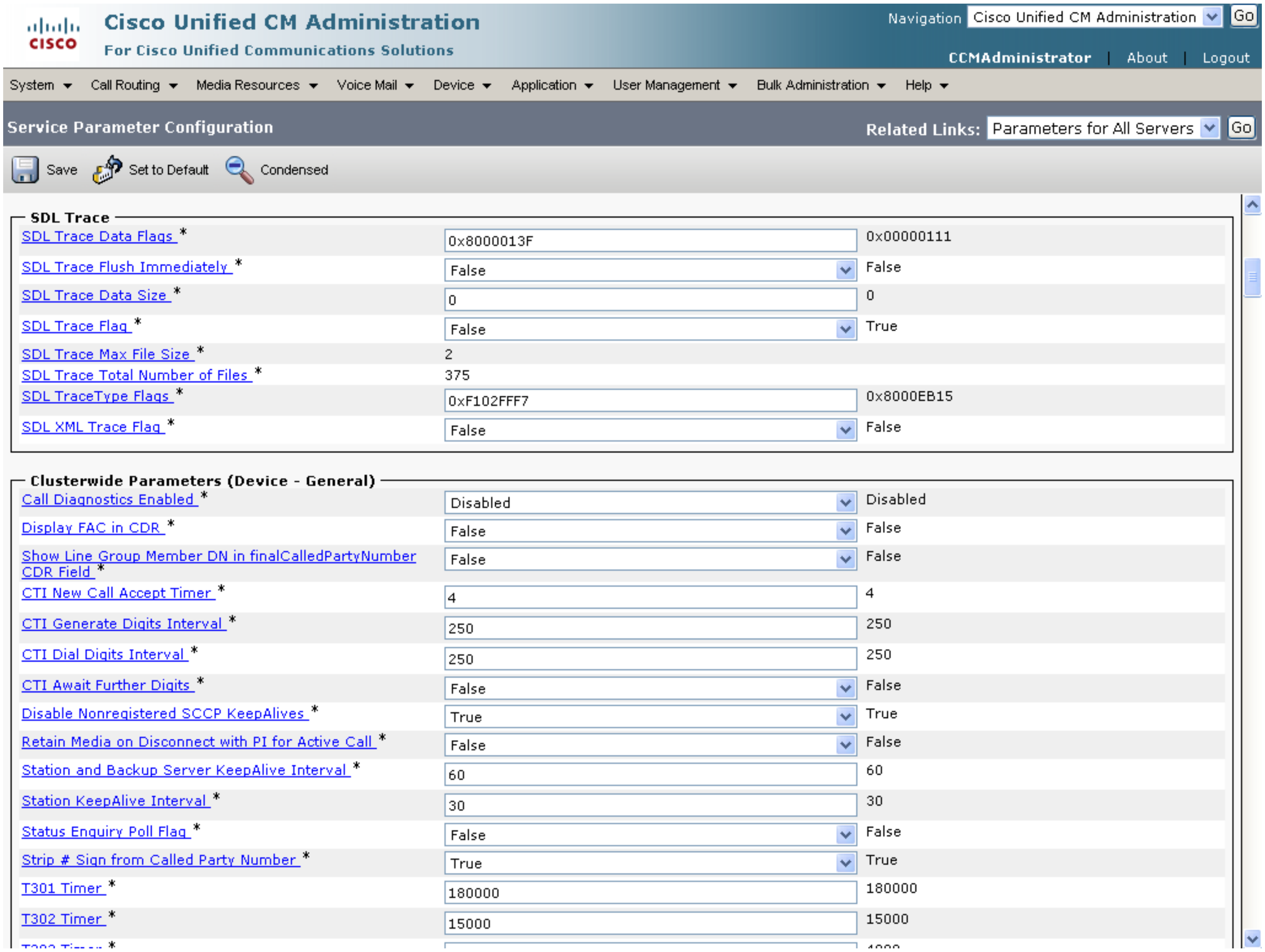
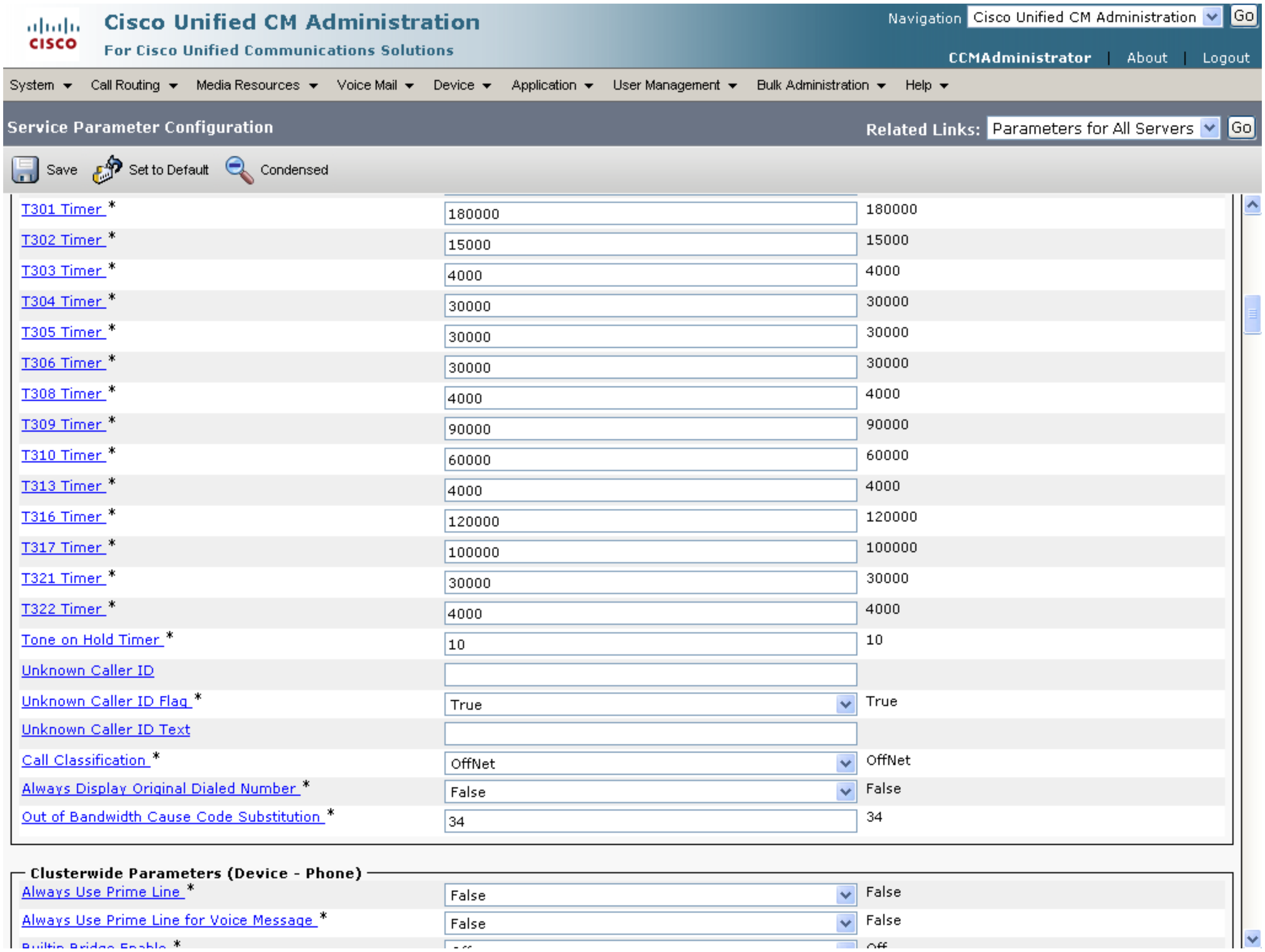
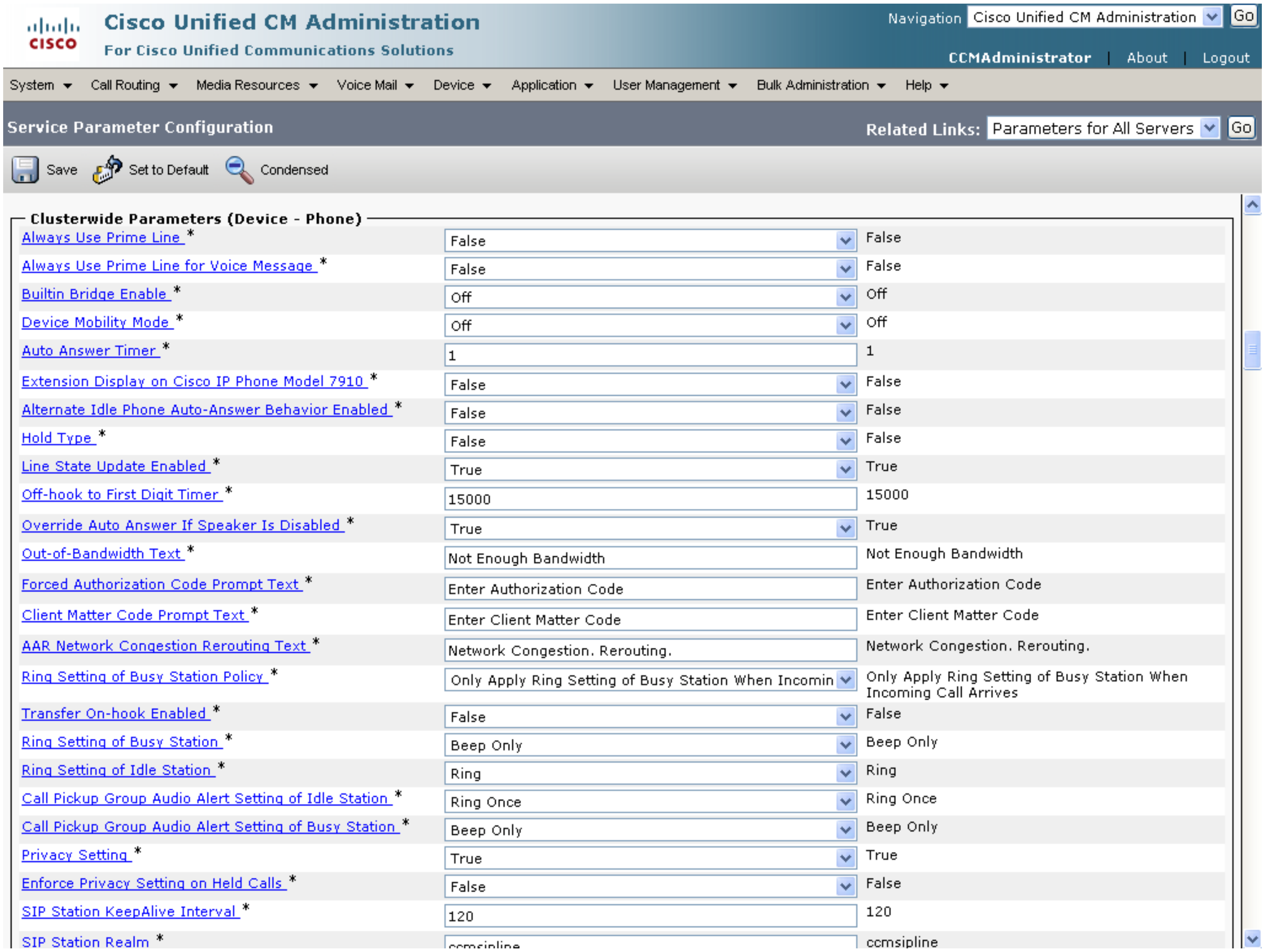


Service Parameters – 3 of 19.



Service Parameters – 4 of 19.



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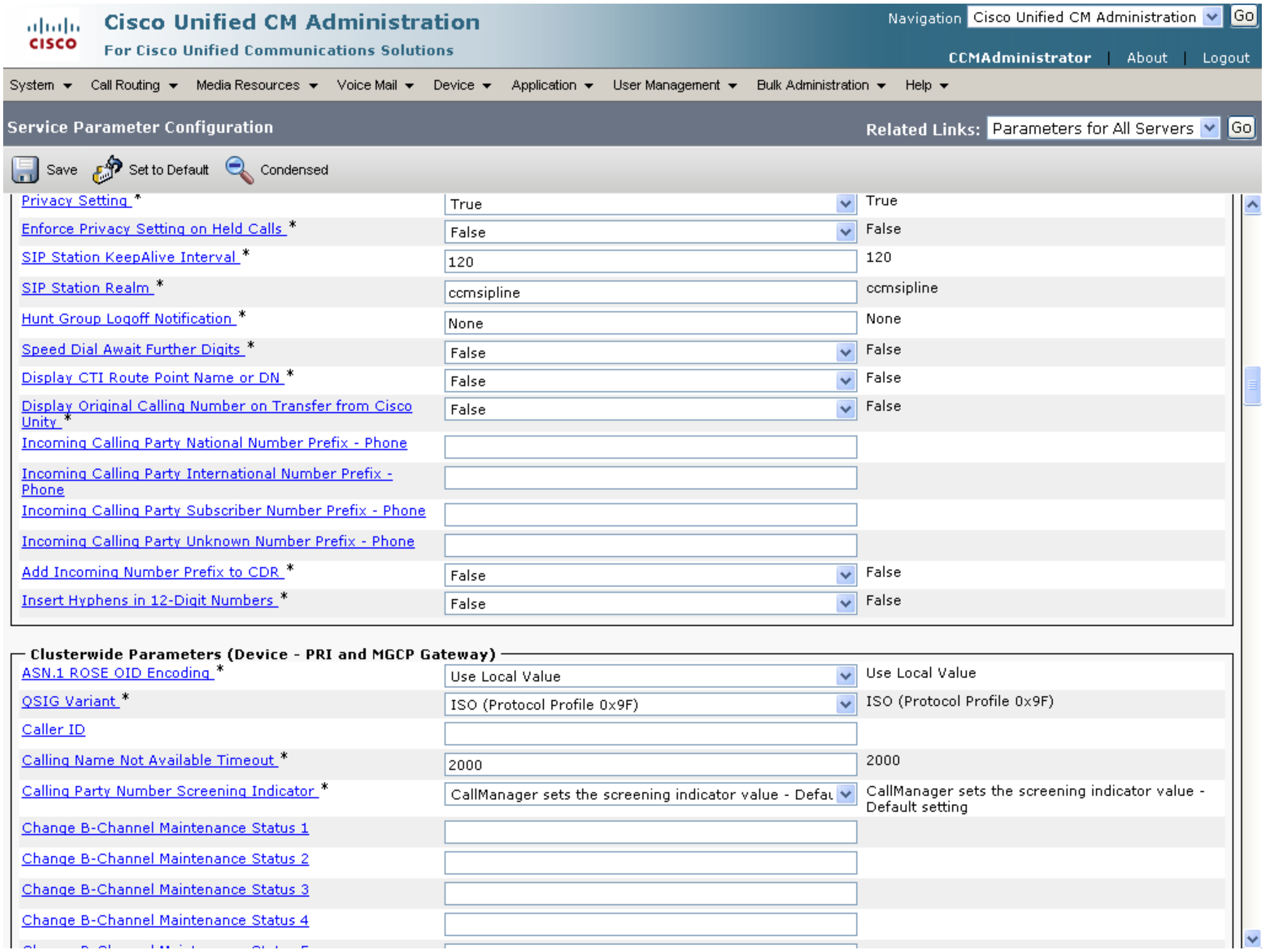
Clusterwide Parameters (Device - Phone)

Always Use Prime Line *	False	False
Always Use Prime Line for Voice Message *	False	False
Builtin Bridge Enable *	Off	Off
Device Mobility Mode *	Off	Off
Auto Answer Timer *	1	1
Extension Display on Cisco IP Phone Model 7910 *	False	False
Alternate Idle Phone Auto-Answer Behavior Enabled *	False	False
Hold Type *	False	False
Line State Update Enabled *	True	True
Off-hook to First Digit Timer *	15000	15000
Override Auto Answer If Speaker Is Disabled *	True	True
Out-of-Bandwidth Text *	Not Enough Bandwidth	Not Enough Bandwidth
Forced Authorization Code Prompt Text *	Enter Authorization Code	Enter Authorization Code
Client Matter Code Prompt Text *	Enter Client Matter Code	Enter Client Matter Code
AAR Network Congestion Rerouting Text *	Network Congestion. Rerouting.	Network Congestion. Rerouting.
Ring Setting of Busy Station Policy *	Only Apply Ring Setting of Busy Station When Incomin	Only Apply Ring Setting of Busy Station When Incoming Call Arrives
Transfer On-hook Enabled *	False	False
Ring Setting of Busy Station *	Beep Only	Beep Only
Ring Setting of Idle Station *	Ring	Ring
Call Pickup Group Audio Alert Setting of Idle Station *	Ring Once	Ring Once
Call Pickup Group Audio Alert Setting of Busy Station *	Beep Only	Beep Only
Privacy Setting *	True	True
Enforce Privacy Setting on Held Calls *	False	False
SIP Station KeepAlive Interval *	120	120
SIP Station Realm *	ccmsipline	ccmsipline

Service Parameters – 5 of 19.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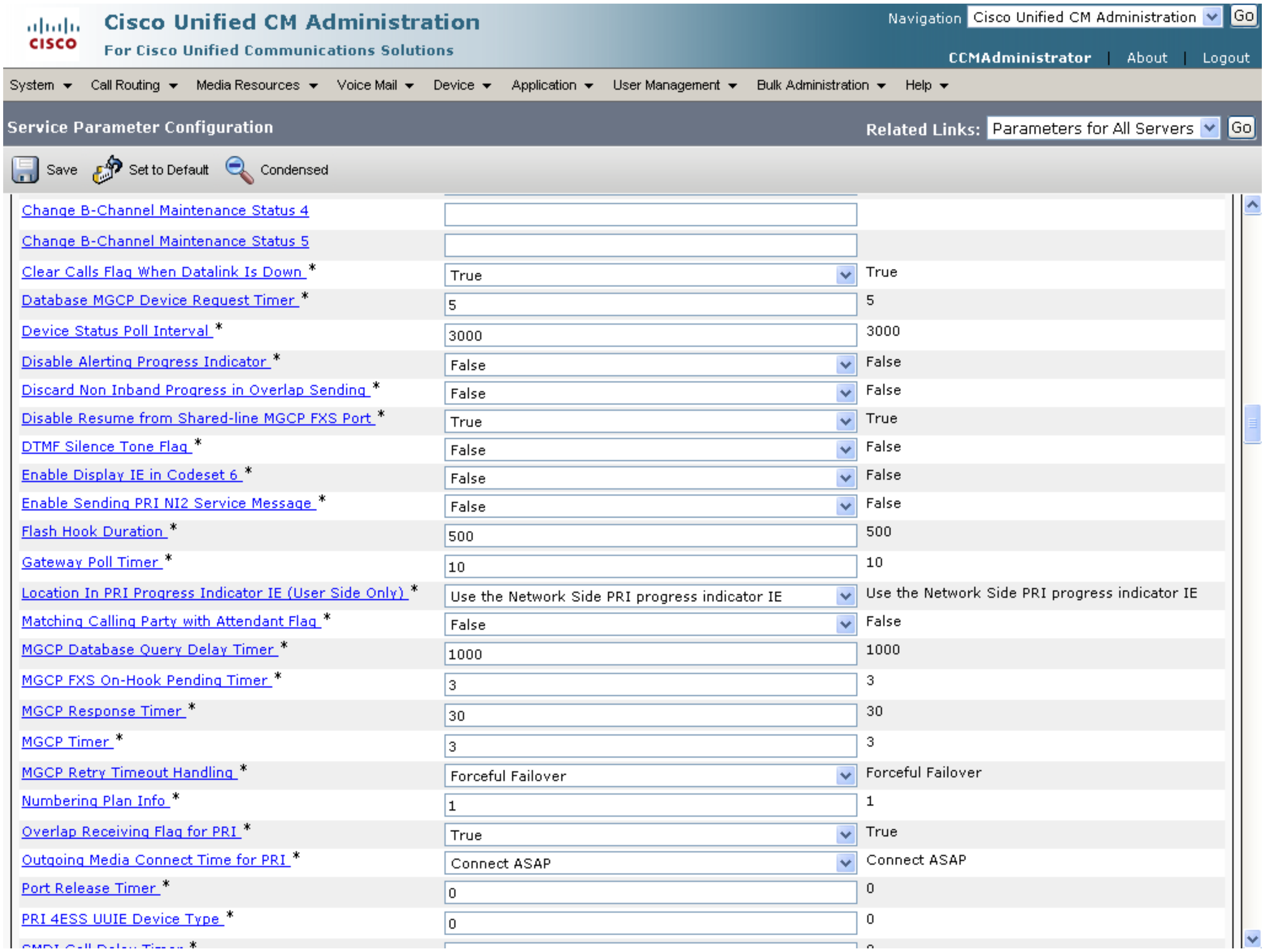
Parameter	Value	Default
Privacy Setting *	True	True
Enforce Privacy Setting on Held Calls *	False	False
SIP Station KeepAlive Interval *	120	120
SIP Station Realm *	ccmsipline	ccmsipline
Hunt Group Logoff Notification *	None	None
Speed Dial Await Further Digits *	False	False
Display CTI Route Point Name or DN *	False	False
Display Original Calling Number on Transfer from Cisco Unity *	False	False
Incoming Calling Party National Number Prefix - Phone		
Incoming Calling Party International Number Prefix - Phone		
Incoming Calling Party Subscriber Number Prefix - Phone		
Incoming Calling Party Unknown Number Prefix - Phone		
Add Incoming Number Prefix to CDR *	False	False
Insert Hyphens in 12-Digit Numbers *	False	False

Clusterwide Parameters (Device - PRI and MGCP Gateway)

Parameter	Value	Default
ASN.1 ROSE OID Encoding *	Use Local Value	Use Local Value
QSIG Variant *	ISO (Protocol Profile 0x9F)	ISO (Protocol Profile 0x9F)
Caller ID		
Calling Name Not Available Timeout *	2000	2000
Calling Party Number Screening Indicator *	CallManager sets the screening indicator value - Defau	CallManager sets the screening indicator value - Default setting
Change B-Channel Maintenance Status 1		
Change B-Channel Maintenance Status 2		
Change B-Channel Maintenance Status 3		
Change B-Channel Maintenance Status 4		

Service Parameters – 6 of 19.



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Change B-Channel Maintenance Status 4		
Change B-Channel Maintenance Status 5		
Clear Calls Flag When Datalink Is Down *	True	True
Database MGCP Device Request Timer *	5	5
Device Status Poll Interval *	3000	3000
Disable Alerting Progress Indicator *	False	False
Discard Non Inband Progress in Overlap Sending *	False	False
Disable Resume from Shared-line MGCP FXS Port *	True	True
DTMF Silence Tone Flag *	False	False
Enable Display IE in Codeset 6 *	False	False
Enable Sending PRI NI2 Service Message *	False	False
Flash Hook Duration *	500	500
Gateway Poll Timer *	10	10
Location In PRI Progress Indicator IE (User Side Only) *	Use the Network Side PRI progress indicator IE	Use the Network Side PRI progress indicator IE
Matching Calling Party with Attendant Flag *	False	False
MGCP Database Query Delay Timer *	1000	1000
MGCP FXS On-Hook Pending Timer *	3	3
MGCP Response Timer *	30	30
MGCP Timer *	3	3
MGCP Retry Timeout Handling *	Forceful Failover	Forceful Failover
Numbering Plan Info *	1	1
Overlap Receiving Flag for PRI *	True	True
Outgoing Media Connect Time for PRI *	Connect ASAP	Connect ASAP
Port Release Timer *	0	0
PRI 4ESS UUIE Device Type *	0	0

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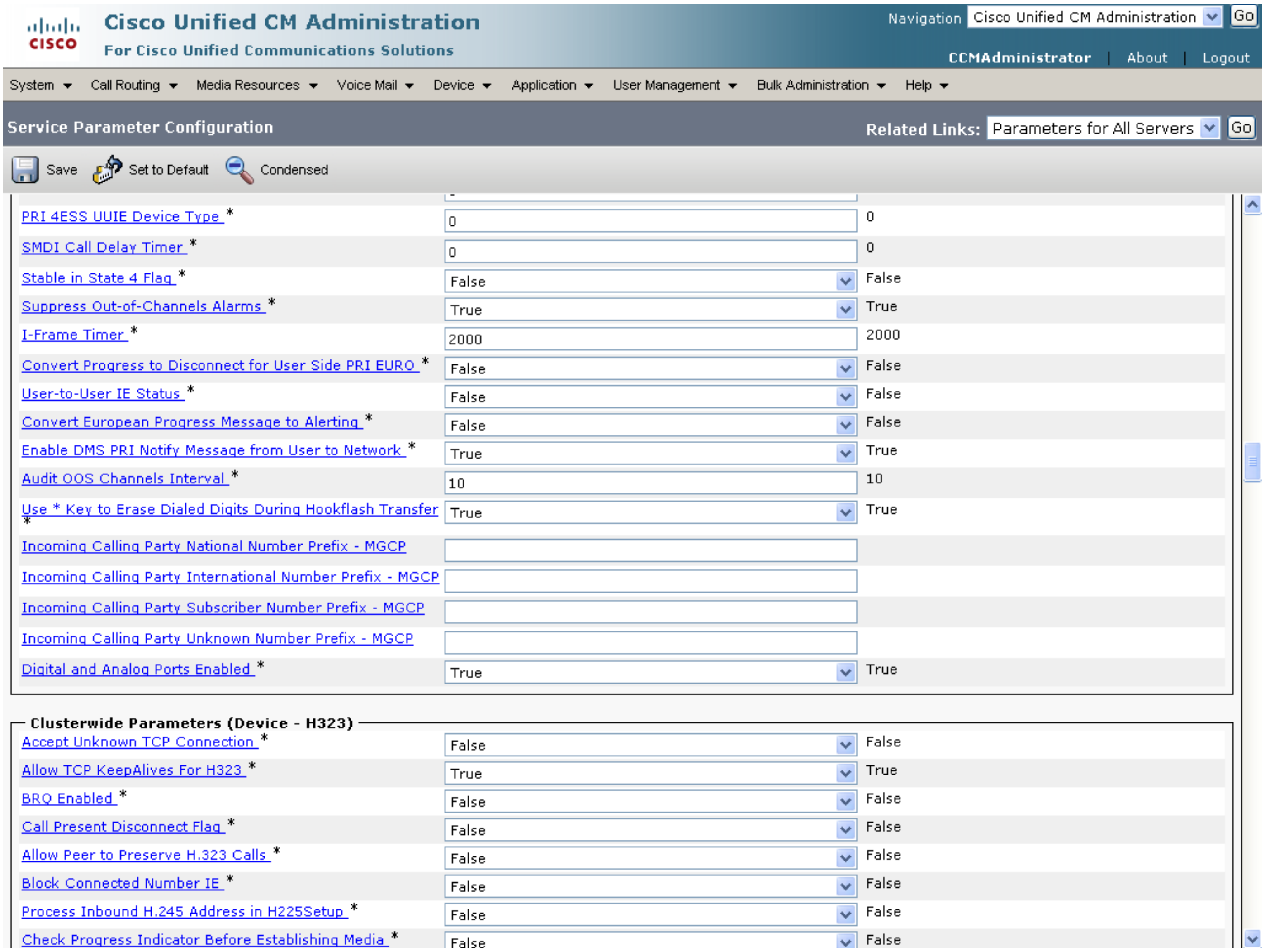

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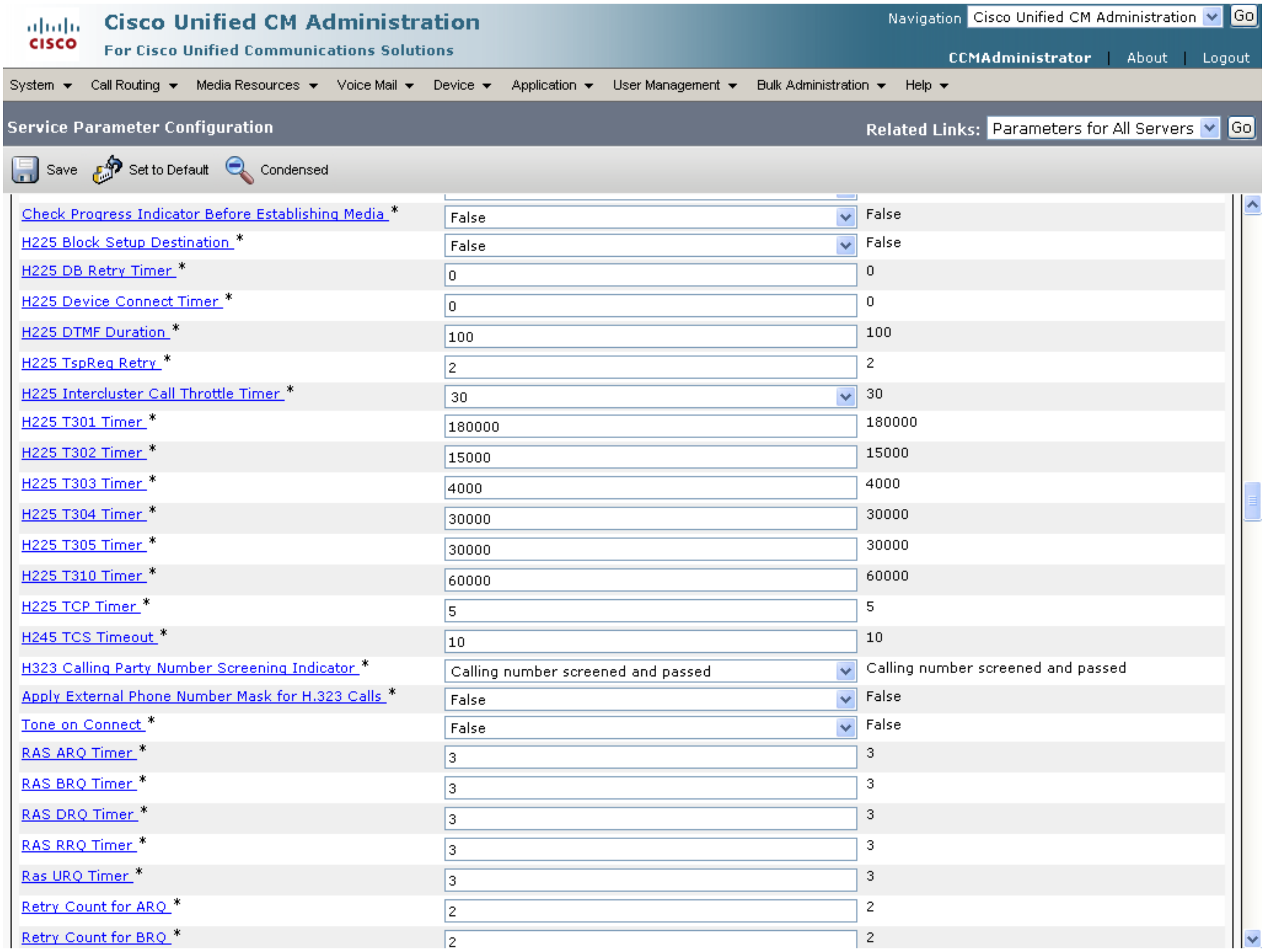

Check Progress Indicator Before Establishing Media *	False	False
H225 Block Setup Destination *	False	False
H225 DB Retry Timer *	0	0
H225 Device Connect Timer *	0	0
H225 DTMF Duration *	100	100
H225 TspReq Retry *	2	2
H225 Intercluster Call Throttle Timer *	30	30
H225 T301 Timer *	180000	180000
H225 T302 Timer *	15000	15000
H225 T303 Timer *	4000	4000
H225 T304 Timer *	30000	30000
H225 T305 Timer *	30000	30000
H225 T310 Timer *	60000	60000
H225 TCP Timer *	5	5
H245 TCS Timeout *	10	10
H323 Calling Party Number Screening Indicator *	Calling number screened and passed	Calling number screened and passed
Apply External Phone Number Mask for H.323 Calls *	False	False
Tone on Connect *	False	False
RAS ARQ Timer *	3	3
RAS BRQ Timer *	3	3
RAS DRQ Timer *	3	3
RAS RRQ Timer *	3	3
Ras URQ Timer *	3	3
Retry Count for ARQ *	2	2
Retry Count for BRQ *	2	2

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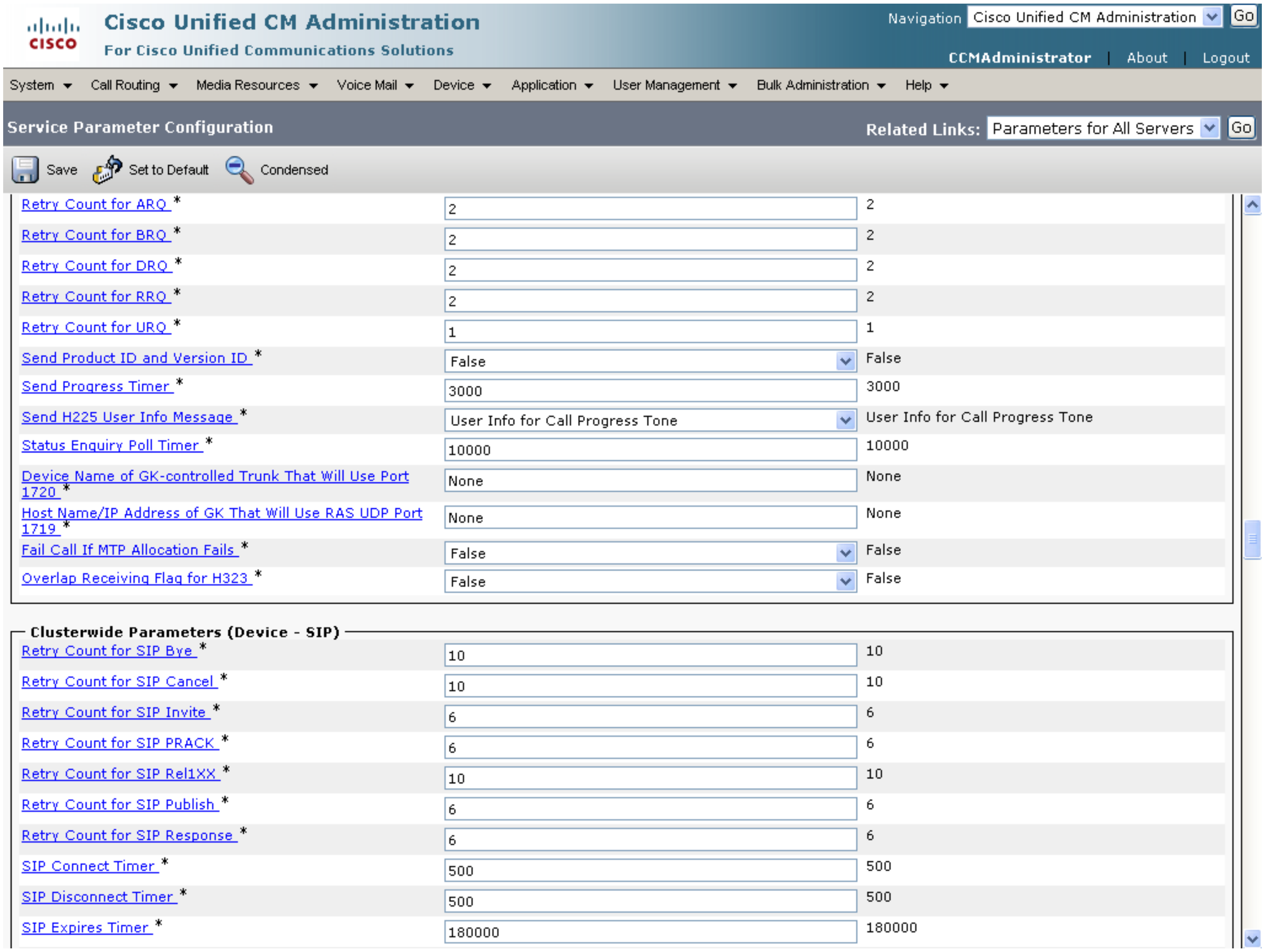

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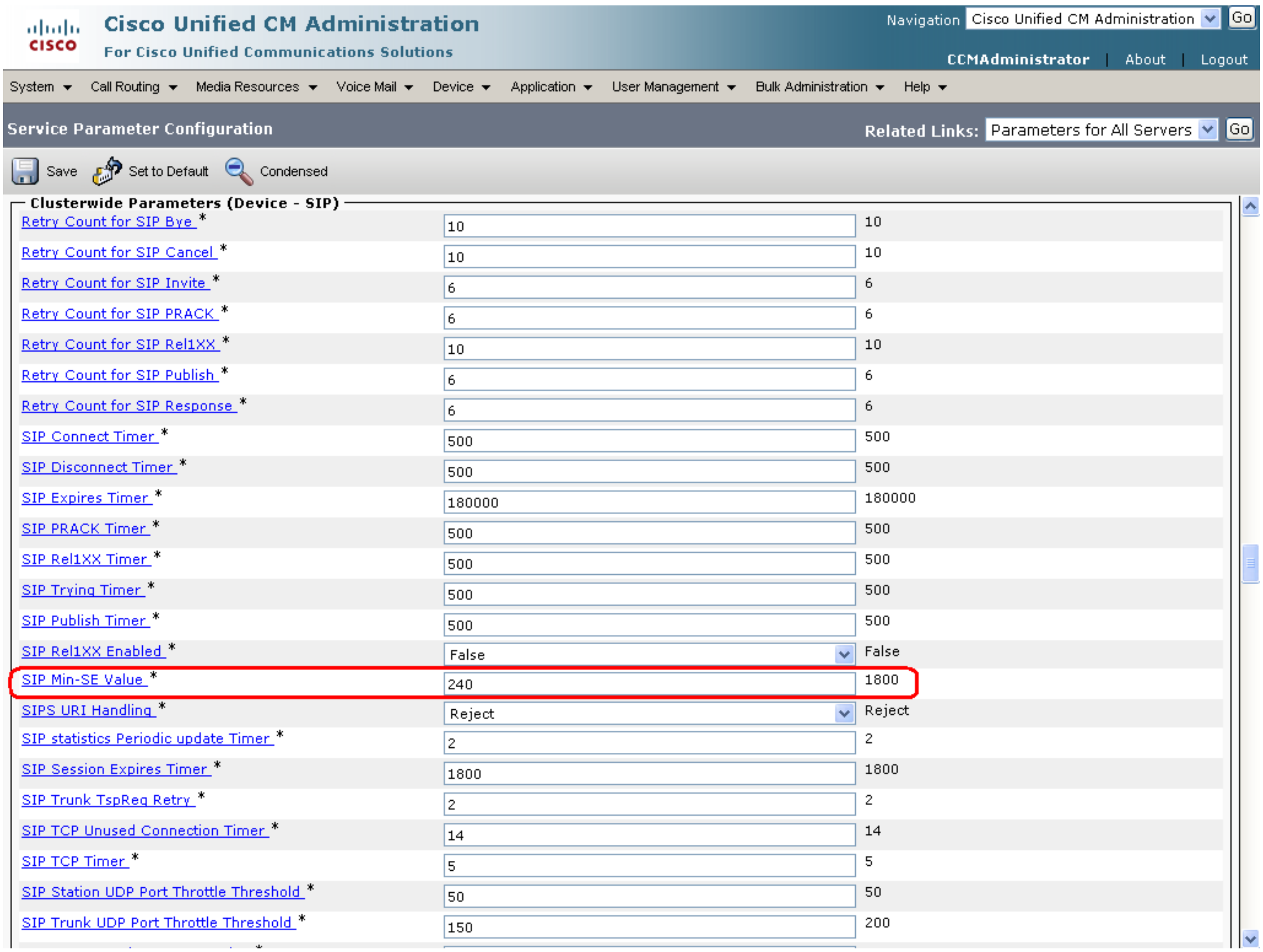
Parameter	Value	Default
Retry Count for ARQ *	2	2
Retry Count for BRQ *	2	2
Retry Count for DRQ *	2	2
Retry Count for RRQ *	2	2
Retry Count for URQ *	1	1
Send Product ID and Version ID *	False	False
Send Progress Timer *	3000	3000
Send H225 User Info Message *	User Info for Call Progress Tone	User Info for Call Progress Tone
Status Enquiry Poll Timer *	10000	10000
Device Name of GK-controlled Trunk That Will Use Port 1720 *	None	None
Host Name/IP Address of GK That Will Use RAS UDP Port 1719 *	None	None
Fail Call If MTP Allocation Fails *	False	False
Overlap Receiving Flag for H323 *	False	False

Clusterwide Parameters (Device - SIP)

Parameter	Value	Default
Retry Count for SIP Bye *	10	10
Retry Count for SIP Cancel *	10	10
Retry Count for SIP Invite *	6	6
Retry Count for SIP PRACK *	6	6
Retry Count for SIP Rel1XX *	10	10
Retry Count for SIP Publish *	6	6
Retry Count for SIP Response *	6	6
SIP Connect Timer *	500	500
SIP Disconnect Timer *	500	500
SIP Expires Timer *	180000	180000

Service Parameters – 10 of 19.

Note: The Mitel 3300 ICP SIP Session Timer value must equal or exceed the Cisco Unified Communications Manager 6.1 SIP Min-SE Value. The range for the Mitel SIP Session Timer is 90 – 9999 seconds, and the default is 90 seconds. The range for the Cisco Unified Communications Manager 6.1 SIP Min-SE Value is 60 – 86400 seconds, and the default is 1800 seconds. The value shown here is 240 seconds, because that reflects the value on Cisco Unified Communications Manager 6.1 at the time of testing. However, if the Cisco Unified Communications Manager 6.1 SIP Min-SE Value had been set to default, the Mitel SIP Session Timer value would have been set to 1800 – 9999. In general, it is recommended to follow the default values on Cisco Unifed Communications Manager Service Parameters. See note on page 11.



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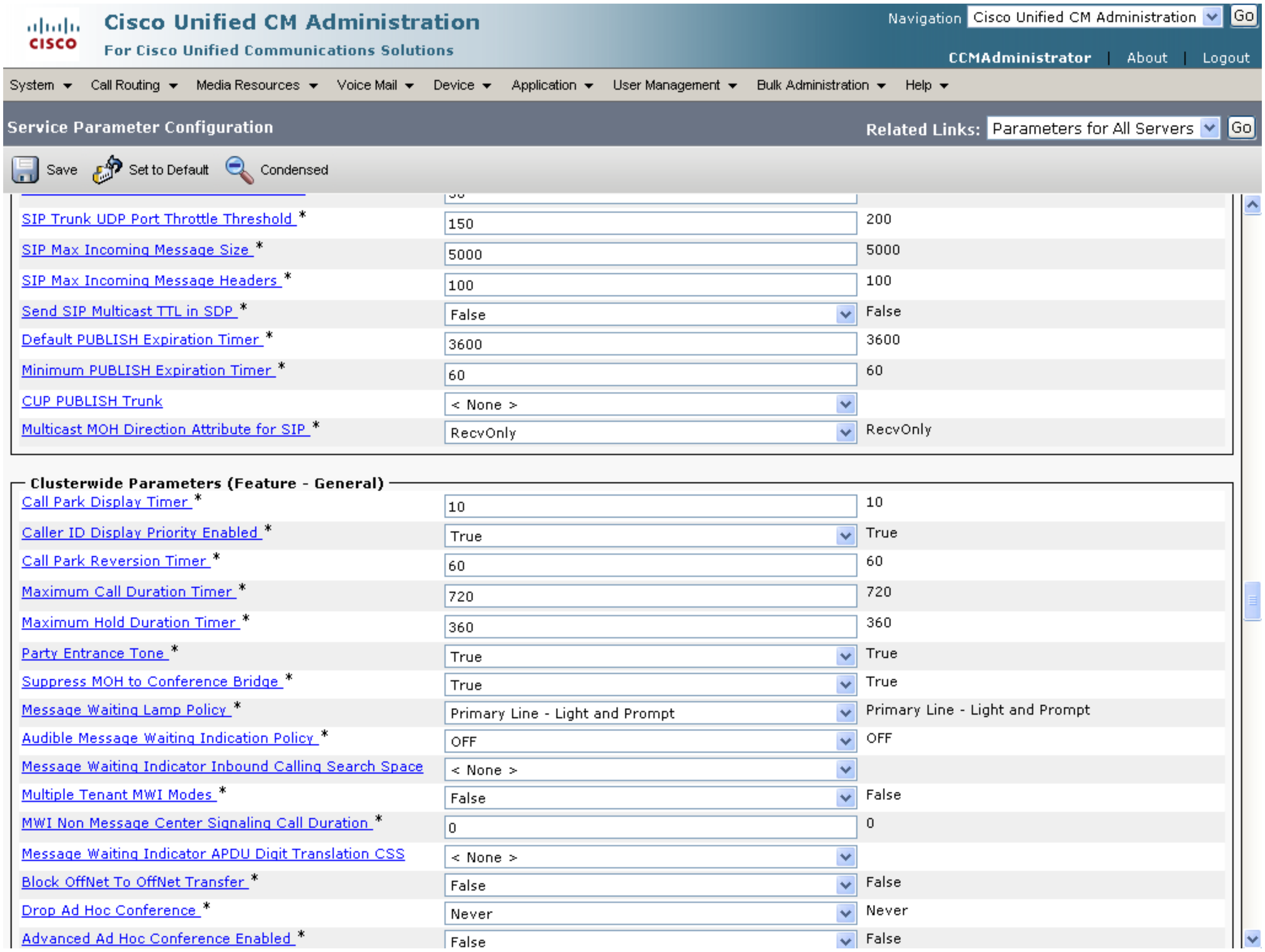

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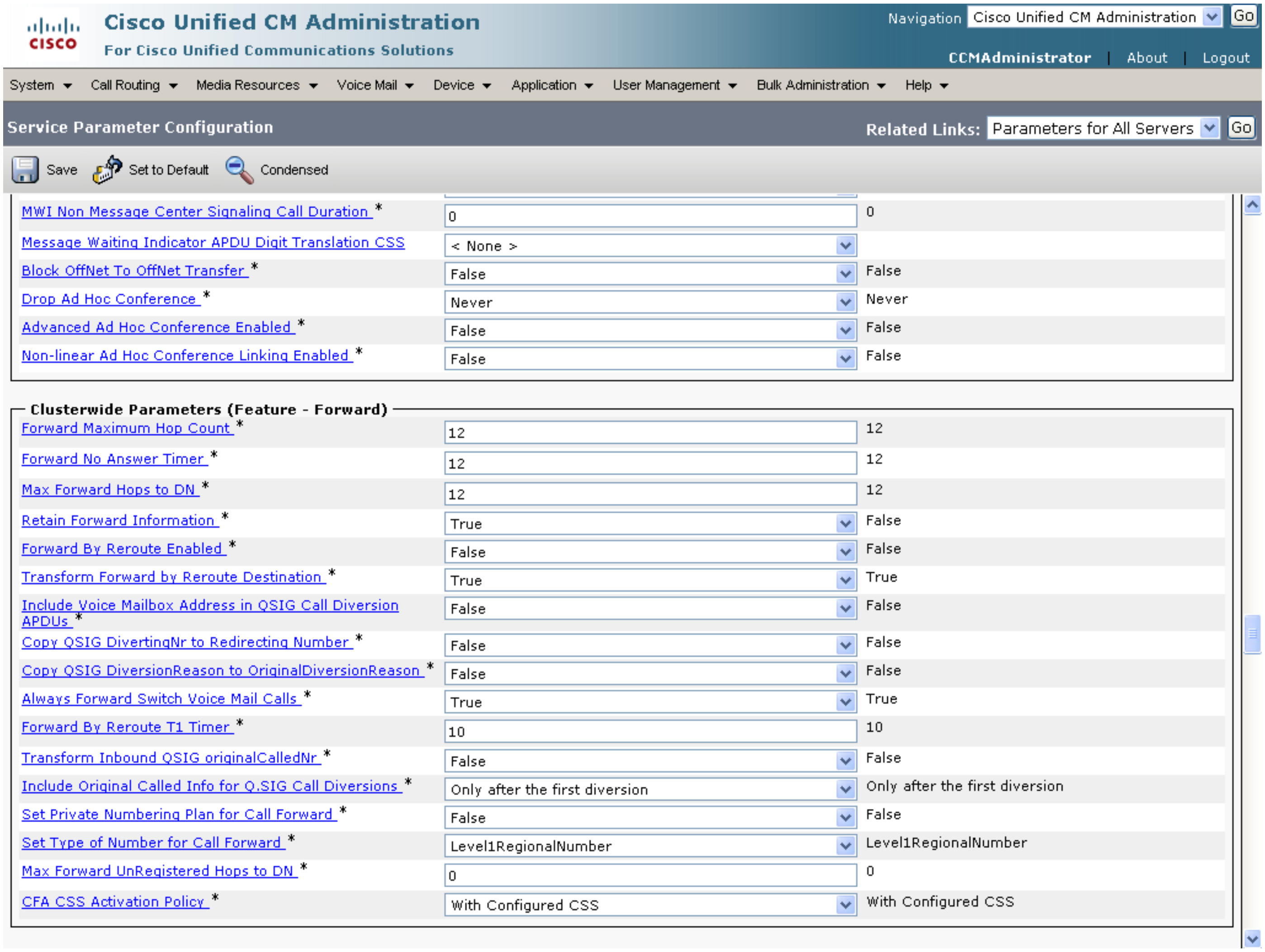

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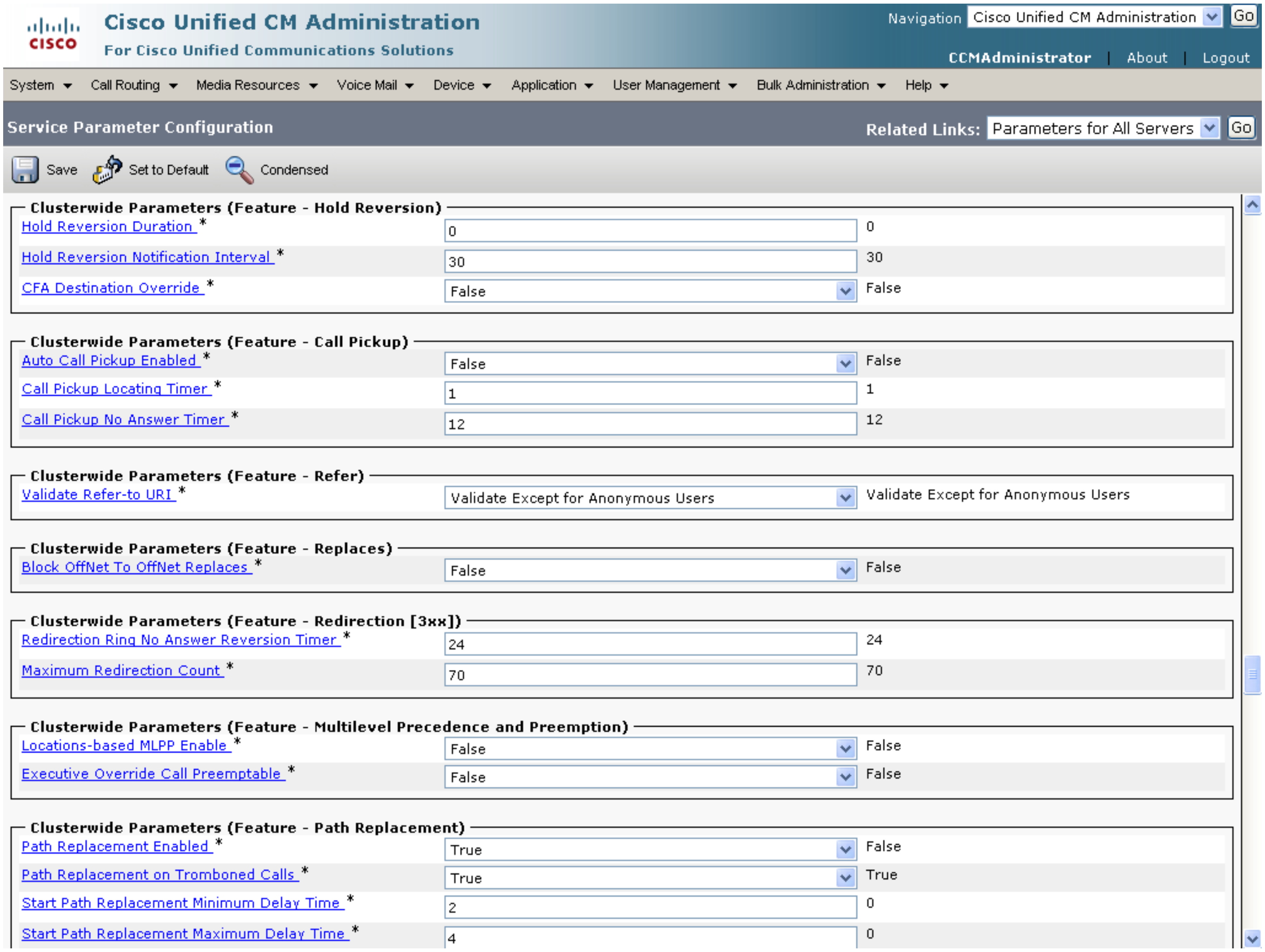
Parameter	Value	Default
MWI Non Message Center Signaling Call Duration *	0	0
Message Waiting Indicator APDU Digit Translation CSS	< None >	
Block OffNet To OffNet Transfer *	False	False
Drop Ad Hoc Conference *	Never	Never
Advanced Ad Hoc Conference Enabled *	False	False
Non-linear Ad Hoc Conference Linking Enabled *	False	False

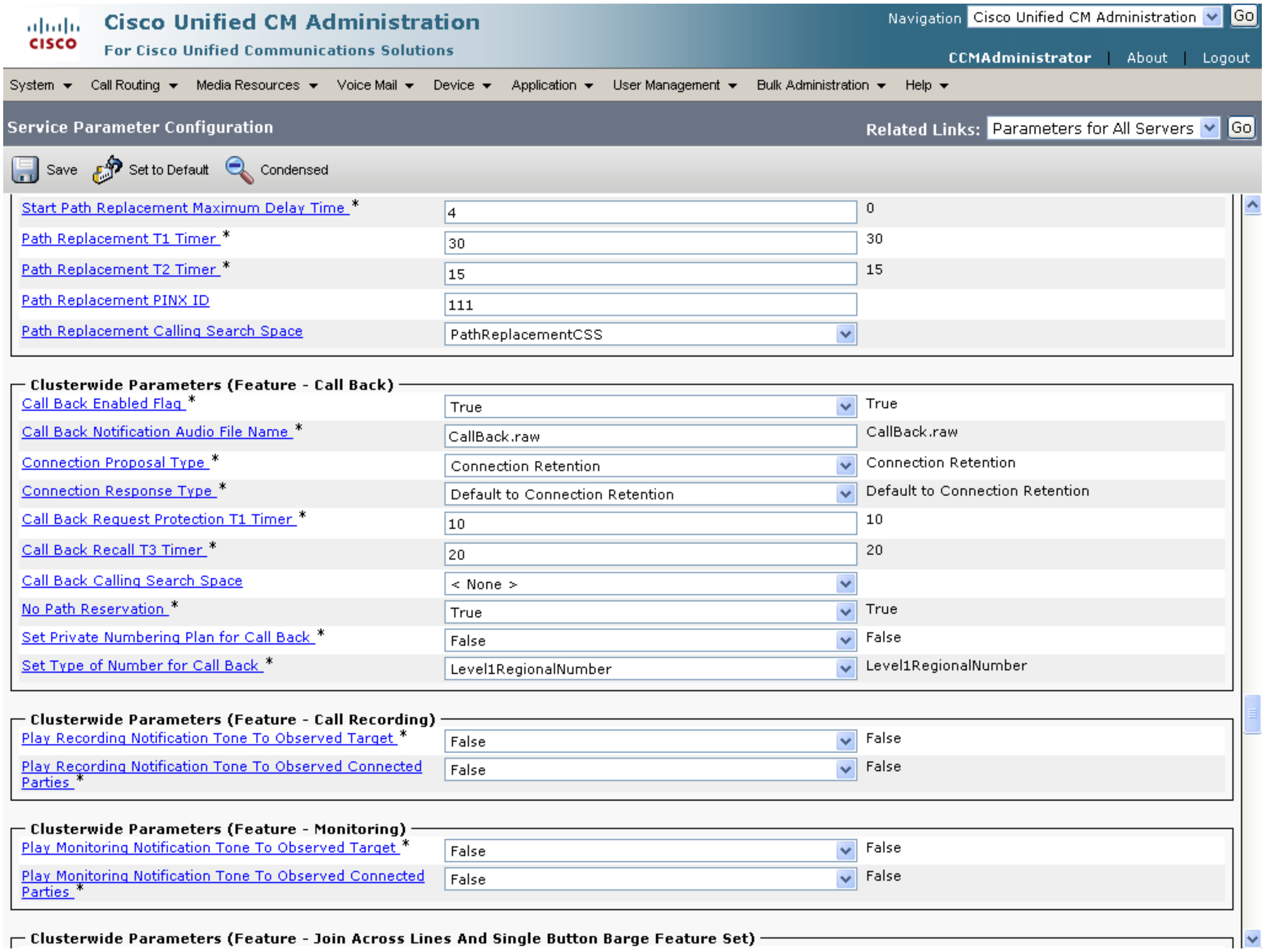
Clusterwide Parameters (Feature - Forward)

Parameter	Value	Default
Forward Maximum Hop Count *	12	12
Forward No Answer Timer *	12	12
Max Forward Hops to DN *	12	12
Retain Forward Information *	True	False
Forward By Reroute Enabled *	False	False
Transform Forward by Reroute Destination *	True	True
Include Voice Mailbox Address in QSIG Call Diversion APDUs *	False	False
Copy QSIG DivertingNr to Redirecting Number *	False	False
Copy QSIG DiversionReason to OriginalDiversionReason *	False	False
Always Forward Switch Voice Mail Calls *	True	True
Forward By Reroute T1 Timer *	10	10
Transform Inbound QSIG originalCalledNr *	False	False
Include Original Called Info for Q.SIG Call Diversions *	Only after the first diversion	Only after the first diversion
Set Private Numbering Plan for Call Forward *	False	False
Set Type of Number for Call Forward *	Level1RegionalNumber	Level1RegionalNumber
Max Forward UnRegistered Hops to DN *	0	0
CFA CSS Activation Policy *	With Configured CSS	With Configured CSS

Service Parameters – 13 of 19.



Service Parameters – 14 of 19.



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Parameter	Value	Default
Start Path Replacement Maximum Delay Time *	4	0
Path Replacement T1 Timer *	30	30
Path Replacement T2 Timer *	15	15
Path Replacement PINX ID	111	
Path Replacement Calling Search Space	PathReplacementCSS	

Clusterwide Parameters (Feature - Call Back)

Parameter	Value	Default
Call Back Enabled Flag *	True	True
Call Back Notification Audio File Name *	CallBack.raw	CallBack.raw
Connection Proposal Type *	Connection Retention	Connection Retention
Connection Response Type *	Default to Connection Retention	Default to Connection Retention
Call Back Request Protection T1 Timer *	10	10
Call Back Recall T3 Timer *	20	20
Call Back Calling Search Space	< None >	
No Path Reservation *	True	True
Set Private Numbering Plan for Call Back *	False	False
Set Type of Number for Call Back *	Level1RegionalNumber	Level1RegionalNumber

Clusterwide Parameters (Feature - Call Recording)

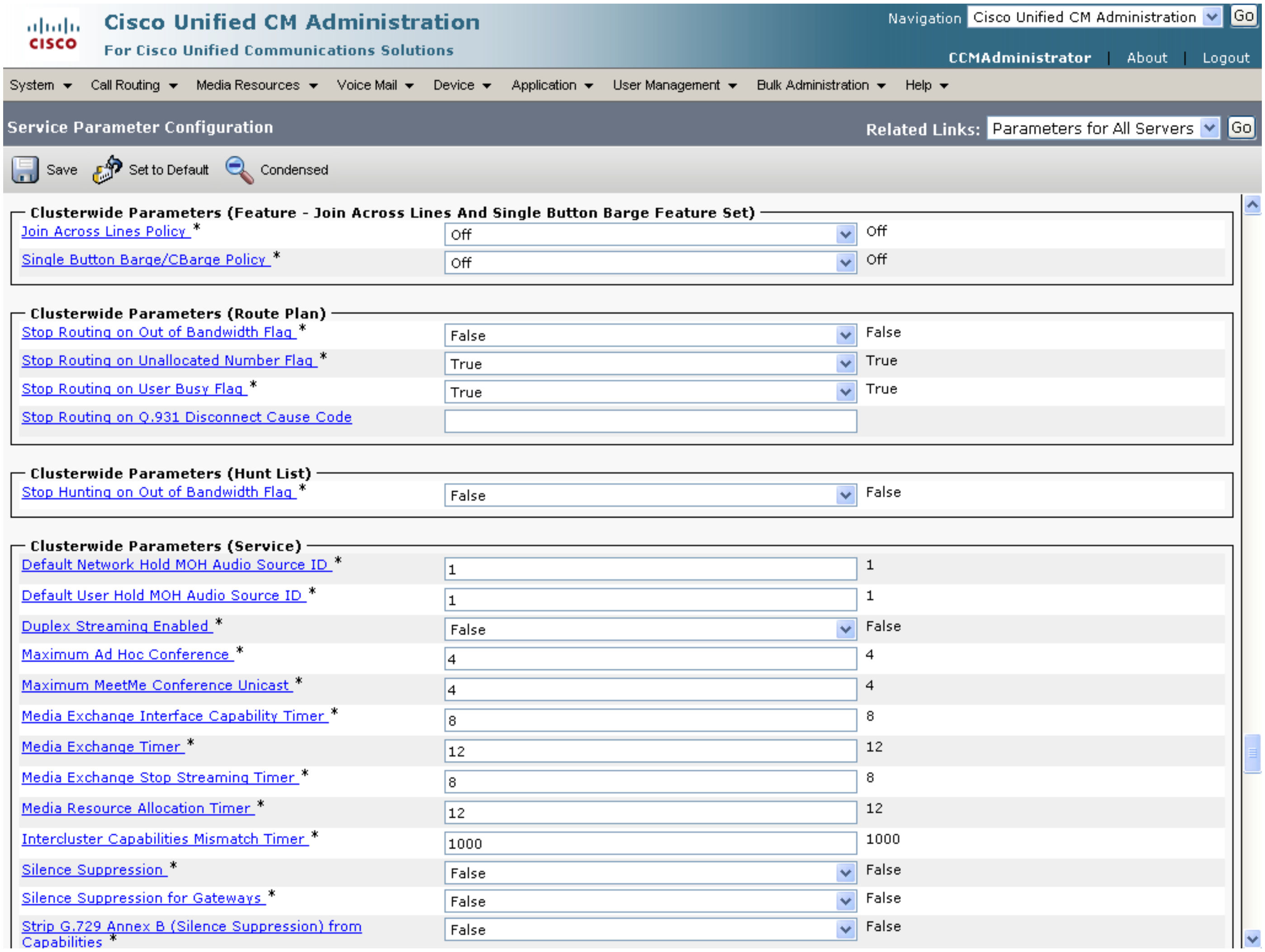
Parameter	Value	Default
Play Recording Notification Tone To Observed Target *	False	False
Play Recording Notification Tone To Observed Connected Parties *	False	False

Clusterwide Parameters (Feature - Monitoring)

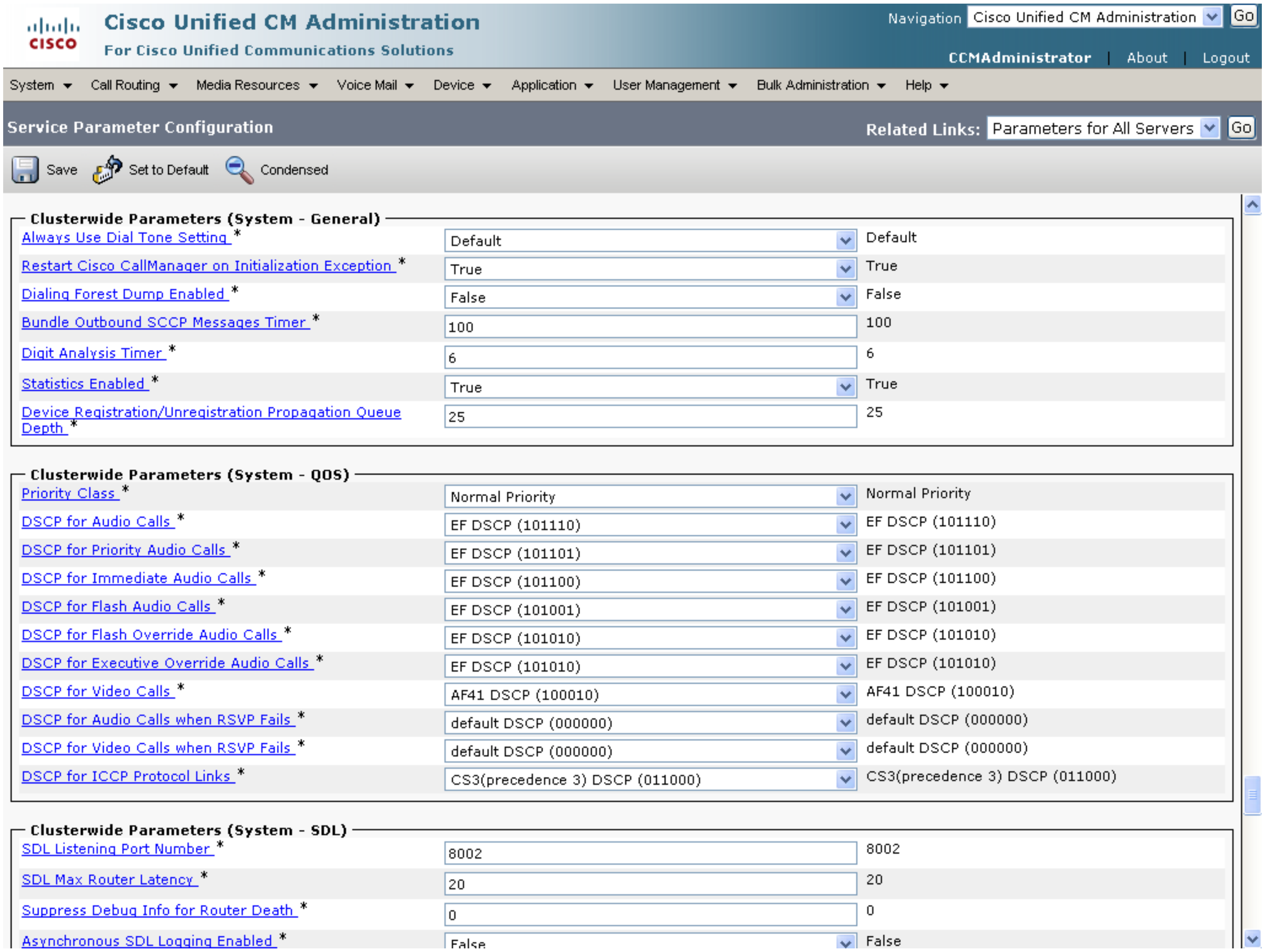
Parameter	Value	Default
Play Monitoring Notification Tone To Observed Target *	False	False
Play Monitoring Notification Tone To Observed Connected Parties *	False	False

Clusterwide Parameters (Feature - Join Across Lines And Single Button Barge Feature Set)

Service Parameters – 15 of 19.



Service Parameters – 16 of 19.



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Clusterwide Parameters (System - General)

Always Use Dial Tone Setting *	Default	Default
Restart Cisco CallManager on Initialization Exception *	True	True
Dialing Forest Dump Enabled *	False	False
Bundle Outbound SCCP Messages Timer *	100	100
Digit Analysis Timer *	6	6
Statistics Enabled *	True	True
Device Registration/Unregistration Propagation Queue Depth *	25	25

Clusterwide Parameters (System - QOS)

Priority Class *	Normal Priority	Normal Priority
DSCP for Audio Calls *	EF DSCP (101110)	EF DSCP (101110)
DSCP for Priority Audio Calls *	EF DSCP (101101)	EF DSCP (101101)
DSCP for Immediate Audio Calls *	EF DSCP (101100)	EF DSCP (101100)
DSCP for Flash Audio Calls *	EF DSCP (101001)	EF DSCP (101001)
DSCP for Flash Override Audio Calls *	EF DSCP (101010)	EF DSCP (101010)
DSCP for Executive Override Audio Calls *	EF DSCP (101010)	EF DSCP (101010)
DSCP for Video Calls *	AF41 DSCP (100010)	AF41 DSCP (100010)
DSCP for Audio Calls when RSVP Fails *	default DSCP (000000)	default DSCP (000000)
DSCP for Video Calls when RSVP Fails *	default DSCP (000000)	default DSCP (000000)
DSCP for ICCP Protocol Links *	CS3(precedence 3) DSCP (011000)	CS3(precedence 3) DSCP (011000)

Clusterwide Parameters (System - SDL)

SDL Listening Port Number *	8002	8002
SDL Max Router Latency *	20	20
Suppress Debug Info for Router Death *	0	0
Asynchronous SDL Logging Enabled *	False	False

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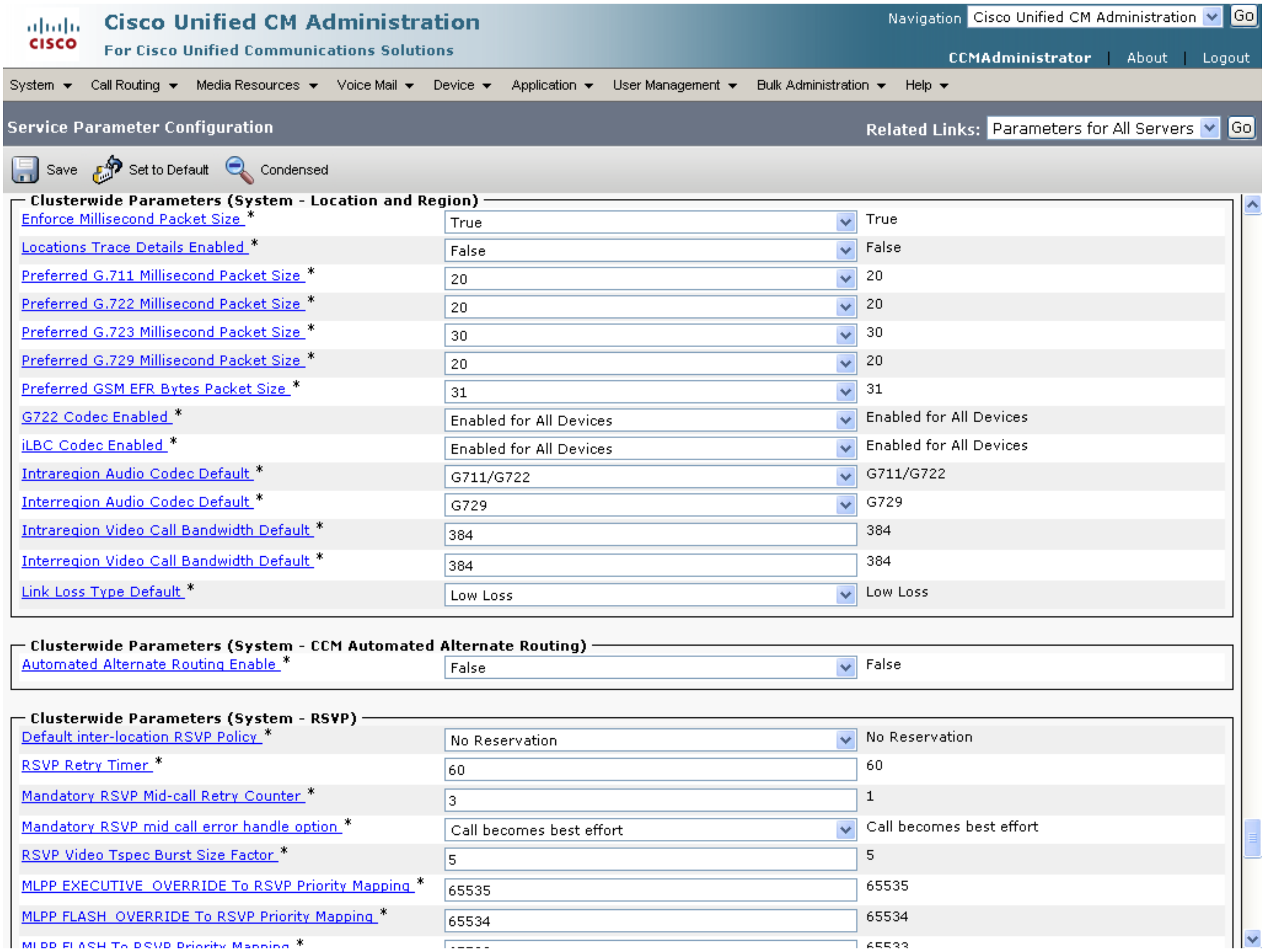

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Clusterwide Parameters (System - Location and Region)

Parameter	Value	Default
Enforce Millisecond Packet Size *	True	True
Locations Trace Details Enabled *	False	False
Preferred G.711 Millisecond Packet Size *	20	20
Preferred G.722 Millisecond Packet Size *	20	20
Preferred G.723 Millisecond Packet Size *	30	30
Preferred G.729 Millisecond Packet Size *	20	20
Preferred GSM EFR Bytes Packet Size *	31	31
G722 Codec Enabled *	Enabled for All Devices	Enabled for All Devices
iLBC Codec Enabled *	Enabled for All Devices	Enabled for All Devices
Intraregion Audio Codec Default *	G711/G722	G711/G722
Interregion Audio Codec Default *	G729	G729
Intraregion Video Call Bandwidth Default *	384	384
Interregion Video Call Bandwidth Default *	384	384
Link Loss Type Default *	Low Loss	Low Loss

Clusterwide Parameters (System - CCM Automated Alternate Routing)

Parameter	Value	Default
Automated Alternate Routing Enable *	False	False

Clusterwide Parameters (System - RSVP)

Parameter	Value	Default
Default inter-location RSVP Policy *	No Reservation	No Reservation
RSVP Retry Timer *	60	60
Mandatory RSVP Mid-call Retry Counter *	3	1
Mandatory RSVP mid call error handle option *	Call becomes best effort	Call becomes best effort
RSVP Video Tspec Burst Size Factor *	5	5
MLPP EXECUTIVE_OVERRIDE To RSVP Priority Mapping *	65535	65535
MLPP FLASH_OVERRIDE To RSVP Priority Mapping *	65534	65534
MLPP FLASH To RSVP Priority Mapping *	[illegible]	65533

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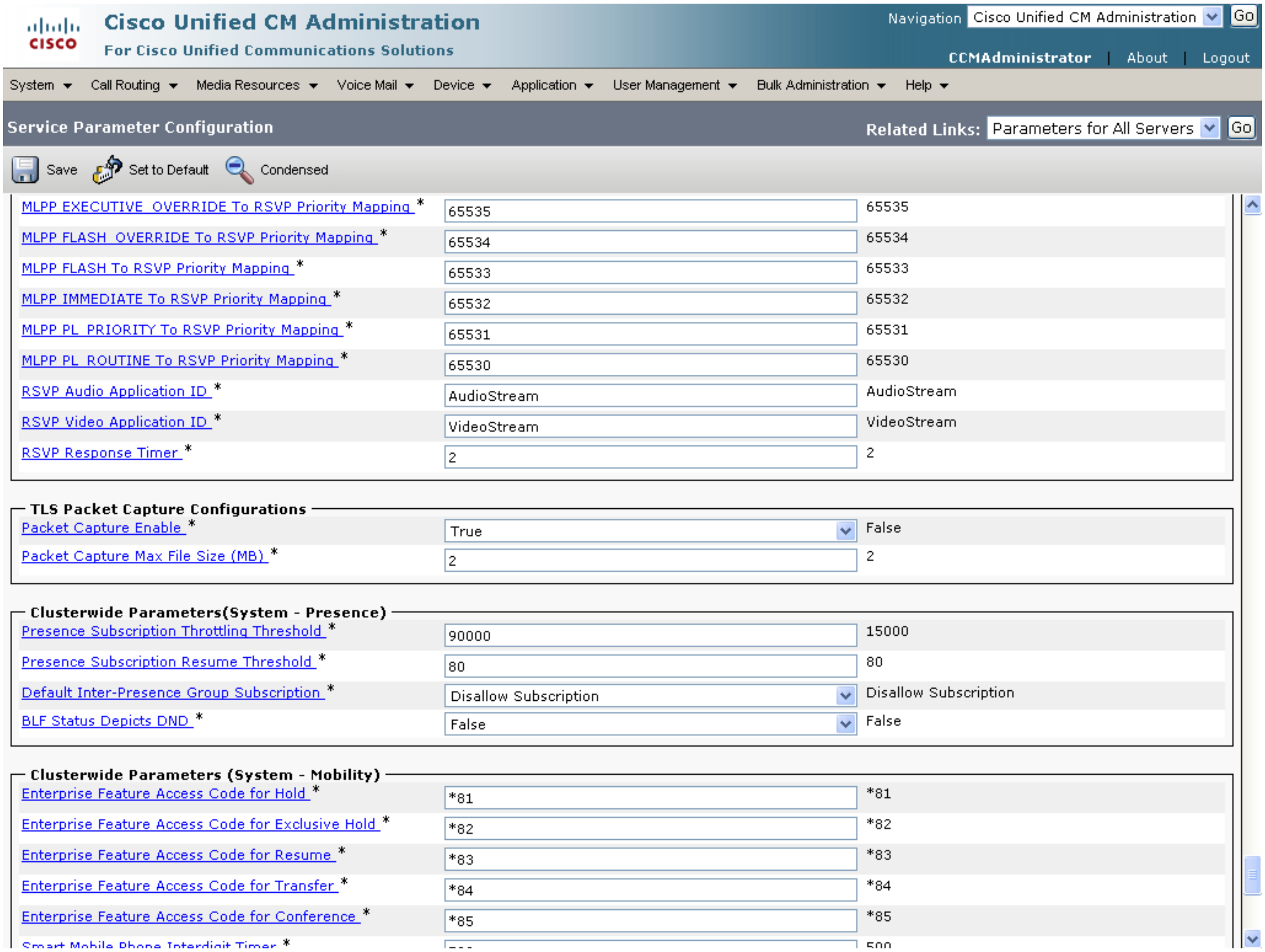

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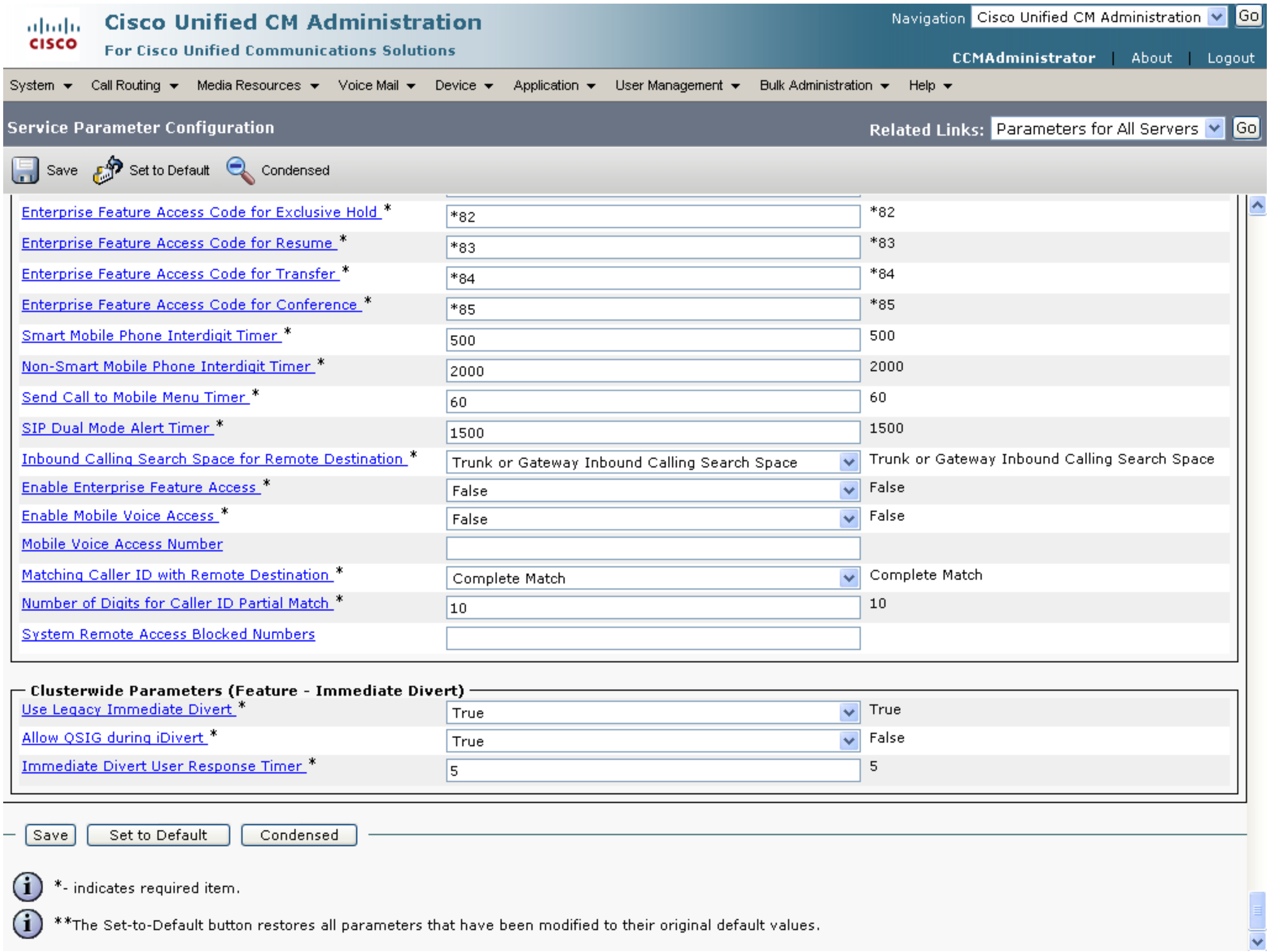
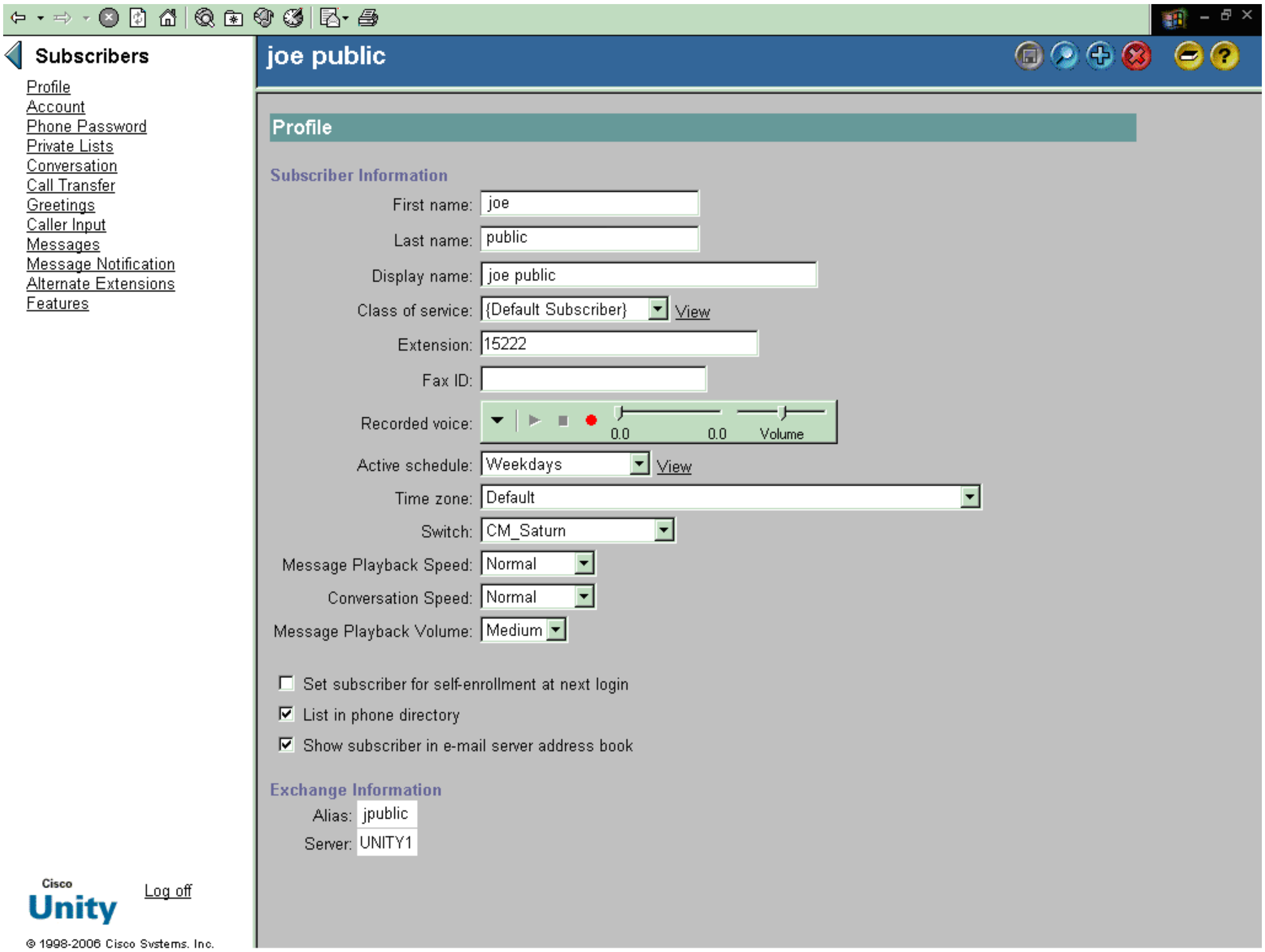
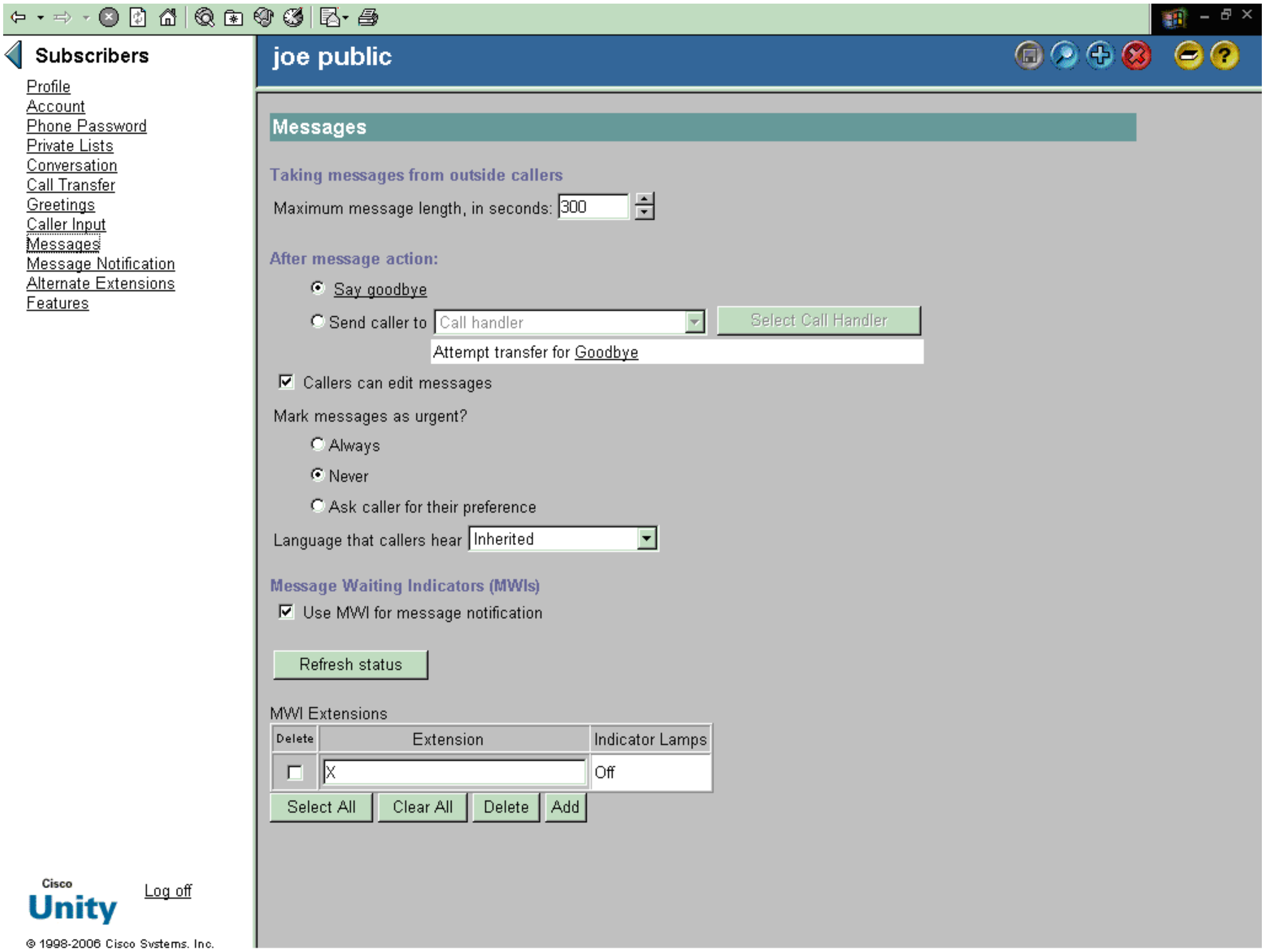


Configuring Mailboxes on Cisco Unity

Subscriber Profile – 1 of 1.



Subscriber Messages – 1 of 1.



Acronyms

Acronym	Definition
CUCM	Cisco Unified Communications Manager
CCBS	Call Completion to Busy Subscriber
CCNR	Call Completion on No Reply
CFB	Call Forwarding on Busy
CFNR	Call Forwarding No Reply
CFU	Call Forwarding Unconditional
CLIP	Calling Line (Number) Identification Presentation
CLIR	Calling Line (Number) Identification Restriction
CNIP	Calling Name Identification Presentation
CNIR	Calling Name Identification Restriction
COLP	Connected Line (Number) Identification Presentation
COLR	Connected Line (Number) Identification Restriction
CONP	Connected Line (Name) Identification Presentation
CONR	Connected Line (Name) Identification Restriction
MWI	Message Waiting Indication
PAI	P-Asserted Identity
RPID	Remote Party ID
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol

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