

XO[®] SIP Service

Customer Configuration Guide for Cisco Unified Communications Manager (CUCM) 7.0.1

XO SIP Packages 1 and 2, implemented With Cisco Unified Border Control Element (CUBE)



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About This Document

This document describes interoperability between XO SIP Package 1 (G.711) and 2 (G.729a) and the Cisco Unified Communications Manager (CUCM) 7.0.1 IP PBX , implemented with Cisco Unified Border Control Element (CUBE), deployed with an XO-provided Cisco 2432 IAD as the router/demarcation device. This document assumes the audience has a general understanding of network provisioning and the connectivity requirements of XO Communications SIP service offering. A Cisco engineer should be consulted for assistance with CUBE specifications.

Known Issues

While XO certifies interoperability between XO SIP service and the IP PBX as outlined herein, the following known issues were identified during Interoperability testing. The customer should be aware that certain features and functions may not be fully supportable. In some cases, special configurations and/or PBX software patches may be available from the vendor.

Known Issues for XO SIP Services Packages 1 and 2 with CUBE:

- ***67 for Outgoing Calling Line ID Delivery Blocking Per Call is not supported.**
Optional Workaround: Create a separate SIP Trunk where in the Outbound Calls Section the Calling Line ID Presentation is set to Restricted. A route pattern is assigned to that SIP Trunk with a unique single digit access code to access that SIP Trunk which blocks the caller ID for all calls when requested by the user via the access code.
- **PSTN-to-PSTN call transfers using Call Forward Always (CFA), Call Forward On Busy (CFOB), and Call Forward No Answer (CFNA) when using 4-Digit CUCM extensions** - in order for these call scenarios to function, a the SIP profile in CUBE must be used to modify the diversion header to add the NPA-NXX to the 4 digit extension in the user portion of the diversion header. Please see section 2.11 for details.
- **Conferencing** – Conference Bridge resources rely on CUBE. A Cisco engineer should be consulted to determine the CUBE hardware required to support any customer conference bridge requirements in excess of a 4-way conference call.
- **Music on Hold Settings**
 - **CUCM Music on Hold Server Codec Setting** - when configuring the customer for SP1 or SP2, the Music on Hold Server codec must be set accordingly to G.711ulaw for SP1 and G.729 for SP2 as discussed in detail in section 2.7.
 - **CUBE SIP Profile Rules** - when using Music on Hold, some configuration commands are required in the CUBE sip profile section of the configuration for the remote party to hear Music on Hold when the call is placed on hold from a CUCM phone. This is discussed in detail in section 2.11.

Registration Method Static registration is utilized between the CUCM 7.0.1 IP PBX and the XO call agent.

XO SIP Service Packages Supported

Pkg	Codec	DTMF	Fax
1	G.711	RFC2833 (in-band RTP DTMF fall-back)	T.38; G.711 pass-through
2	G.729a	RFC2833	T.38; G.711 pass-through is <u>not</u> supported

1. Testing of CUCM 7.0.1

1.1. Software and Hardware Versions Tested

1. CUCM Server
 - a. Hardware: Cisco MCS 7800 Series Product No MCS7825H3-K9-CMB2
 - b. Software Version: CUCM 7.0.1.11000-2
2. Cisco 2821 ISR running CUBE software
 - a. Hardware: Cisco 2821 ISR
 - b. 1 PVDM2-32 and 2 PVDM2-48 DSP modules
 - c. IOS software version "c2800nm-ipvoice_ivs-mz.124-24.T.bin"
3. Cisco Unity Connection (CUC) Voice Mail Server
 - a. Hardware: Cisco MCS 7800 Series Product No MCS7825H3-K9-CMB2
 - b. Software Version: CUC 7.0.1.11000-13-2
4. CUCM and Cisco Unity Connection (CUC) PC GUI Access
 - a. Microsoft Internet Explorer version 6.0.2900.2180.xpsp_sp2_gdr.080814-1233
5. Cisco Phones
 - a. Cisco 7961
 - b. Software Version: SCCP41.8-4-1S
6. Cisco 2432-24FXS IAD
 - a. Software Version: c2430-mz.xo
7. Cisco Catalyst 3560 PoE series P-24
 - a. Software Version: c3560-advipservicesk9-mz.122-44.SE2.bin

2. CUCM 7.0.1 Configuration

CUCM Configuration

This section includes high level configuration captures of the CUCM configuration screens relevant to configuring a SIP trunk.

XO performed the minimum amount of configuration required to achieve successful completion of test calls over XO SIP. It is beyond the scope of this document and the testing efforts to show a complete CUCM 7.0.1 configuration, therefore screenshots of the GUI interface are provided only for the details of the SIP trunk configuration that are relevant to interfacing with XO's SIP product.

Customers should refer to the CUCM administration guides for additional configuration options:

http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_maintenance_guides_list.html.

For design guidance for CUCM and Cisco Unified Communications products based on CUCM, see the Solution Reference Network Design (SRND) guides:

http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_implementation_design_guides_list.html

2.1 CUCM Region and Device Pool

The following sections contain a brief description of the CUCM region and device pool configurations used during testing.

CUCM Region and Device Pool Configuration for SP1 with CUBE

In this setup there is only one region called the default region and it is configured to use G.711 codec. The device pool used is the default device pool. All CUCM phones and the SIP trunk are assigned to the default device pool and use the G.711 codec to communicate over the default region and over the SIP trunk.

CUCM Region and Device Pool Configuration for SP2 with CUBE

In this setup there are two regions where one is the default region and the other is named the XO g729 region. The default region is configured to use the G.711 codec. The XO g729 region is configured to use the G.729 codec. All CUCM phones are configured in the default device pool and the XO SIP trunk is configured in the XO SIP Trunk Device Pool. All phones communicate with each other over the default region using G.711. When the phones establish an outbound call or receive an inbound call over the SIP trunk these calls are processed over the XO g729 region configured for G.729 codec. On the CUCM a single SIP trunk can only support one codec. This means that all calls whether they are inbound or outbound over the SIP trunk will establish using only the G.729 codec.

2.2 CUCM SIP Profile Screen Capture

This section contains the SIP Profile used during service package 1 and 2 testing.

Figure 1, SIP Profile Screen Capture Part 1

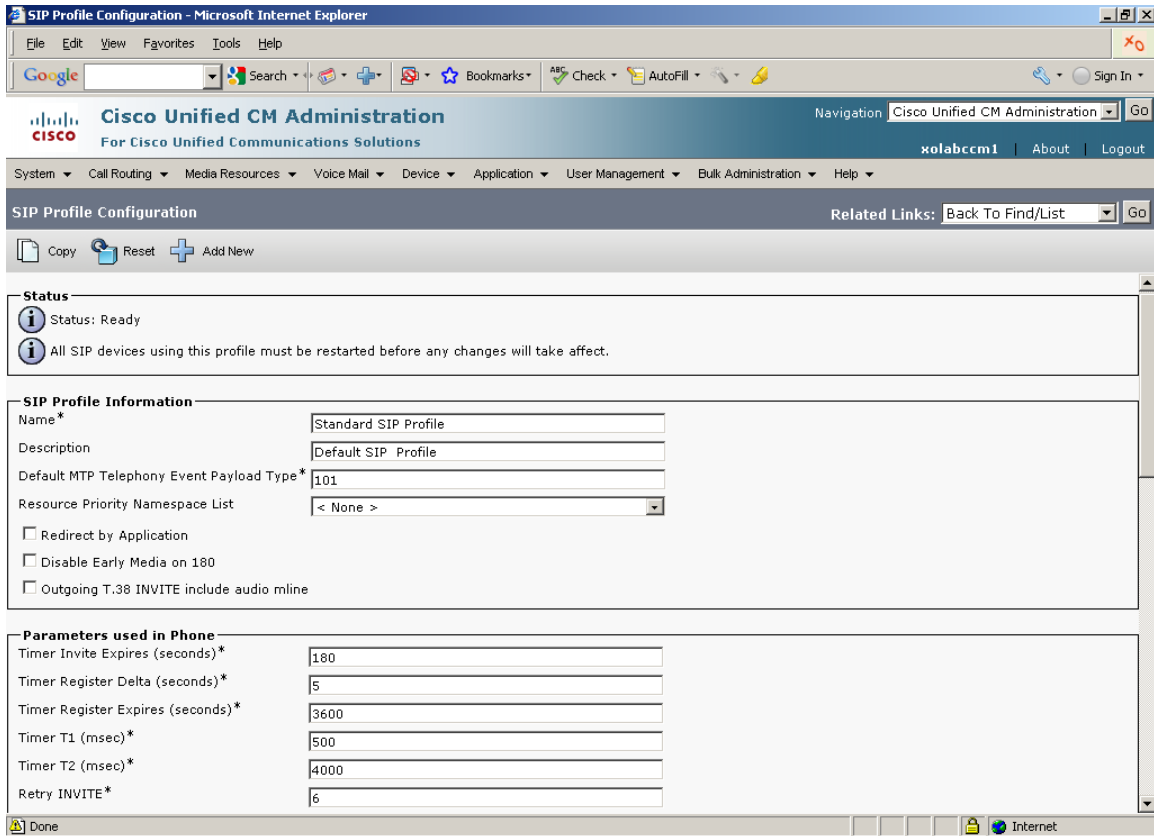


Figure 2, SIP Profile Screen Capture Part 2

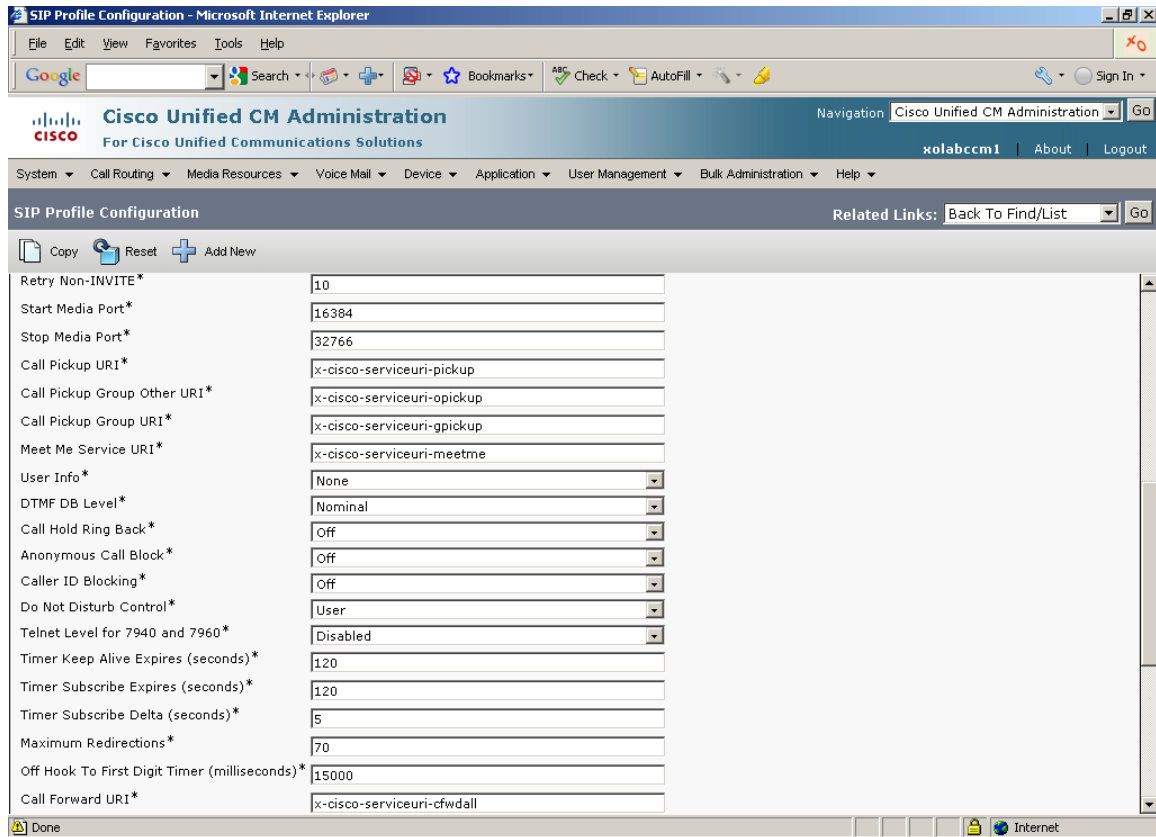
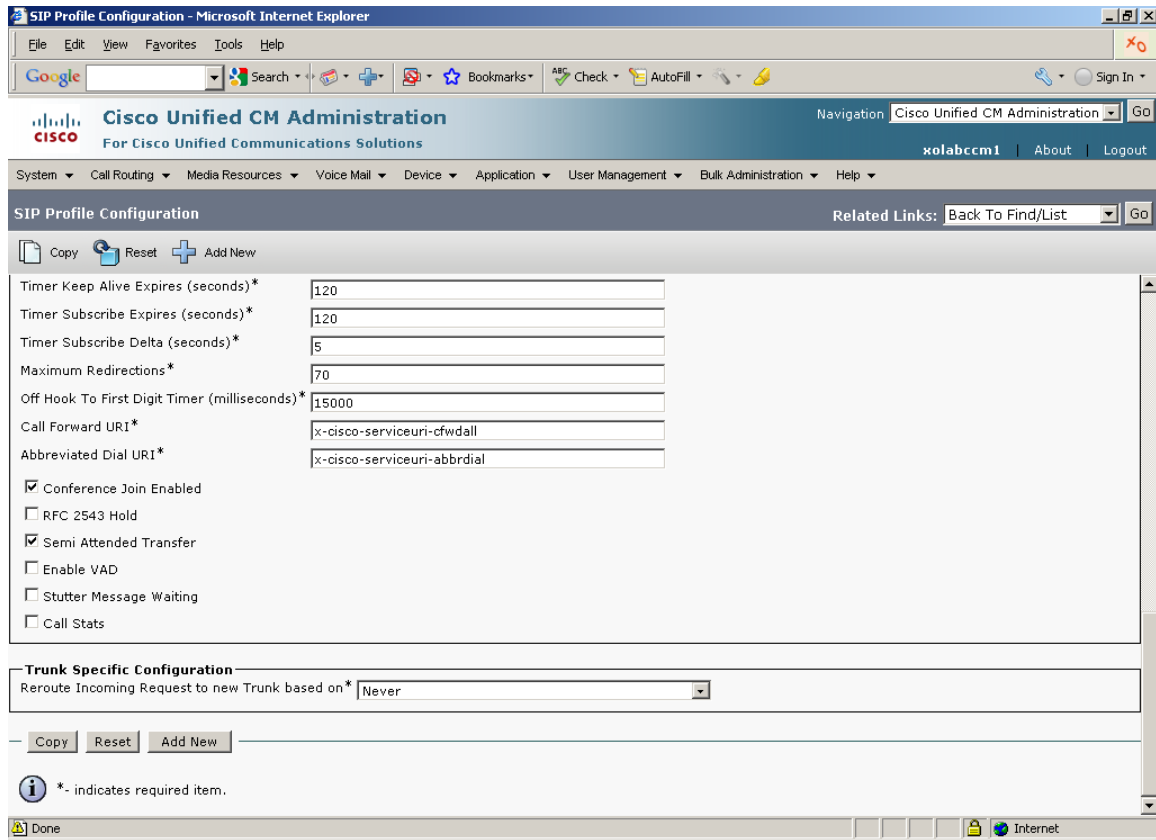
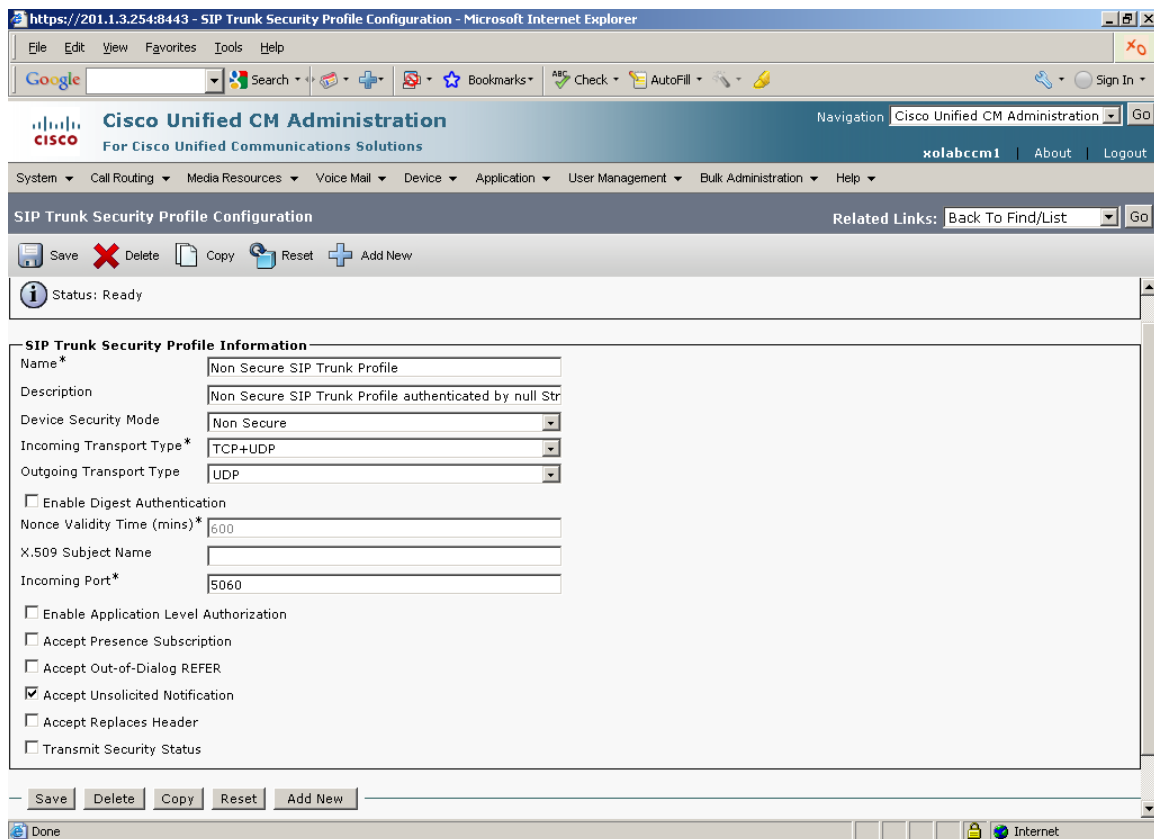


Figure 3, SIP Profile Screen Capture Part 3



2.3 CUCM SIP Trunk Security Profile

This section contains the SIP Trunk Security Profile used during SP1 and SP2 testing.



The screenshot shows the Cisco Unified CM Administration interface for configuring a SIP Trunk Security Profile. The browser address bar shows the URL: https://201.1.3.254:8443 - SIP Trunk Security Profile Configuration - Microsoft Internet Explorer.

The page title is "SIP Trunk Security Profile Configuration". The navigation menu includes: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The current page is "SIP Trunk Security Profile Configuration".

The configuration form includes the following fields and options:

- Name***: Non Secure SIP Trunk Profile
- Description**: Non Secure SIP Trunk Profile authenticated by null Str
- Device Security Mode**: Non Secure
- Incoming Transport Type***: TCP+UDP
- Outgoing Transport Type**: UDP
- Enable Digest Authentication
- Nonce Validity Time (mins)***: 600
- X.509 Subject Name**:
- Incoming Port***: 5060
- Enable Application Level Authorization
- Accept Presence Subscription
- Accept Out-of-Dialog REFER
- Accept Unsolicited Notification
- Accept Replaces Header
- Transmit Security Status

At the bottom of the form, there are buttons for Save, Delete, Copy, Reset, and Add New.

2.4 CUCM SIP Trunk Screen Captures for SP1

This section contains the SIP trunk settings used during SP1 testing. Please note that the Media Termination Point Required box is not checked. This allows CUBE to perform the Early Offer/Delayed Offer (EO/DO) conversion. The screen captures in this section use a 4 digit phone extension. Within the Call Routing Information, the Inbound Calls section has the Significant Digits* set to 4 and the option for Redirecting Diversion Header Delivery - Inbound is checked. Within the Outbound Calls section the Calling Party Selection* is set to Originator and the Caller ID DN is left blank because CUBE is configured to add the NPA-NXX via a SIP profile rule. The option for Redirecting Diversion Header Delivery - Outbound is checked. In the CUCM with CUBE SIP Trunk Screen Capture Part 3 under the SIP Information parameters the Destination Address must be set to CUBE's IP address. CUBE's sip-server address must be set to XO Communications' Sonus Networks NBS signaling IP address.

Figure 4, CUCM with CUBE SIP Trunk Screen Capture Part 1

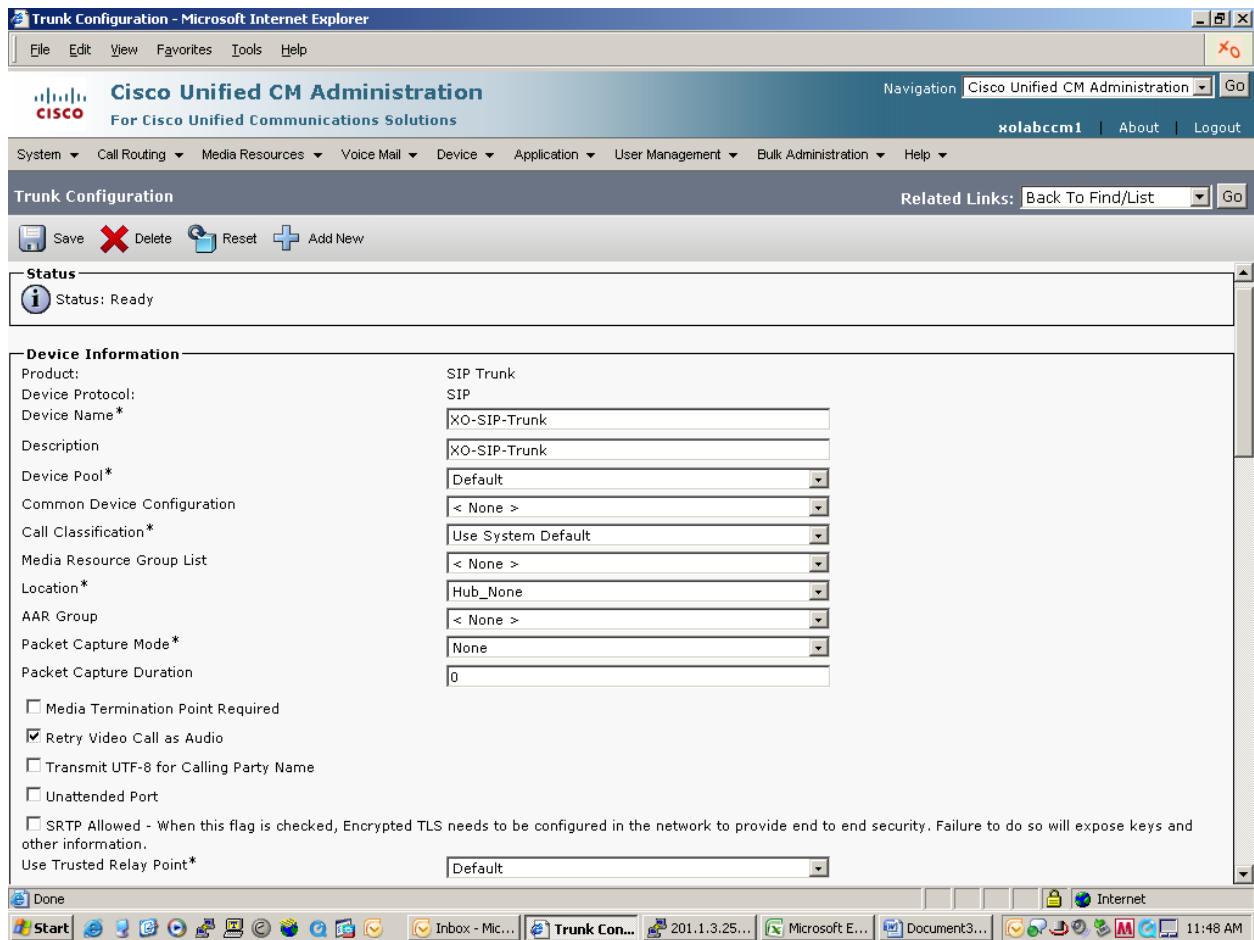


Figure 5, CUCM with CUBE SIP Trunk Screen Capture Part 2

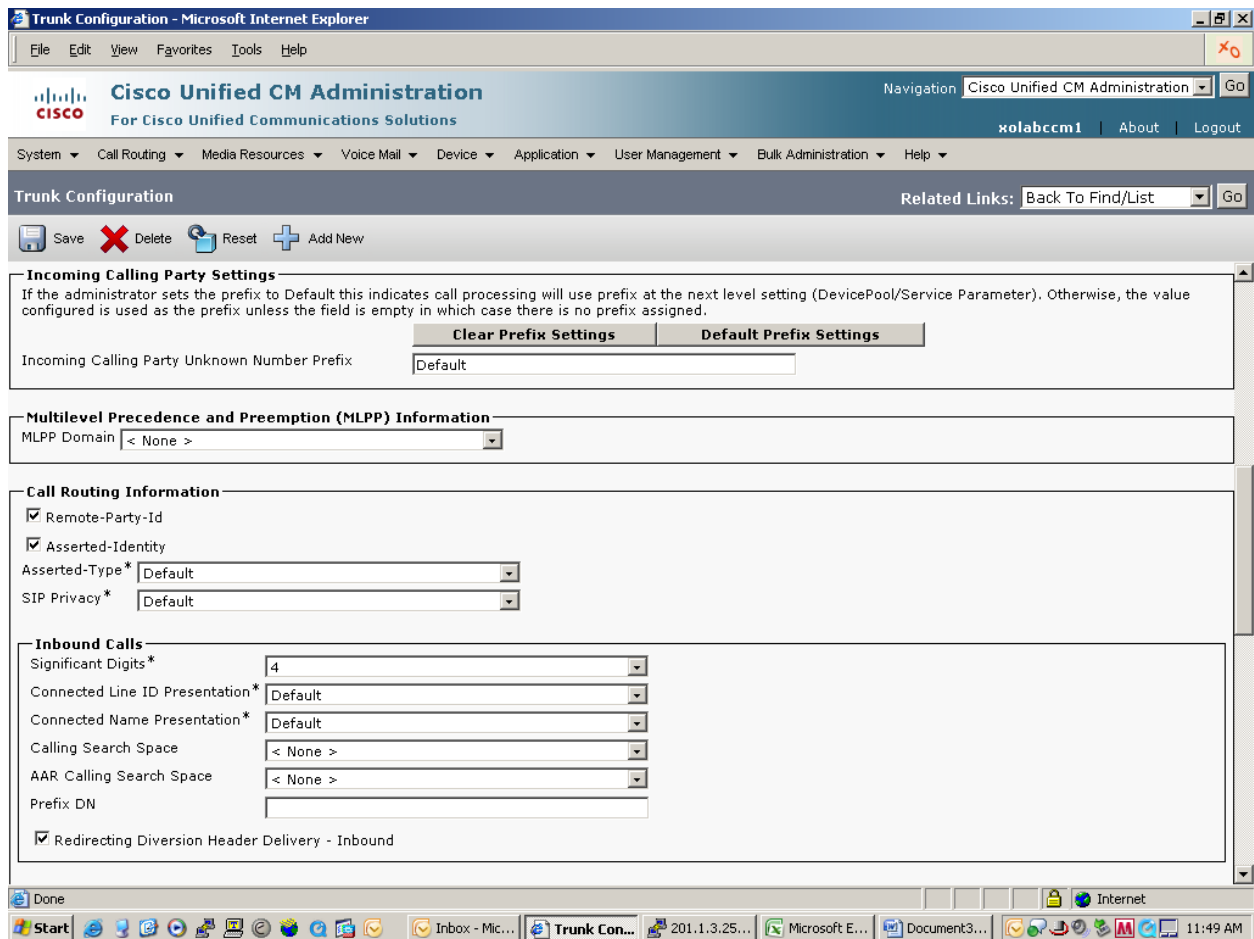


Figure 6, CUCM with CUBE SIP Trunk Screen Capture Part 3

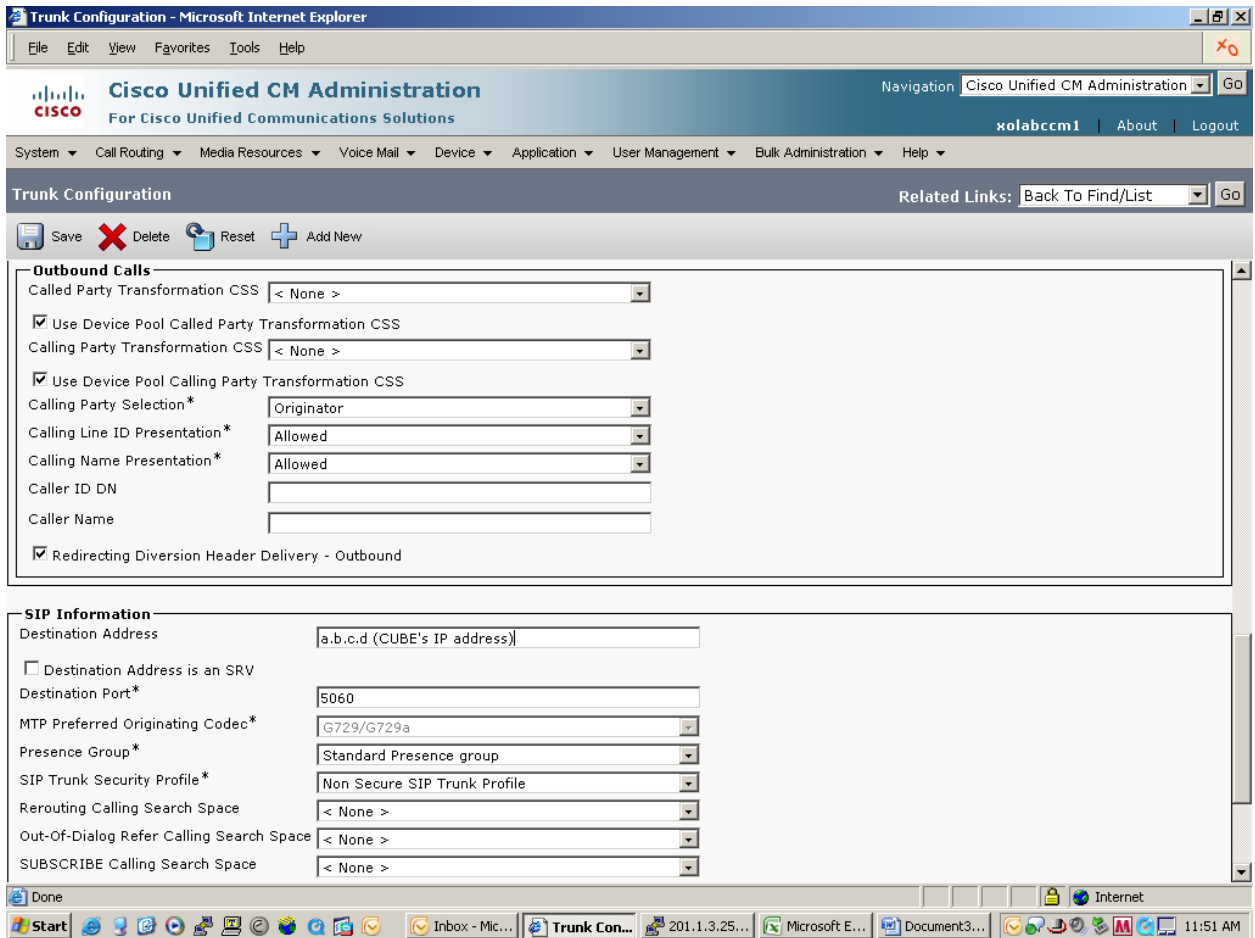
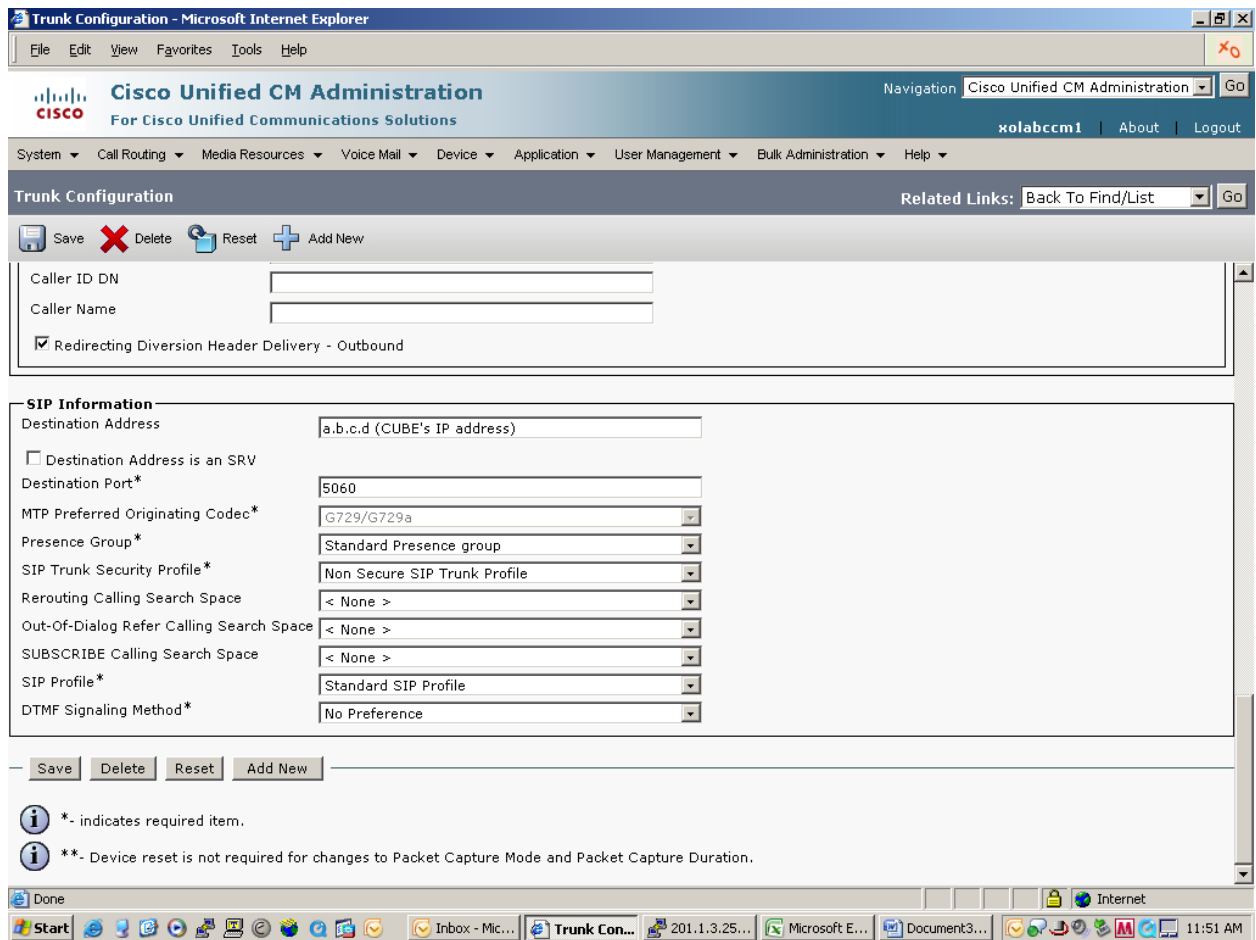


Figure 7, CUCM with CUBE SIP Trunk Screen Capture Part 4



CUCM SIP Trunk Screen Captures with CLID Blocked for SP1

The screen captures in this section show the SIP trunk configuration settings where the caller ID is blocked by using a separate route pattern. In the CUCM with CUBE CLID Blocked SIP Trunk Screen Capture Part 3 under the Outbound Calls section, the “Calling Line ID Presentation*” and the “Calling Name Presentation*” fields are set to Restricted. This SIP trunk will block the caller ID for all outbound calls. In the same screen capture under the SIP Information parameters the Destination Address must be set to CUBE’s IP address. CUBE’s sip-server address must be set to XO Communications’ Sonus Networks NBS signaling IP address.

Figure 8, CUCM with CUBE CLID Blocked SIP Trunk Screen Capture Part 1

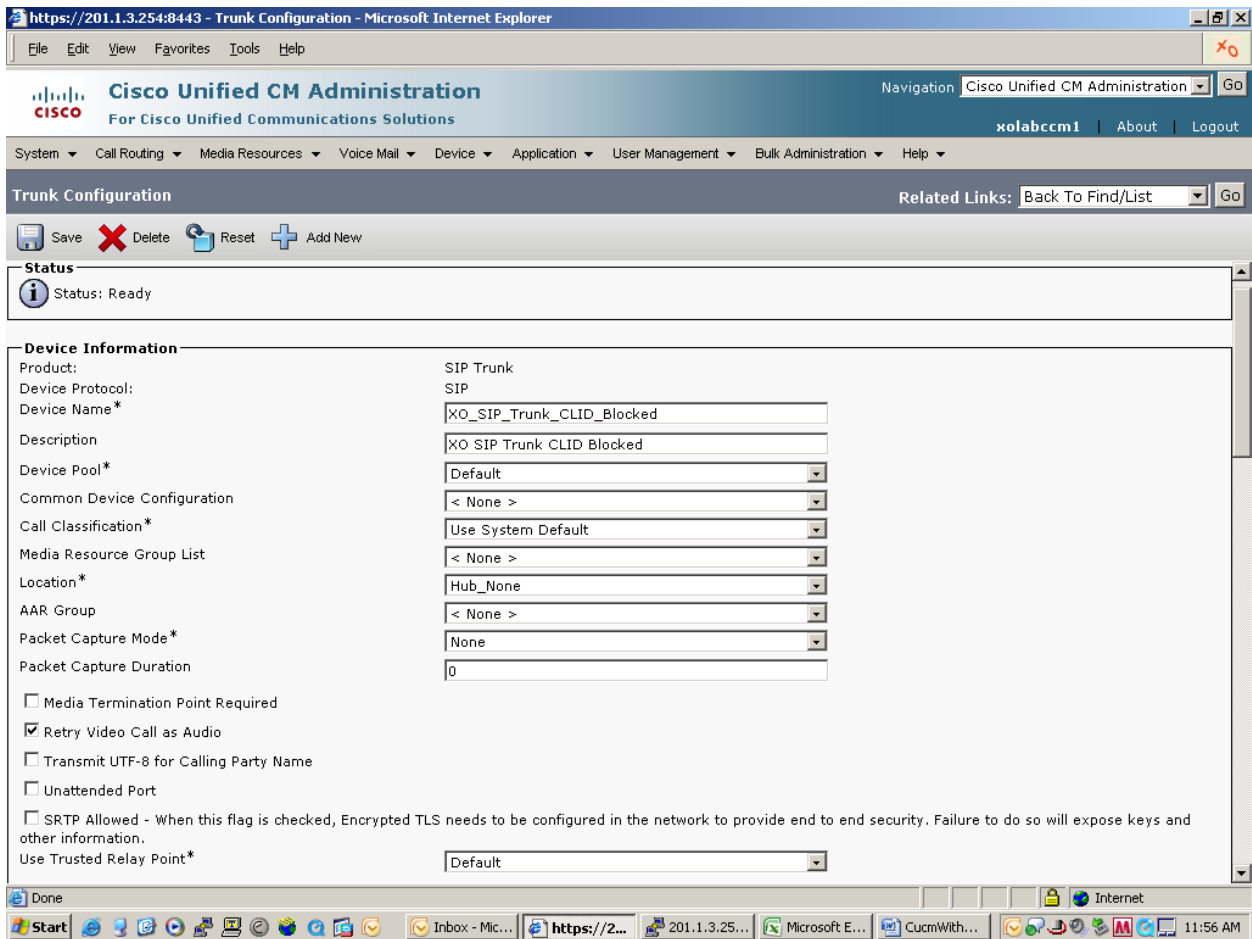


Figure 9, CUCM with CUBE CLID Blocked SIP Trunk Screen Capture Part 2

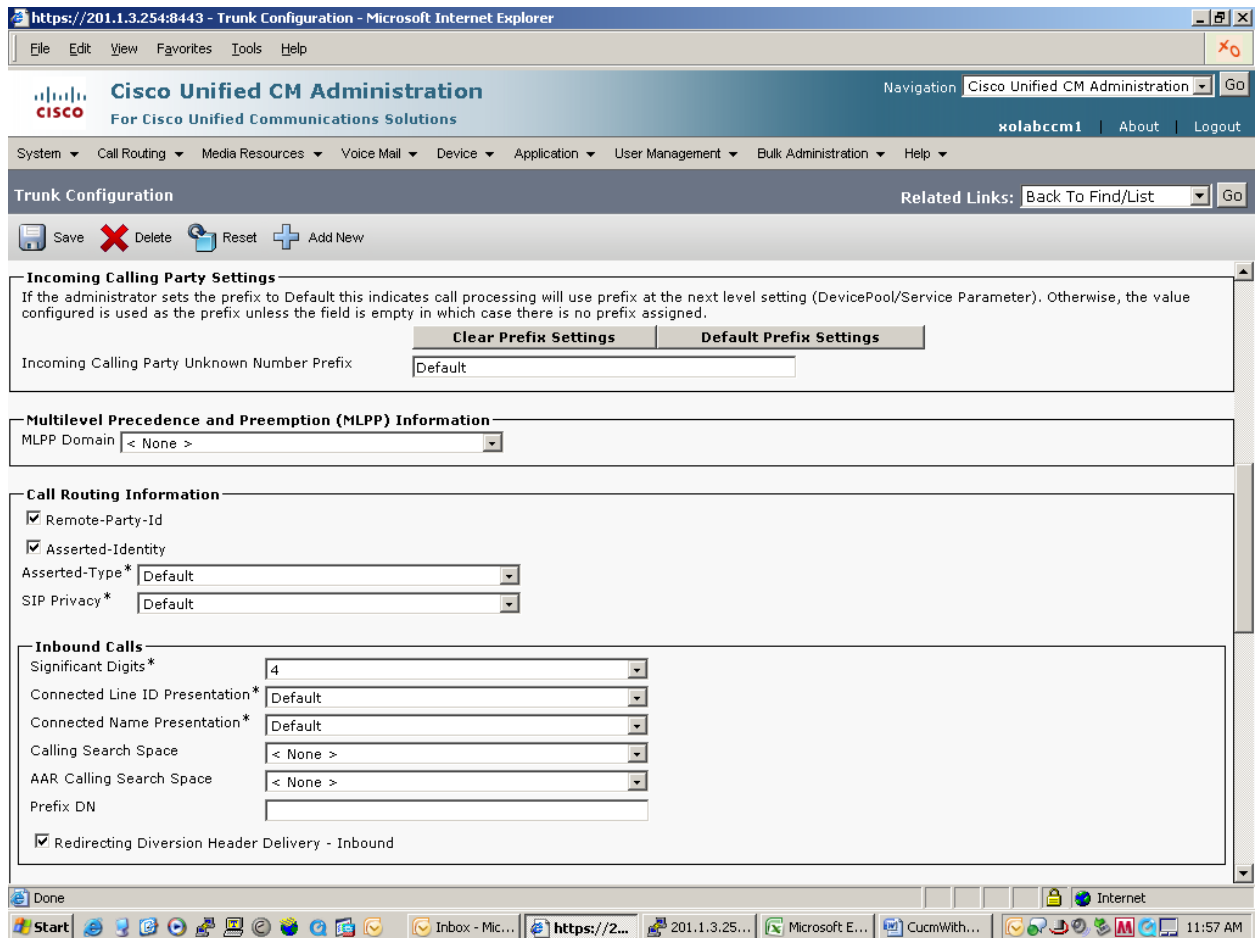


Figure 10, CUCM with CUBE CLID Blocked SIP Trunk Screen Capture Part 3

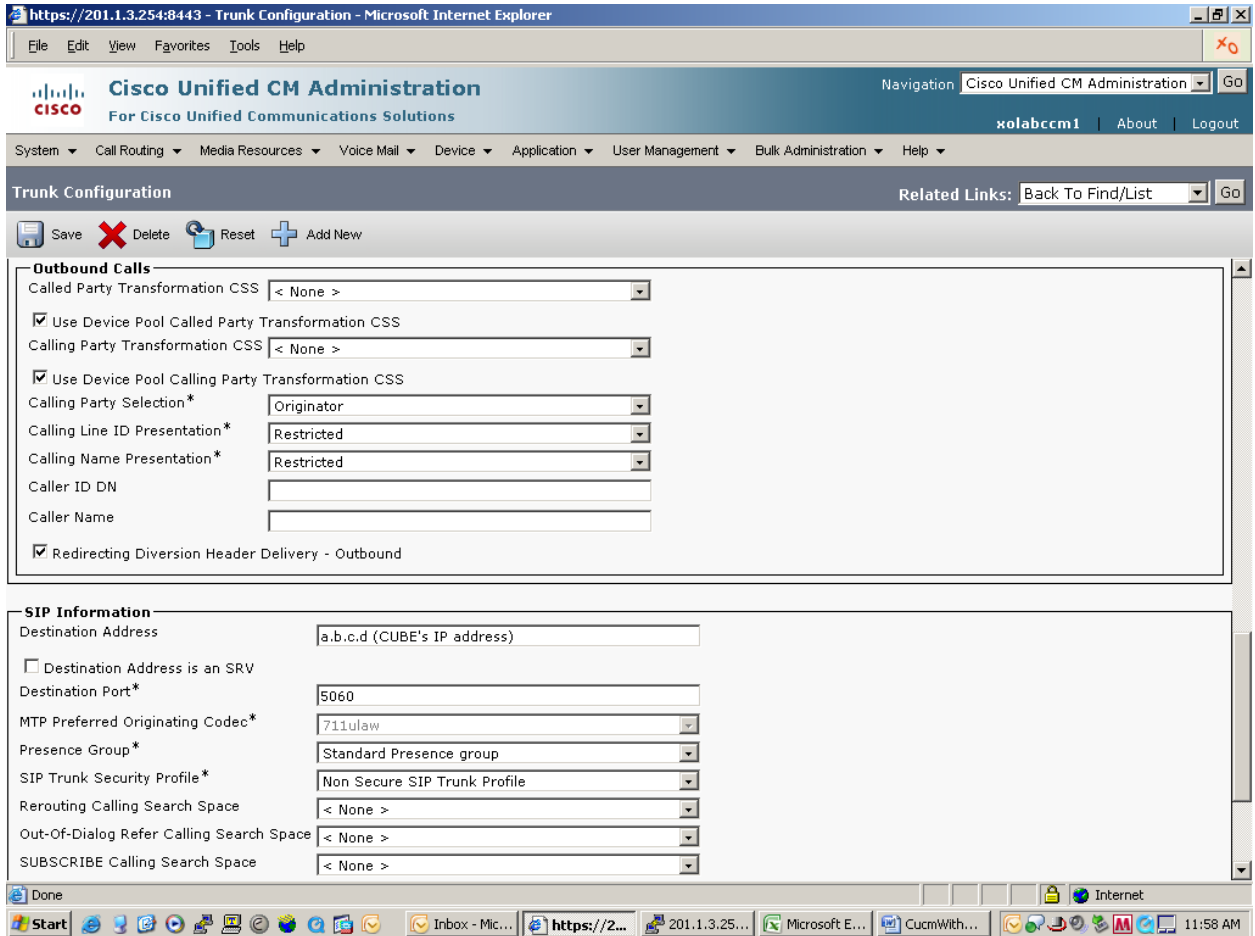
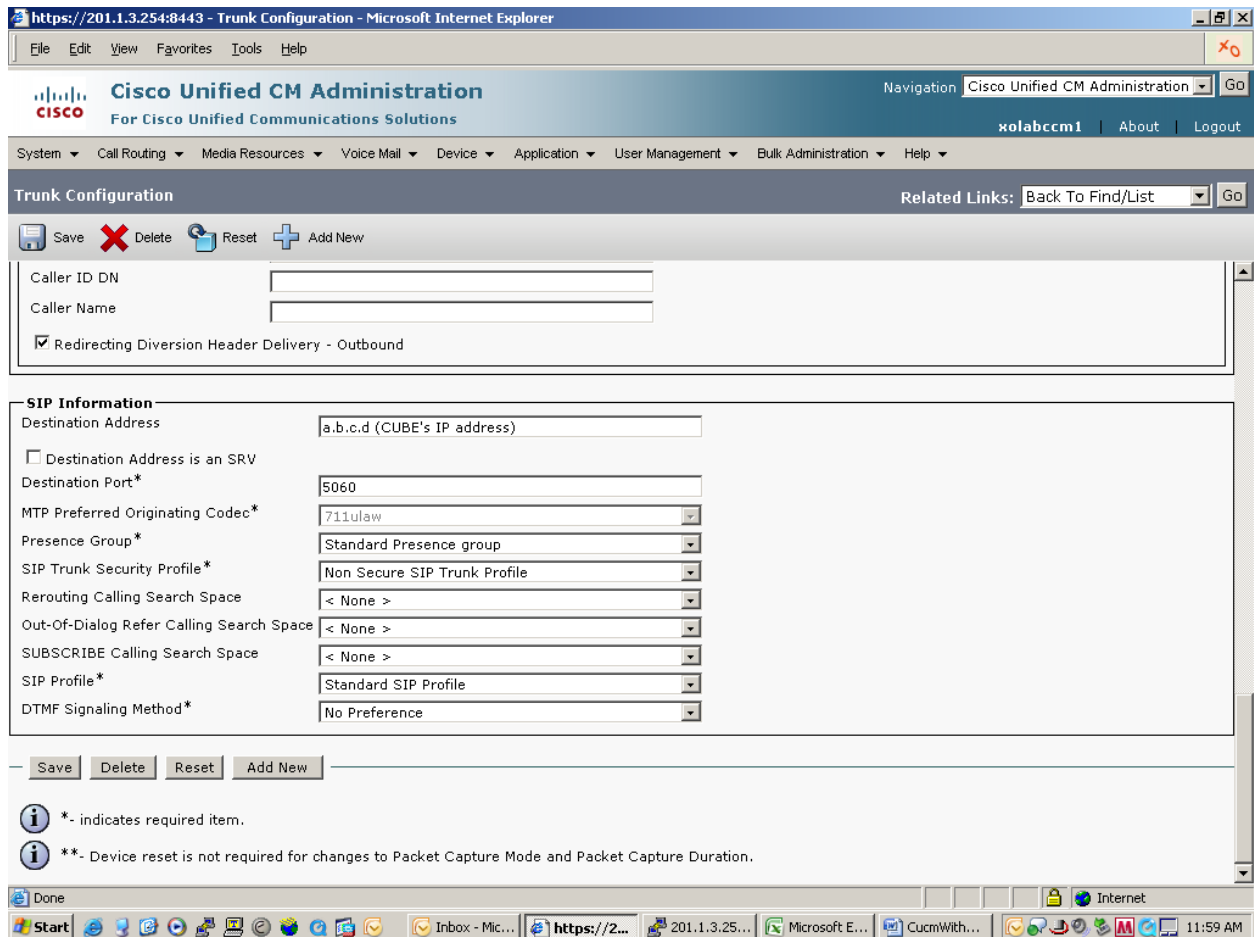


Figure 11, CUCM with CUBE CLID Blocked SIP Trunk Screen Capture Part 4



2.5 CUCM SIP Trunk Screen Captures for SP2

This section contains the SIP trunk settings used during SP2 testing. Please note that the Media Termination Point Required box is not checked. This allows CUBE to perform the Early Offer/Delayed Offer (EO/DO) conversion. The screen captures in this section use a 4 digit phone extension. Within the Call Routing Information, the Inbound Calls section has the Significant Digits* set to 4 and the option for Redirecting Diversion Header Delivery - Inbound is checked. Within the Outbound Calls section the Calling Party Selection* is set to Originator and the Caller ID DN is left blank because CUBE is configured to add the NPA-NXX via a SIP profile rule. The option for Redirecting Diversion Header Delivery - Outbound is checked. In the CUCM with CUBE SIP Trunk Screen Capture Part 3 under the SIP Information parameters the Destination Address must be set to CUBE's IP address. CUBE's sip-server address must be set to the Sonus Networks NBS signaling IP address.

Figure 12, CUCM with CUBE SIP Trunk Screen Capture Part 1

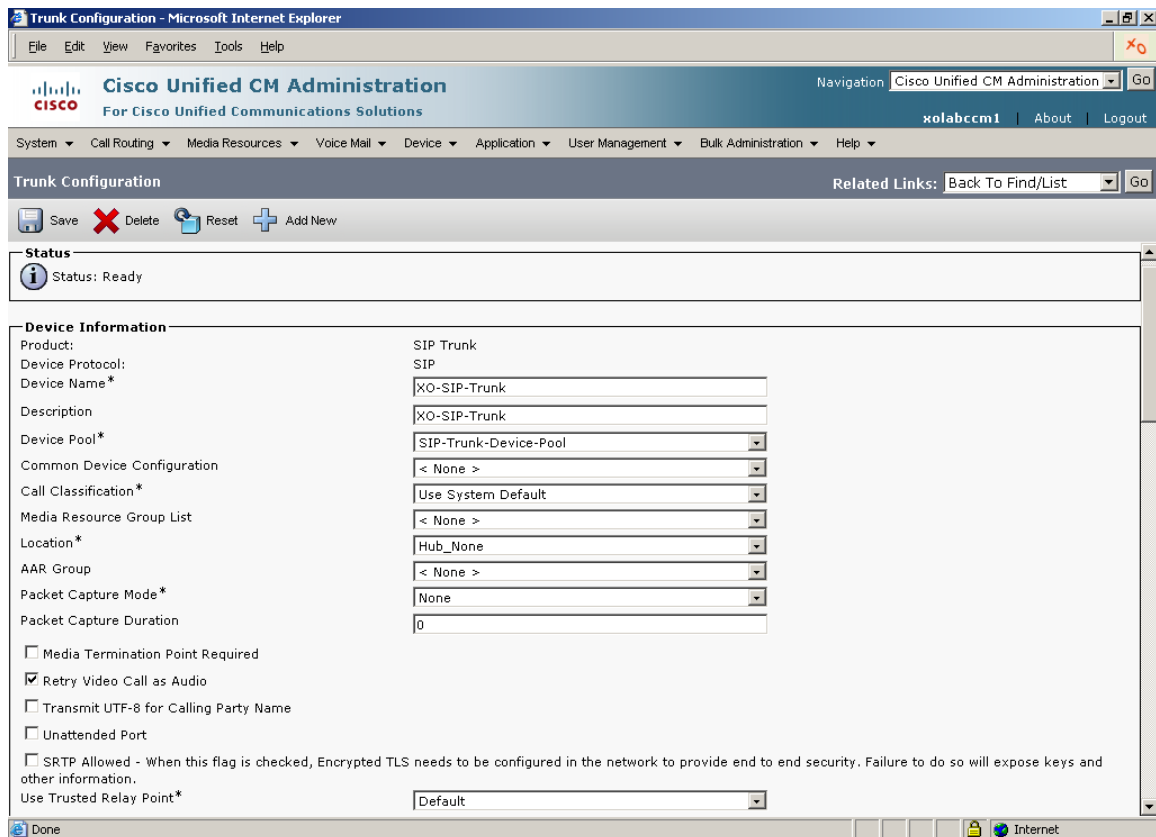


Figure 13, CUCM with CUBE SIP Trunk Screen Capture Part 2

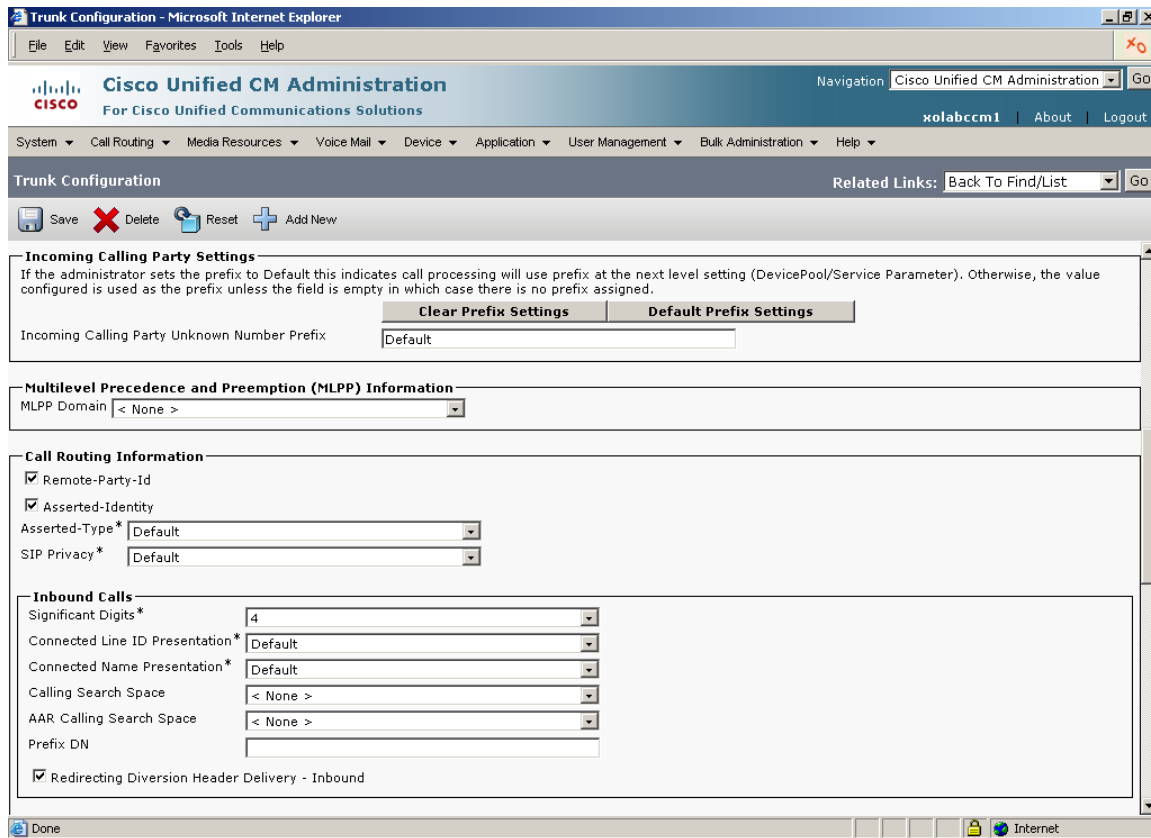


Figure 14, CUCM with CUBE SIP Trunk Screen Capture Part 3

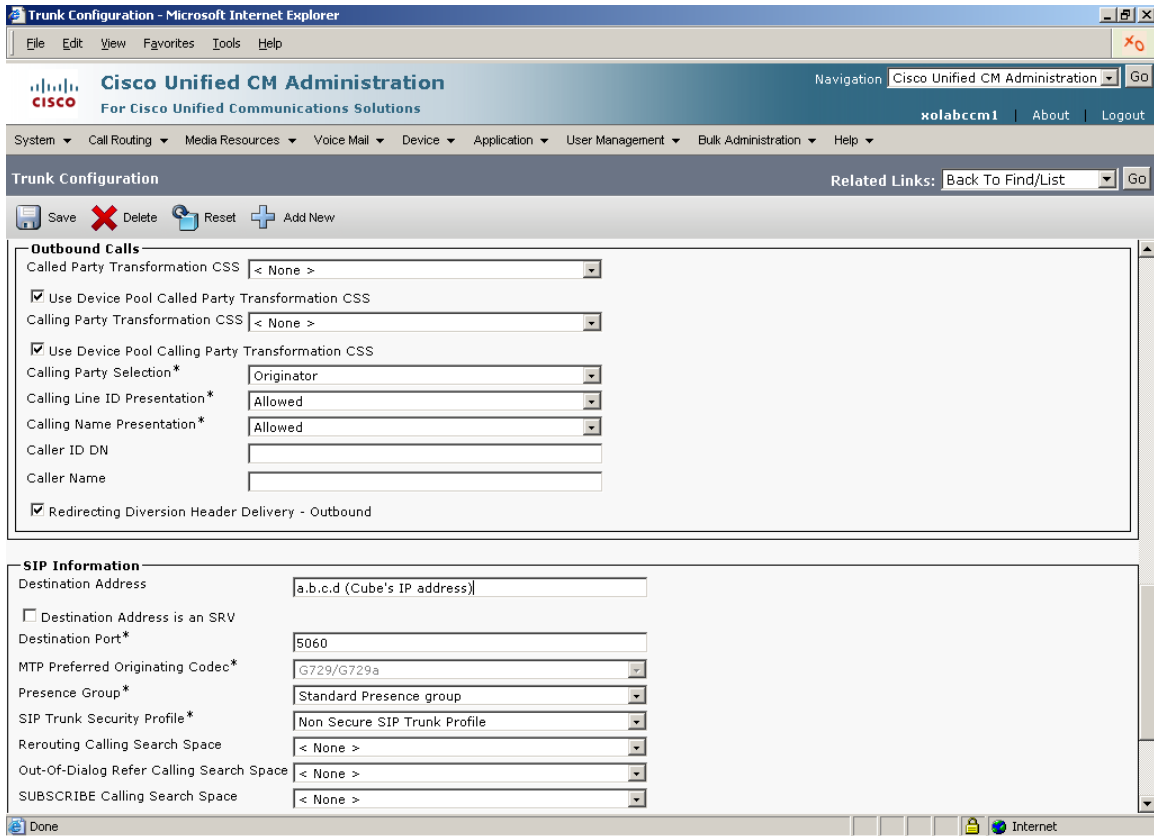
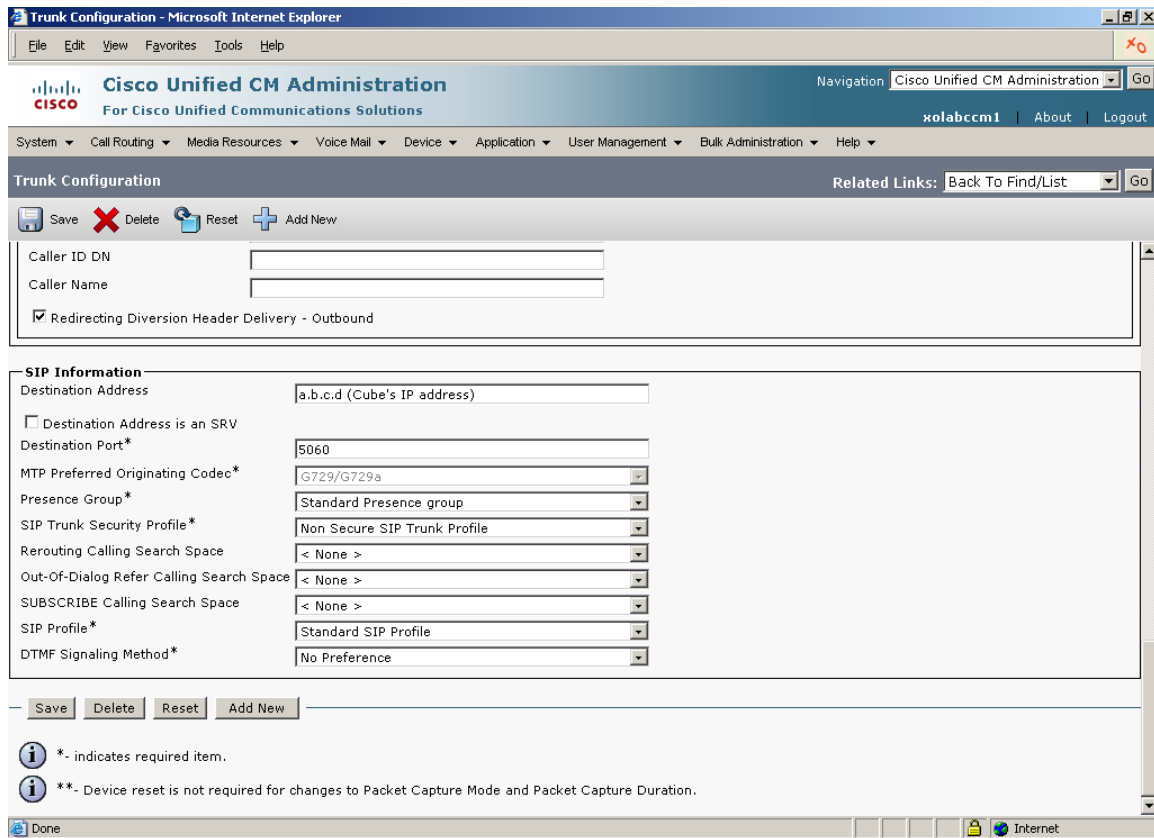


Figure 15, CUCM with CUBE SIP Trunk Screen Capture Part 4



CUCM SIP Trunk Screen Captures with CLID Blocked for SP2

The screen captures in this section show the SIP trunk configuration settings where the caller ID is blocked by using a separate route pattern. This SIP trunk will block the caller ID for all outbound calls. In the CUCM with CUBE CLID Blocked SIP Trunk Screen Capture Part 3 under the Outbound Calls section, the Calling Line ID Presentation* and the Calling Name Presentation* fields are set to Restricted. In the same screen capture under the SIP Information parameters the Destination Address must be set to CUBE's IP address. CUBE's sip-server address must be set to the Sonus Networks NBS signaling IP address.

Figure 16, CUCM with CUBE CLID Blocked SIP Trunk Screen Capture Part 1

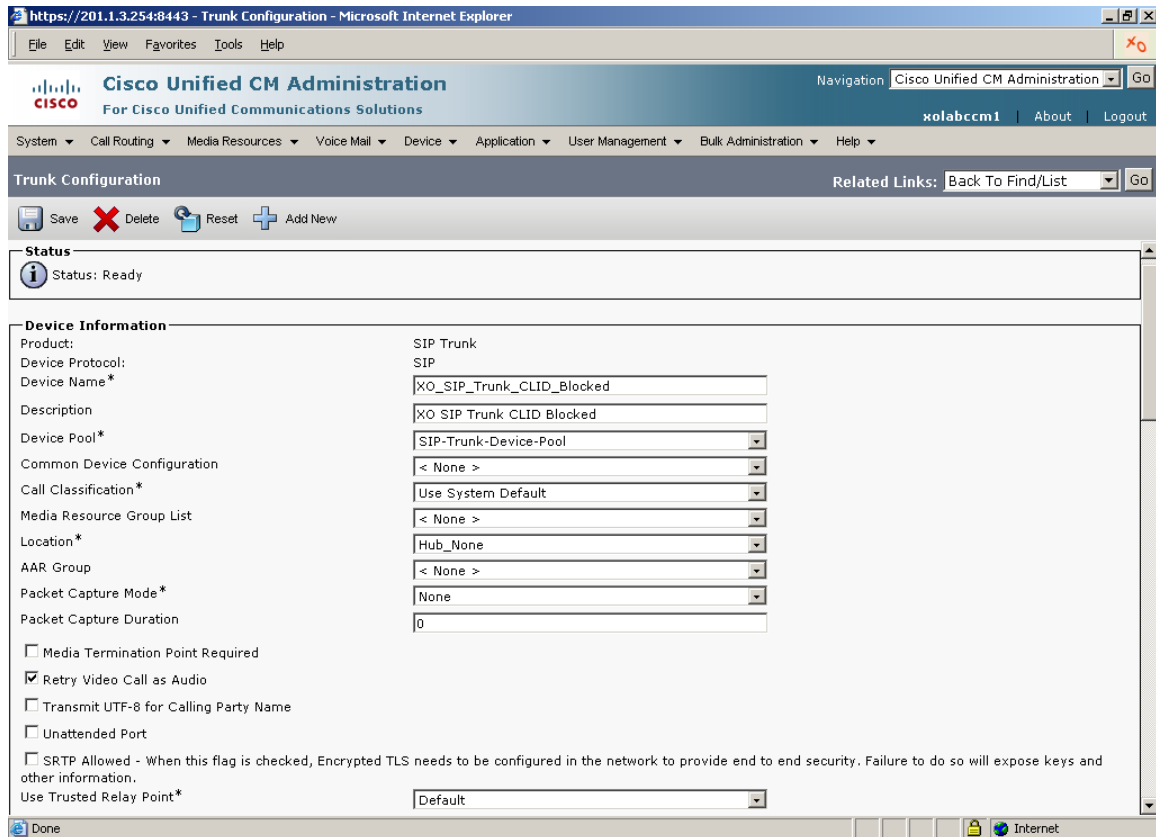


Figure 17, CUCM with CUBE CLID Blocked SIP Trunk Screen Capture Part 2

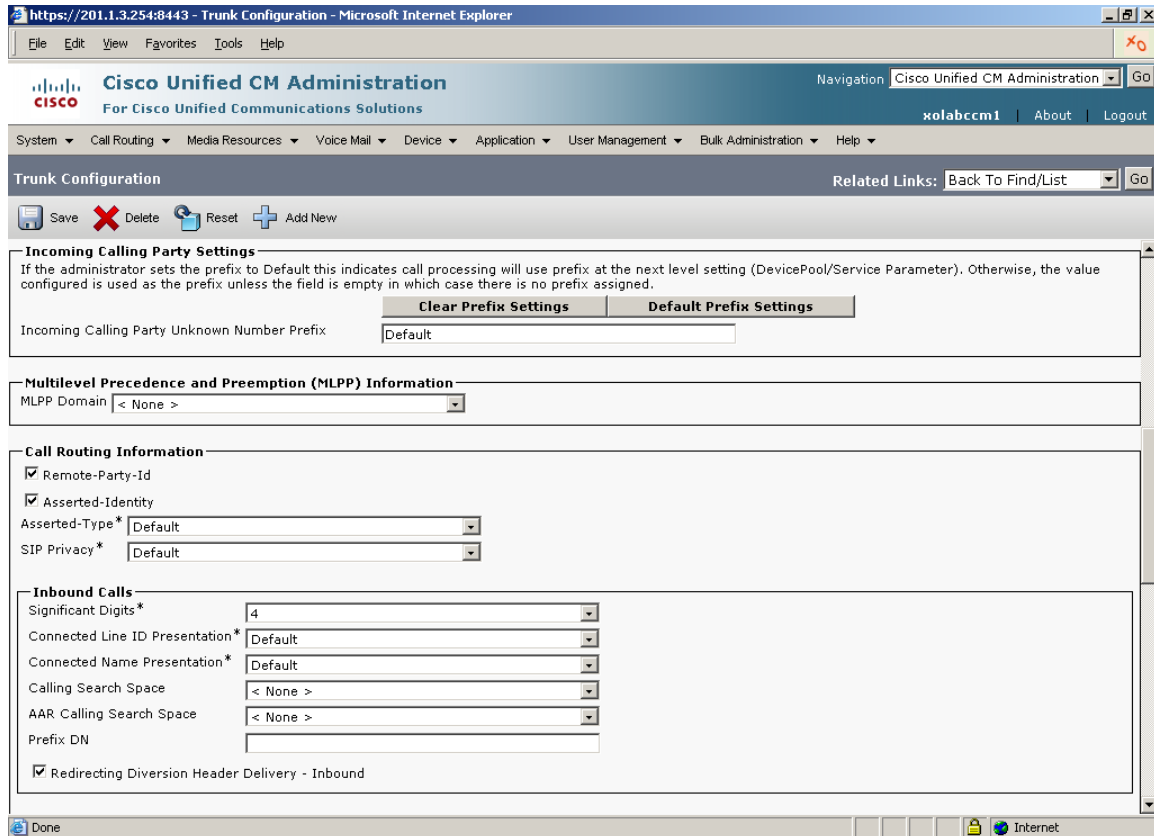


Figure 18, CUCM with CUBE CLID Blocked SIP Trunk Screen Capture Part 3

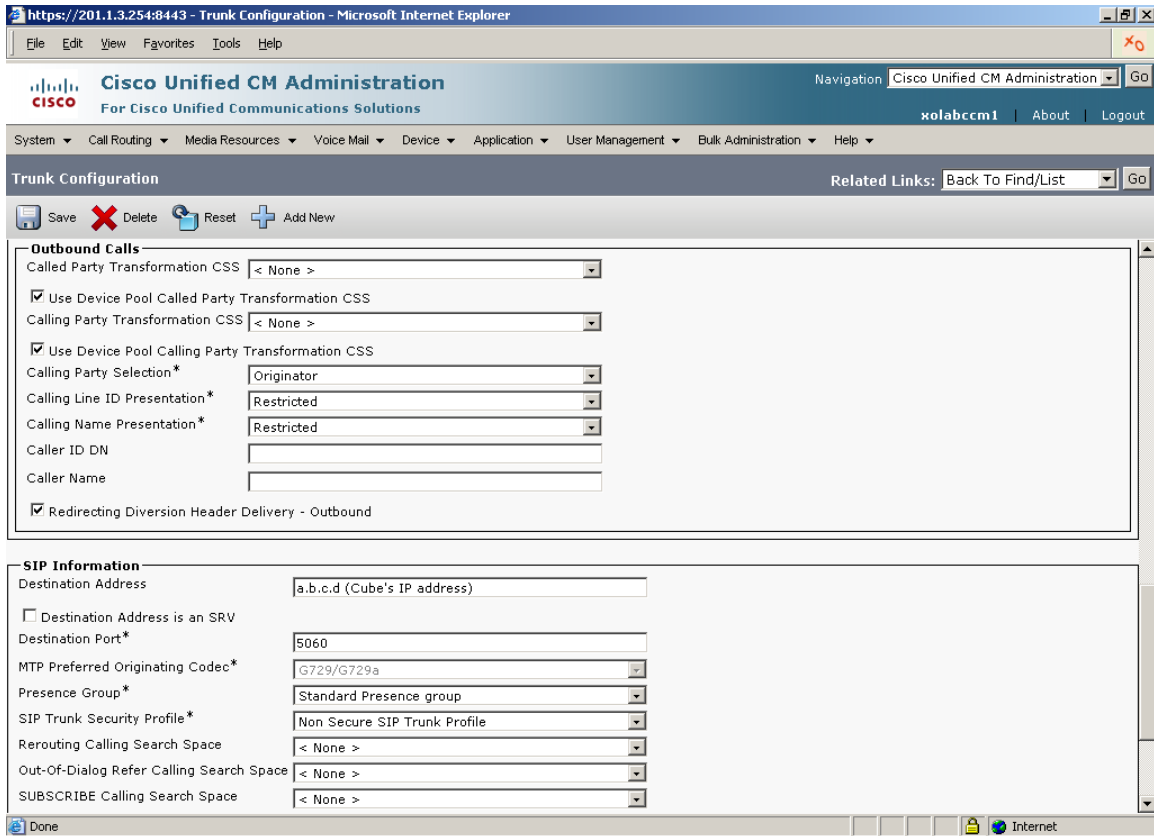
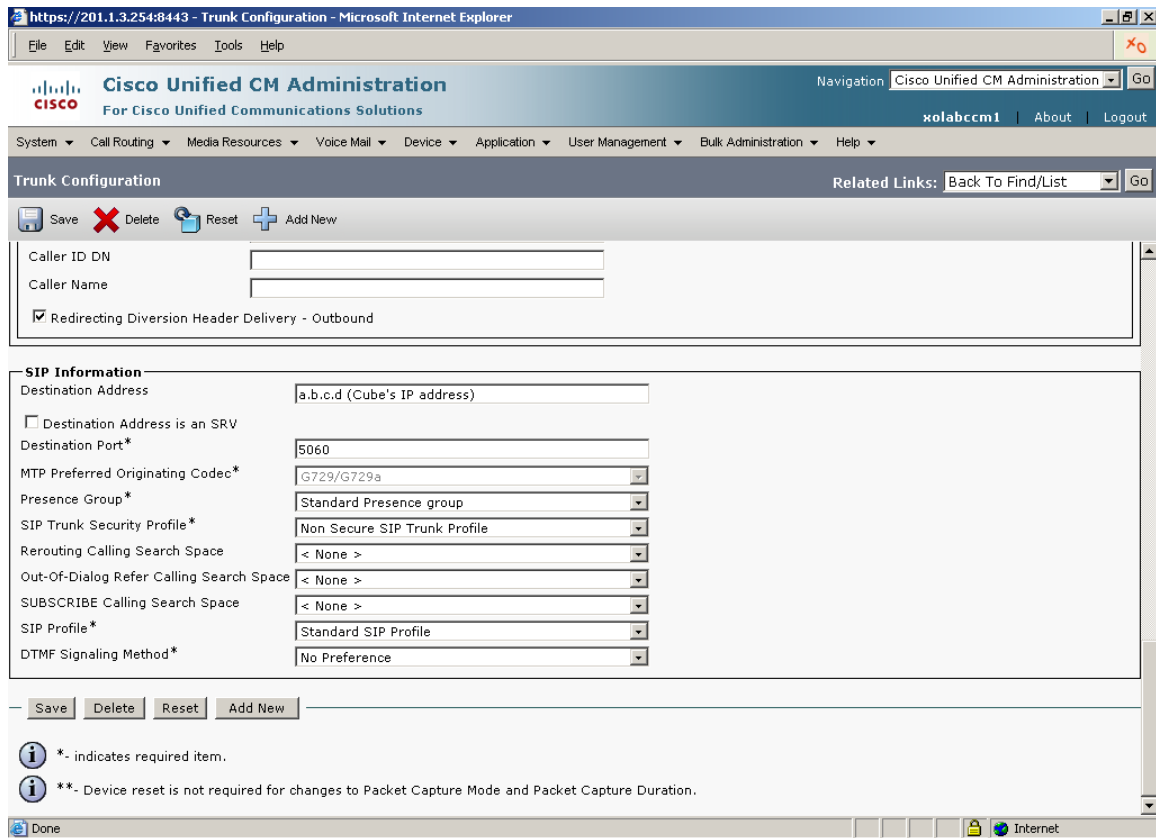
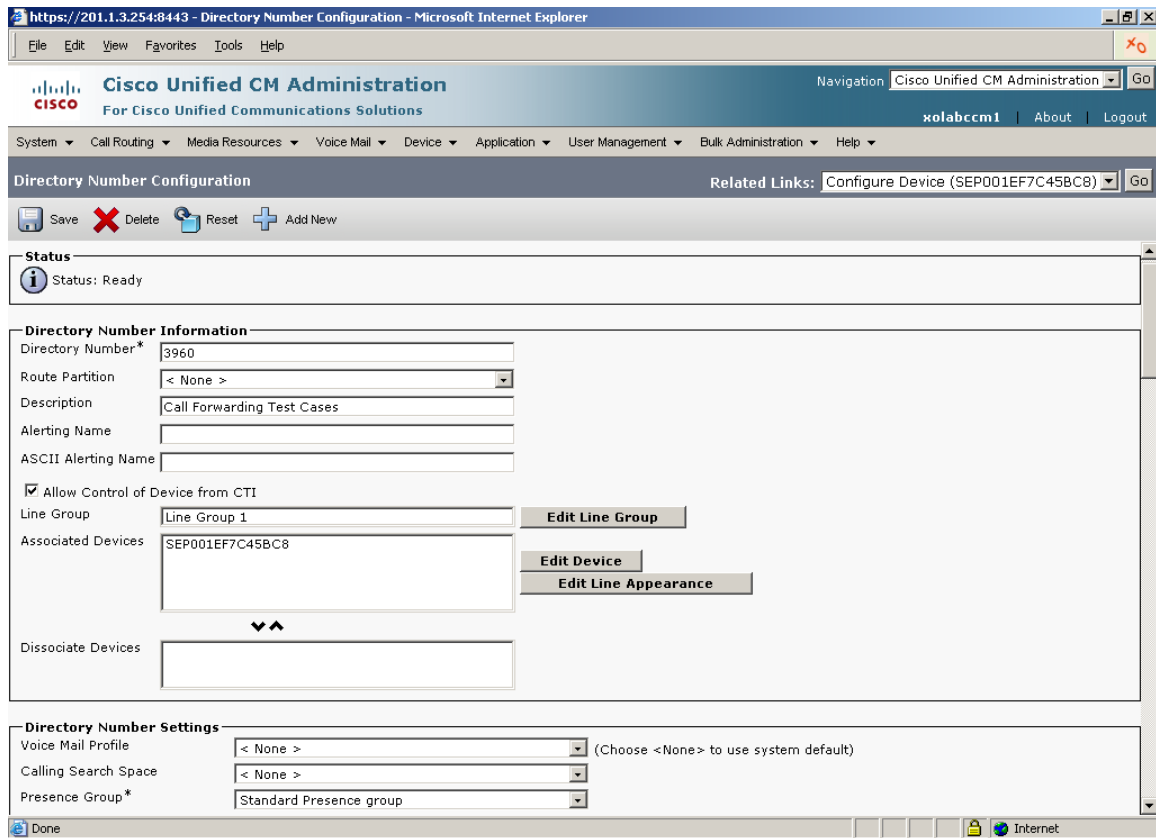


Figure 19, CUCM with CUBE CLID Blocked SIP Trunk Screen Capture Part 4



2.6 CUCM Phone Configuration Using a Four Digit Extension

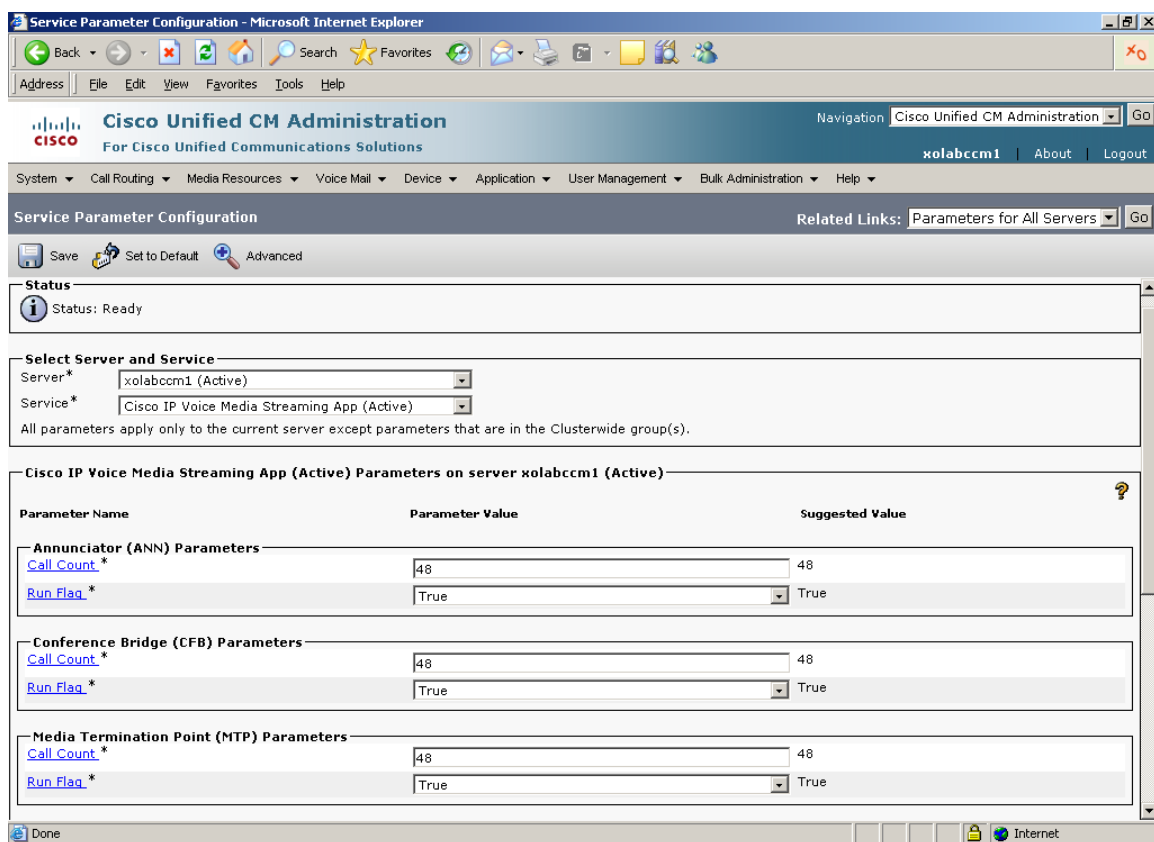
Figure 20, CUCM Phone Screen Configuration Using a Four Digit Extension



2.7 CUCM Music on Hold Server Codec Selection

When configuring the customer for SP1 or SP2, the Music on Hold Server Codec setting under CUCM administration, Service Parameters, Cisco IP Voice Media Streaming Application, Clusterwide Parameters, the codec selection displayed under Supported Music on Hold Codecs must be set to G.711ulaw for SP1 and G.729 for SP2 accordingly. G.711ulaw is the system default. The codec must be selected and saved as shown in the screen captures below. The highlighted codec indicates the codec that is currently in use.

Figure 21, Music on Hold Server Codec Screen Capture Part 1



The screenshot shows the Cisco Unified CM Administration interface in Microsoft Internet Explorer. The page title is "Service Parameter Configuration - Microsoft Internet Explorer". The navigation menu includes System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The current page is "Service Parameter Configuration" for the server "xolabccm1".

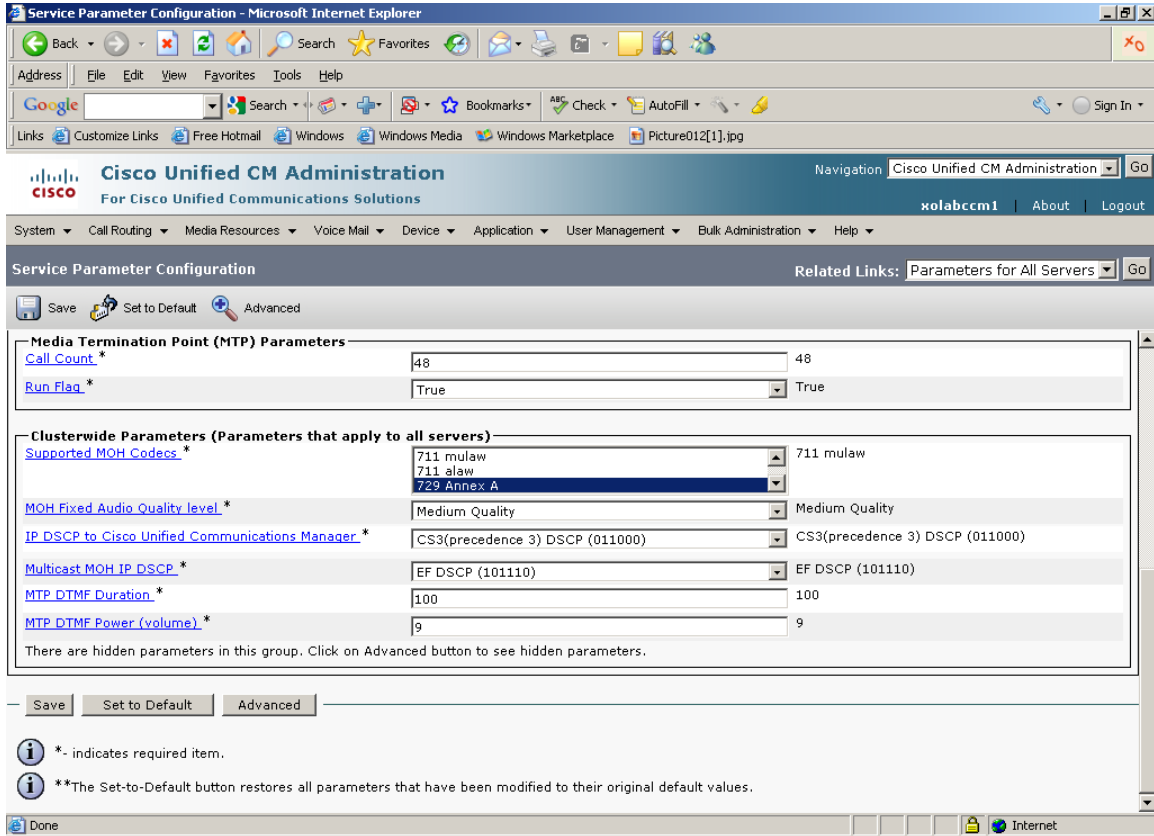
The "Status" section shows "Status: Ready".

The "Select Server and Service" section shows the "Server" dropdown set to "xolabccm1 (Active)" and the "Service" dropdown set to "Cisco IP Voice Media Streaming App (Active)". A note below states: "All parameters apply only to the current server except parameters that are in the Clusterwide group(s)."

The "Cisco IP Voice Media Streaming App (Active) Parameters on server xolabccm1 (Active)" section contains a table of parameters:

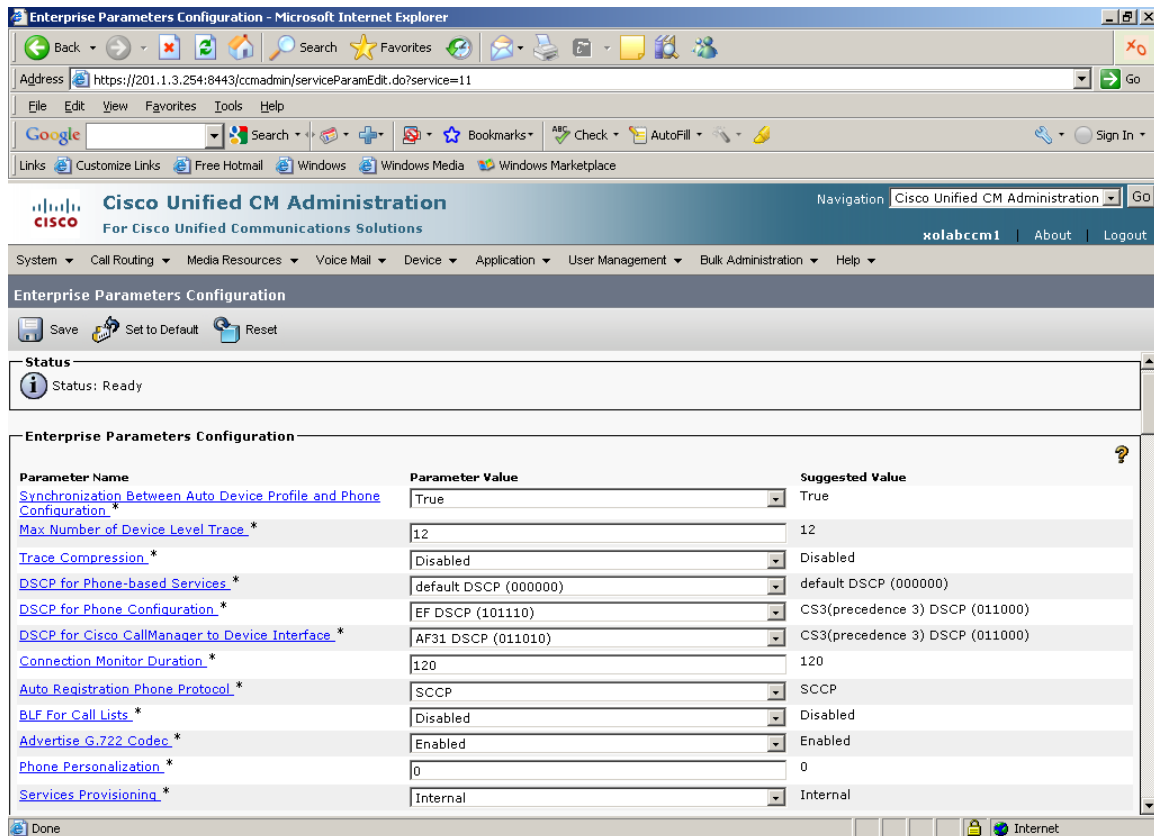
Parameter Name	Parameter Value	Suggested Value
Annunciator (ANN) Parameters		
Call Count *	48	48
Run Flag *	True	True
Conference Bridge (CFB) Parameters		
Call Count *	48	48
Run Flag *	True	True
Media Termination Point (MTP) Parameters		
Call Count *	48	48
Run Flag *	True	True

Figure 22, Music on Hold Server Codec Screen Capture Part 2



2.8 CUCM Enterprise Parameters: DSCP Bit Settings for Signaling and Media

The screen capture below shows the DSCP bit settings for the CUCM phones for signaling and media. The DSCP for CUCM to Device Interface* parameter is the signaling setting which is AF31 and the DSCP for Phone Configuration* parameter is the media setting which is EF. The CUCM server must be rebooted for these changes to take effect.

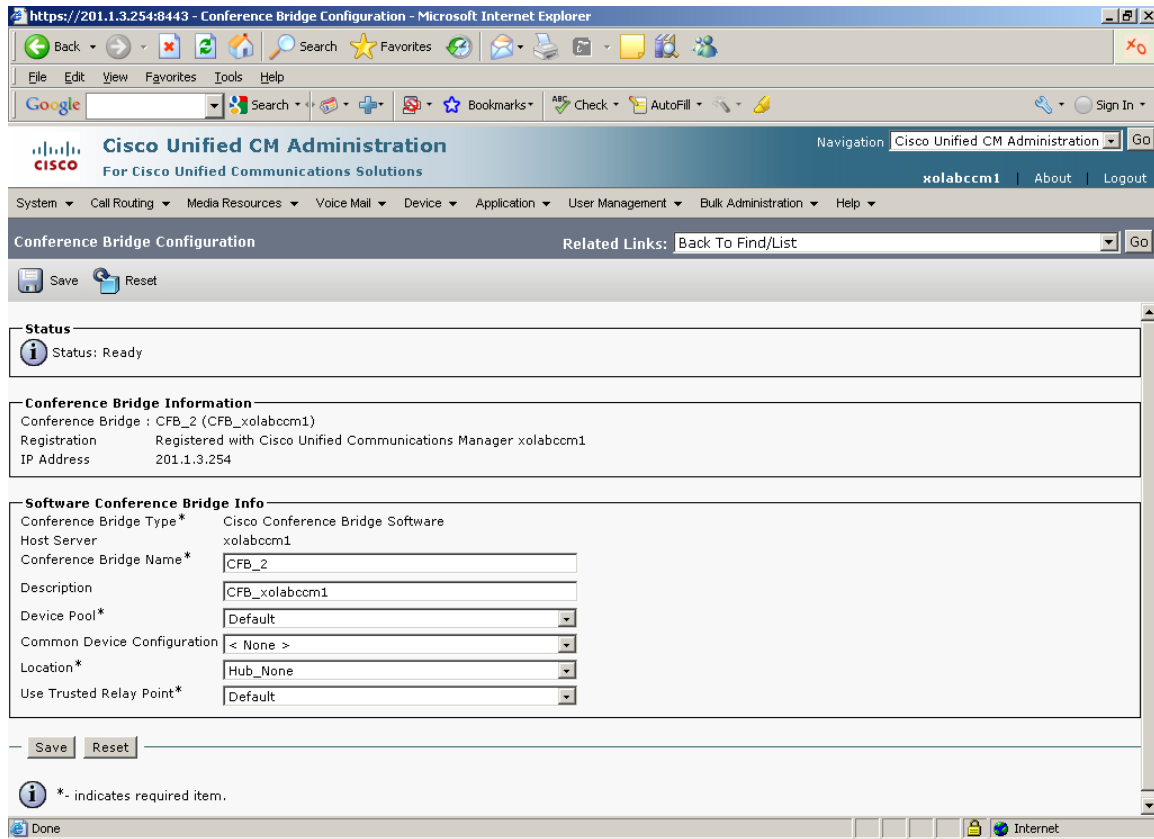


The screenshot displays the Cisco Unified CM Administration web interface. The main content area is titled "Enterprise Parameters Configuration" and shows a table of parameters. The status is "Ready".

Parameter Name	Parameter Value	Suggested Value
Synchronization Between Auto Device Profile and Phone Configuration *	True	True
Max Number of Device Level Trace *	12	12
Trace Compression *	Disabled	Disabled
DSCP for Phone-based Services *	default DSCP (000000)	default DSCP (000000)
DSCP for Phone Configuration *	EF DSCP (101110)	CS3(precedence 3) DSCP (011000)
DSCP for Cisco CallManager to Device Interface *	AF31 DSCP (011010)	CS3(precedence 3) DSCP (011000)
Connection Monitor Duration *	120	120
Auto Registration Phone Protocol *	SCCP	SCCP
BLF For Call Lists *	Disabled	Disabled
Advertise G.722 Codec *	Enabled	Enabled
Phone Personalization *	0	0
Services Provisioning *	Internal	Internal

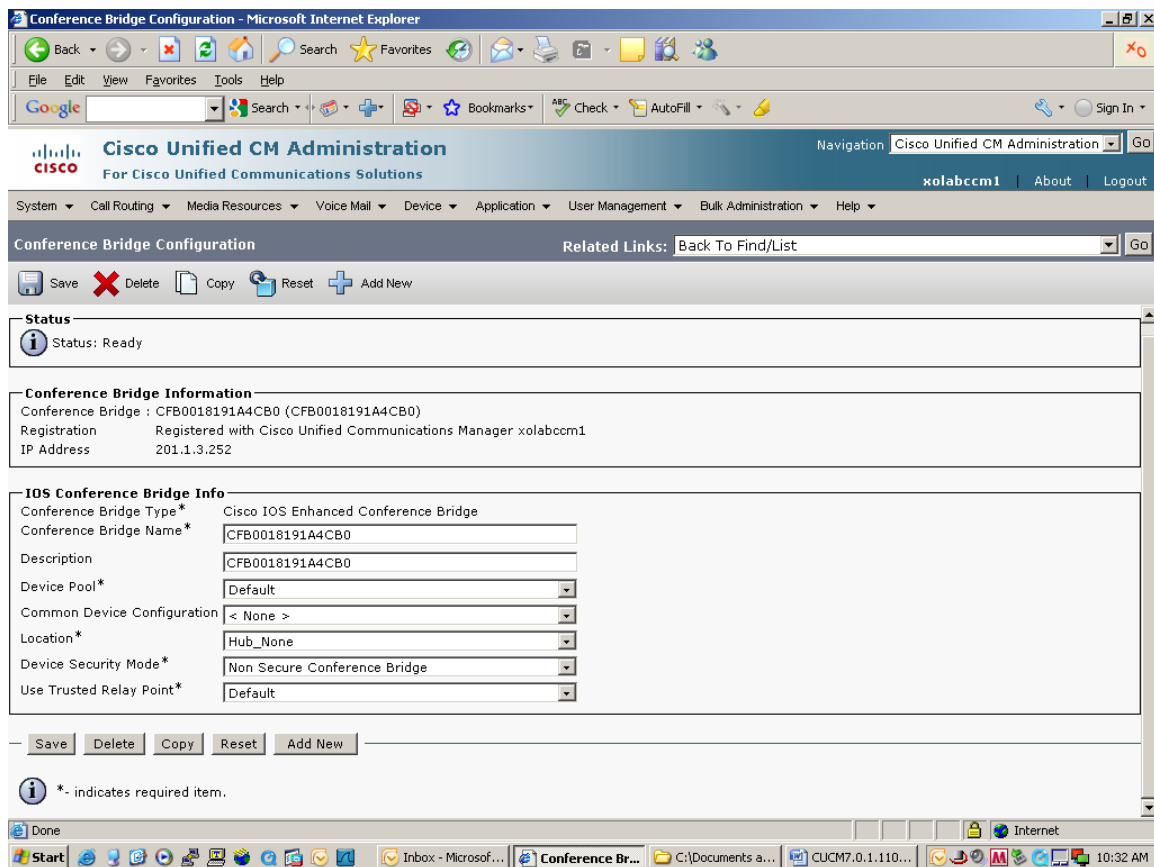
2.9 CUCM Conference Bridge Configuration for SP1

The screen capture below shows the CUCM conference bridge resource parameters listed under CUCM Administration, Media Resources for the default CUCM conference bridge. The CUCM software conference bridge was used to verify SP1 three and four way conference bridging test cases. This is a software resource that runs on the CUCM server itself versus the external conference bridge resource for SP2 which must run on CUBE due to the G.729 codec requirement discussed in the next section. This screen can also be used to verify that the CUCM conference bridge resource is registered by checking the registration state.



2.10 CUCM Conference Bridge Configuration for SP2

The screen capture below shows the Conference Bridge resource parameters listed under CUCM Administration, Media Resources. An external conference bridge resource is configured using CUBE which registers with the CUCM. This conference bridge resource was used in testing SP2 conference bridging for three and four way conference calls. When the CUCM is configured for SP1 which uses the G.711 codec, the CUCM uses its own software conference bridge. However when the CUCM is configured for SP2 which uses the G.729 codec, the conference bridge must be supported on an external device such as CUBE. This screen can also be used to verify that the CUBE conference bridge resource is registered with the CUCM by checking the registration state.



The screenshot displays the Cisco Unified CM Administration web interface in Microsoft Internet Explorer. The page title is "Conference Bridge Configuration - Microsoft Internet Explorer". The browser's address bar shows the URL "http://xolabccm1:8443/cucm/". The page content includes a navigation menu with options like "System", "Call Routing", "Media Resources", "Voice Mail", "Device", "Application", "User Management", "Bulk Administration", and "Help". The main content area is titled "Conference Bridge Configuration" and shows the configuration details for a specific resource.

Conference Bridge Configuration

Related Links: [Back To Find/List](#)

Save Delete Copy Reset Add New

Status

Status: Ready

Conference Bridge Information

Conference Bridge : CFB0018191A4CB0 (CFB0018191A4CB0)
 Registration Registered with Cisco Unified Communications Manager xolabccm1
 IP Address 201.1.3.252

IOS Conference Bridge Info

Conference Bridge Type*	Cisco IOS Enhanced Conference Bridge
Conference Bridge Name*	CFB0018191A4CB0
Description	CFB0018191A4CB0
Device Pool*	Default
Common Device Configuration	< None >
Location*	Hub_None
Device Security Mode*	Non Secure Conference Bridge
Use Trusted Relay Point*	Default

Save Delete Copy Reset Add New

*- indicates required item.

2.11 CUBE Configuration Details

CUCM Region and Device Pool Configuration for SP1 with CUBE

The following sections provide a brief description regarding CUBE's flow-through versus flow-around modes for media.

Flow Through Versus Flow Around Design Considerations

For the lab configuration where the Sonus NBS performed the transcoding, CUBE was configured in flow-through mode. A Cisco engineer should be consulted to determine the appropriate design based on the customer's requirements.

Using 4 Digit CUCM Phone Extensions

For PSTN-to-PSTN call transfers that use Call Forward Always (CFA), Call Forward On Busy (CFOB), and Call Forward No Answer (CFNA)

During the lab CUCM without CUBE SIP trunking evaluations for CUCM software versions 6.1.2.1000-13 and 7.0.1.11000-2, a diversion header issue was discovered where the NPA-NXX is not prefixed to the four digit extension number for the user portion of the diversion header. This problem affected PSTN-to-PSTN call transfers that use Call Forward Always (CFA), Call Forward On Busy (CFOB), and Call Forward No Answer (CFNA) where a four digit extension is used for the CUCM phones because the original caller ID is not delivered to the final PSTN destination. When the CUCM is configured to use CUBE running on a Cisco 2821 ISR, a SIP profile can be used to modify the diversion header to correct this. For the lab scenario a SIP profile was written for CUBE to modify the diversion header by adding the NPA-NXX to the 4 digit extension in the user portion of the diversion header.

Please note that a customer CUCM environment may have several NPA-NXXs or other phone extension digit configurations that will require different SIP profile rules and translation rules to be designed using CUBE that are unique to each customer environment. A Cisco engineer should be consulted for assistance.

CUBE SIP Profile Rules

The following sip profile rules need to be included in the CUBE "voice class sip-profiles" section of the configuration. These rules allow the Music on Hold to be heard at the destination phone when the call is placed on hold from the CUCM phone.

```
voice class sip-profiles 1
request REINVITE sdp-header Audio-Attribute modify "inactive" "sendrecv"
request ACK sdp-header Audio-Attribute modify "sendonly" "sendrecv"
response 200 sdp-header Audio-Attribute modify "sendonly" "sendrecv"
```

***CUBE DSCP Bit
Settings for
Signaling and
Media***

The CUBE voip dial peer default DSCP bit settings for signaling is AF31 and for media is EF. These settings do not appear in the configuration when a show configuration is executed at the CLI prompt. However, the settings can be displayed by executing the command “show dial-peer voice #” at the CLI prompt for each voip dial peer where the # represents the dial peer number.