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XO® SIP Service

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

XO SIP Packages 1 and 2, implemented without Cisco Unified Border Control Element (CUBE)

XO Communications

Contents

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1. Overview

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 without Cisco Unfied 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.

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 without 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.

- **Workarounds for incoming PSTN Calls Forwarded Off-Net to PSTN –**When incoming PSTN calls are delivered to desk phone with Call Forwarding enabled to an off-net PSTN phone, the calls will fail unless one of following workarounds are implemented:

  - **4-digit workaround** - Configure “First Redirect Number (External)” in the Calling Party Selection field and then optionally apply a phone number mask in the “Caller ID DN” field such that the resulting phone number is recognized by the XO network. Details of this configuration can be found in section 3.6.

    Limitation of 4-digit workaround for calls placed to off-net destinations: when calls are placed to an off-net destination, the originating caller-id may or may not be substituted, depending on the configuration.

    Limitation of 4-digit workaround for calls forwarded to off-net destinations: with the calling party selection of “First Redirected Number (External)”, any calls that originate off-net and are forwarded off-net will show the forwarding party’s caller ID, or the number resulting from the calling party mask. The person receiving the call will not see the caller-id of the original off-net PSTN caller.

  - **10-digit workaround for customers using a previously configured CUCM -** re-number all the CUCM extensions with valid 10-digit numbers that correspond with XO’s DIDs. “Redirecting Diversion Header Delivery” should be checked to allow the use of SIP diversion headers. The result of this is that the SIP Diversion headers are properly populated so that calls will not be rejected. Details of this configuration can be found in section 3.6.

    Limitation of 10-digit workaround for customers using a previously configured CUCM: most existing CUCM customers have deployed with 3, 4, or 5-digit extensions so that reconfiguring to use the SIP trunks with 10-digit numbers is not practical.

- **10-digit workaround for customers deploying a new CUCM -** new customers can implement 10-digit "extensions" but use shorter 3-5 digit extensions for dialing purposes. This requires CUCM transformation patterns, in conjunction with partitions and calling search spaces, to enable translations from the dialed short extensions to the actual directory numbers. A transformation pattern such as "4xxx" would capture all numbers dialed in the 4000-4999 range, which the system would then expand to a number such as "4693874xxx". Partitioning and calling search spaces would need to be employed to make the application of the transformation patterns granular enough that it would not interfere with calls to numbers that should NOT be expanded, such as "4085551212," which would match as soon as "4085" were dialed. For the convenience of users, line labels for the phones could be deployed to display the short extensions on each line. "Redirecting Diversion Header Delivery" should be checked to allow the use of SIP diversion headers.

Known Issues for XO SIP Service Package 2 only without CUBE:

- When using XO SIP Service Package 2, G.711 pass-through for fax is not supported
- CUCM does not provide a software based MTP for G.729a as is available when the SIP trunk is configured for MTP using the G.711 codec to support Service Package 1. For this reason an external device such as a Cisco 2821 ISR must be used to provide conference bridging and software MTP resources. The configuration commands that are required in the Cisco 2821 ISR to support conference bridging and software MTPs are addressed in section 3.8. Cisco technical assistance may be needed for the design requirements of this external device.
- When configuring the CUCM for XO SIP Service Package 2, the CUCM Music on Hold Server codec must be set accordingly to G.729, as shown in section 3.8.

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 not supported

2. Testing of CUCM 7.0.1

2.1. Software and Hardware Versions Tested

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 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

3. CUCM and Cisco Unity Connection (CUC) PC GUI Access
   a. Microsoft Internet Explorer version 6.0.2900.2180.xpsp_sp2_gdr.080814-1233

4. Cisco Phones
   a. Cisco 7961
   b. Software Version: SCCP41.8-4-1S

5. Cisco 2432-24FXS IAD
   a. Software Version: c2430-mz.xo

6. Cisco Catalyst 3560 PoE series P-24
   a. Software Version: c3560-advipservicesk9-mz.122-44.SE2.bin

3. CUCM 7.0.1 Configuration

CUCM Configuration Screen Captures for XO SIP Packages 1 (G. 711) and 2 (G.729)

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

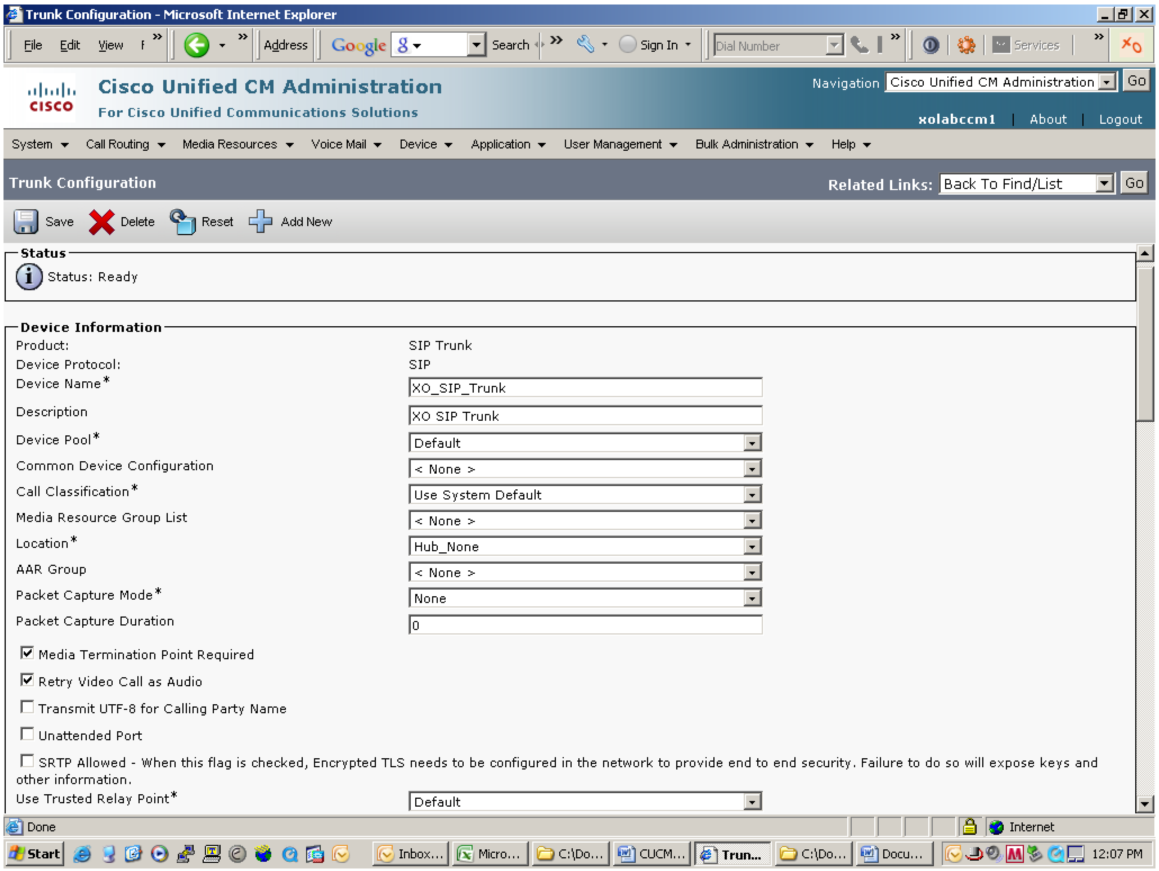
XO performed the minimum amount of configuration required to achieve successful completion of test calls over XO SIP. It is beyond the scope of this document and the testing efforts to show a complete CUCM 7.0.1 configuration, therefore screenshots of the GUI interface are provided only for the details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

Refer to the CUCM administration guides for additional configuration options, at http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_maintenance_guides_list.html.

Design guidance for CUCM and Cisco Unified Communications products based on CUCM may be found in the Solution Reference Network Design (SRND) guides found at http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_implementation_design_guides_list.html

3.1 SIP Trunk Device Information

Figure 1, SIP Trunk Device Information

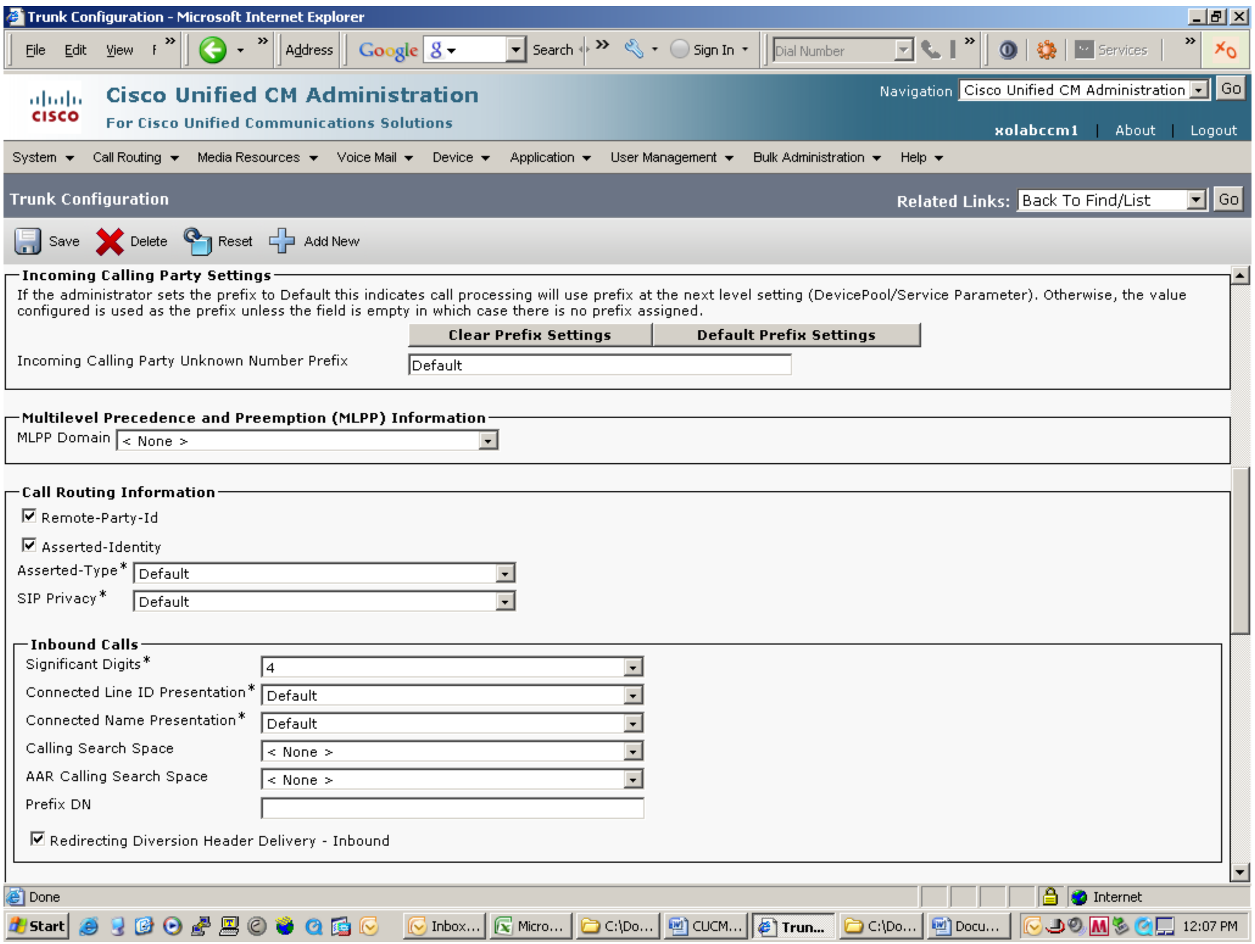


These are default settings for CUCM SIP trunks when CUBE is not used:

- Device Name, Description, and Pool are unique to the customer installation and are only locally significant.
- **MTP is checked to require CUCM to insert a Media Termination Point into the call path. MTPs are required when CUBE is not present to provide SIP DO/EO (delayed offer/early offer) conversions. If MTP is not checked, call hold and music on hold will not work.**

3.2 SIP Trunk Inbound Call Options

Figure 2, SIP Trunk Inbound Call Options



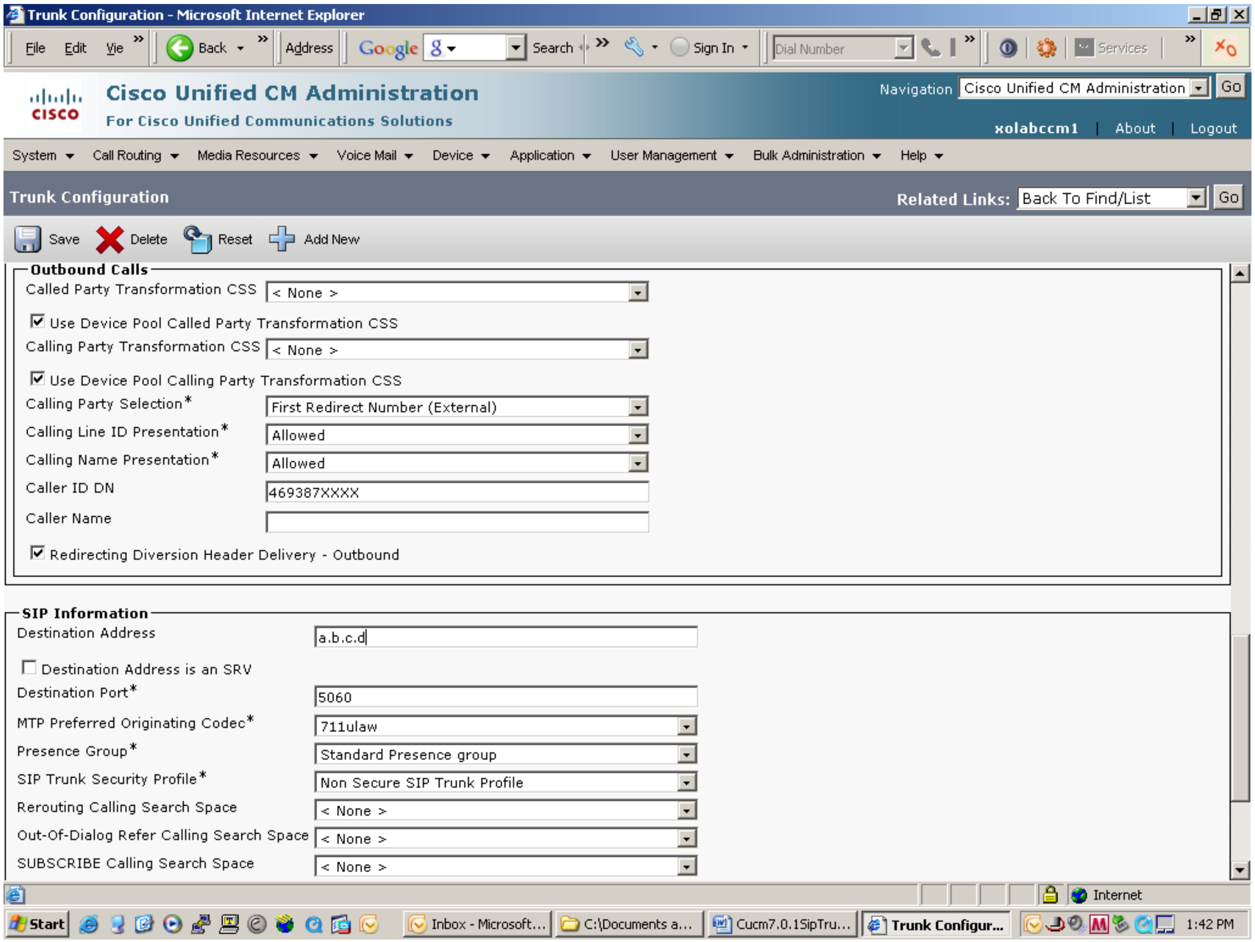
The inbound and outbound call handling section allows control of information elements such as calling number presentation on a per-trunk basis

Note that for inbound calls, only 4 significant digits are accepted by CUCM for processing. This setting is based on using 4 digits for extensions and simplifies the dial-plan.

Please see the Known Issues in section 1 for more details.

3.3 SIP Trunk Outbound Call Options and SIP Information

Figure 3, SIP Trunk Outbound Call Options and SIP Information

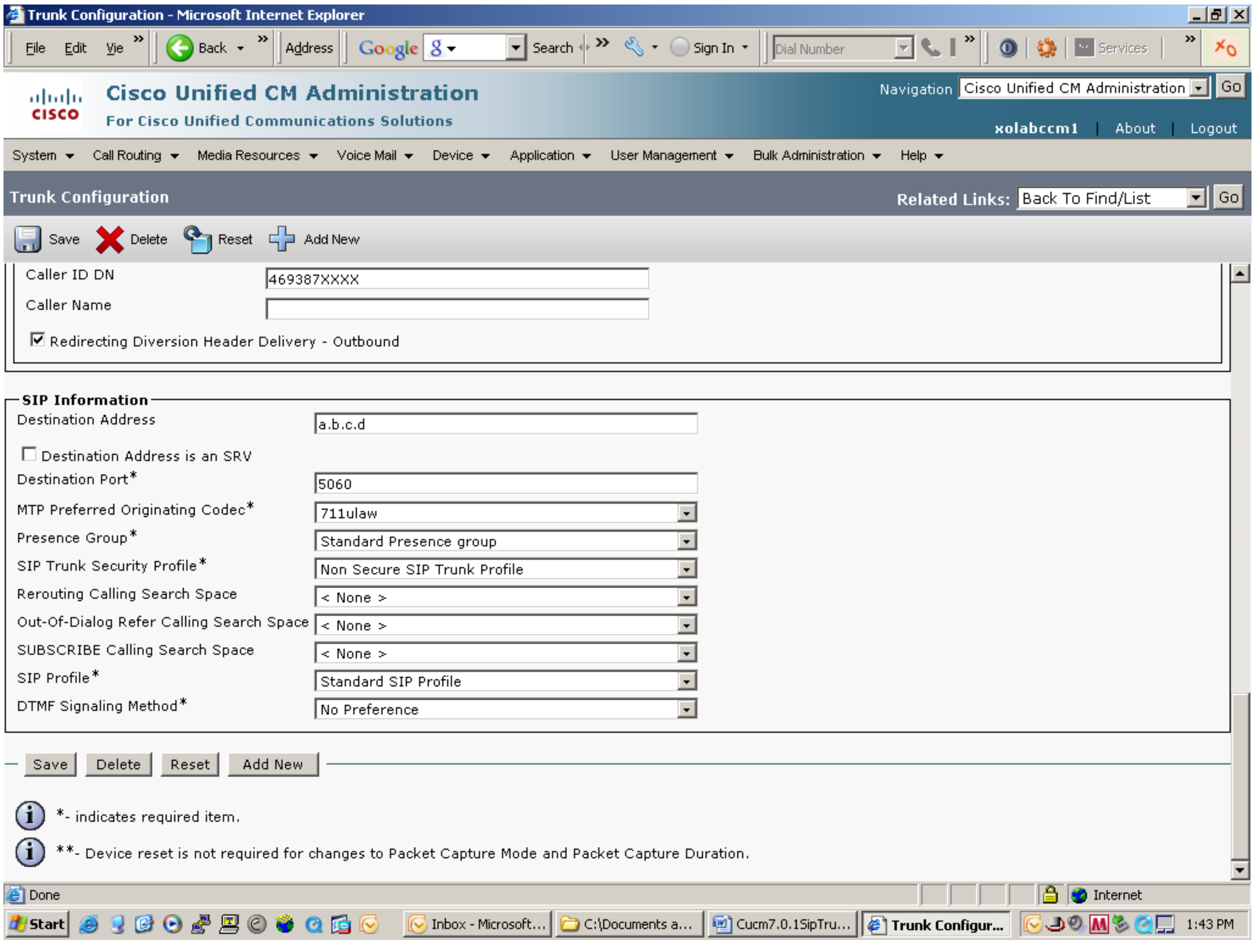


The destination endpoint of the CUCM SIP trunk is to the IP address of XO's SBC, to port 5060.

The IP address shown is for demonstration purposes only; the actual address is assigned by XO.

The remaining settings are defaults and most depend on customer dial-plan provisioning, which is unique for each customer.

Figure 4, SIP Information, cont.



3.4 SIP Profile

Figure 5, SIP Profile Screen Capture Part 1

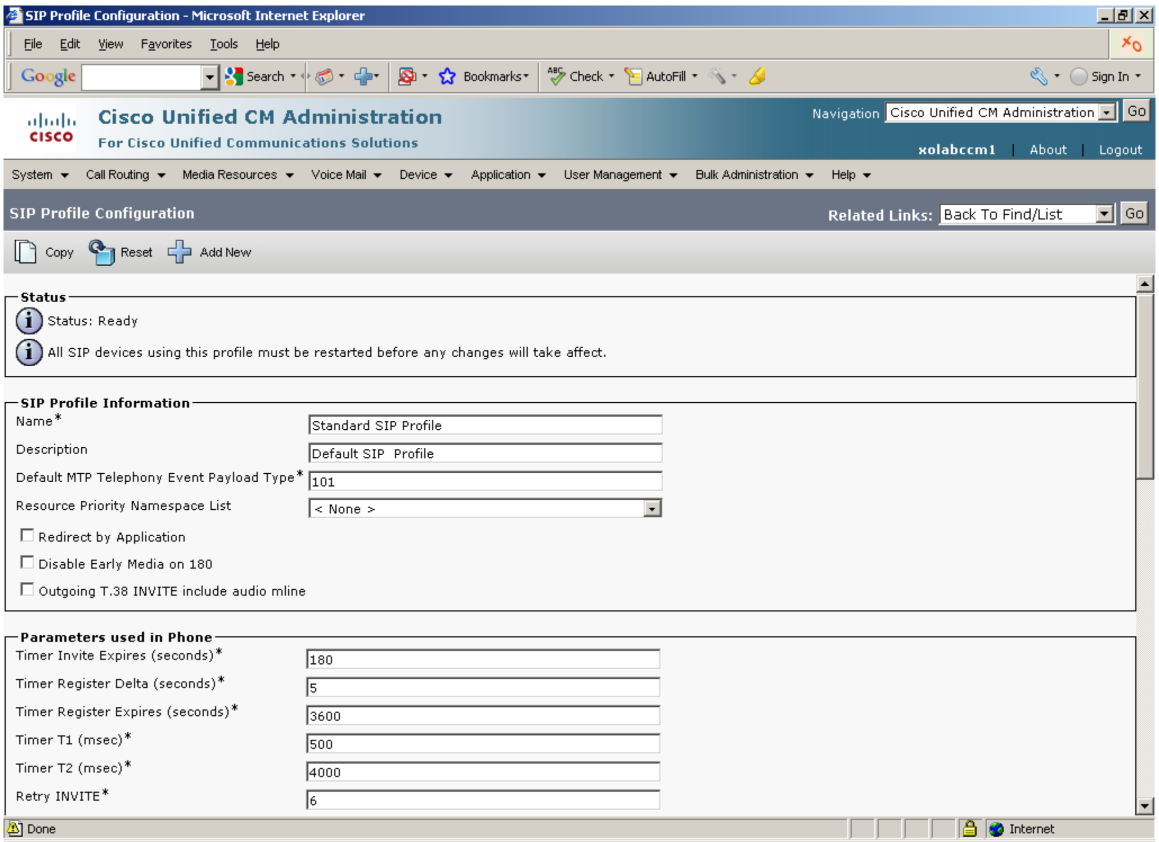


Figure 6, SIP Profile Screen Capture Part 2

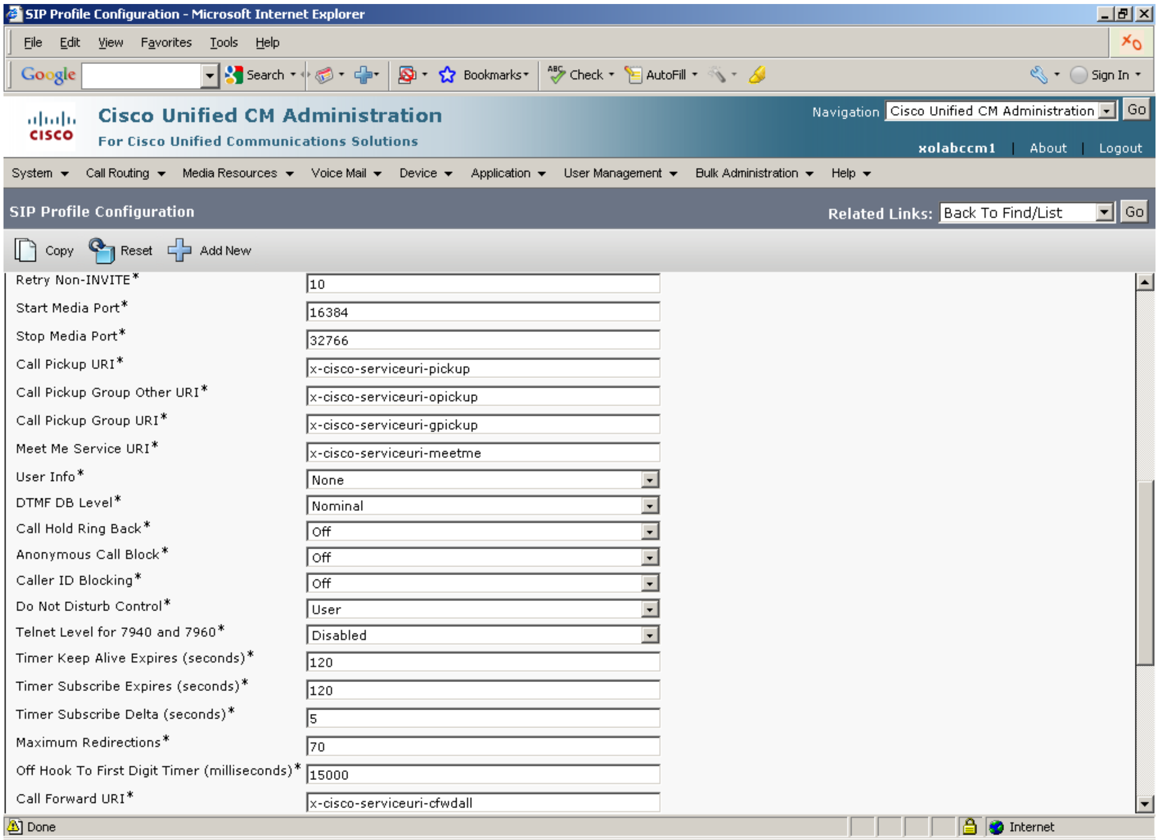
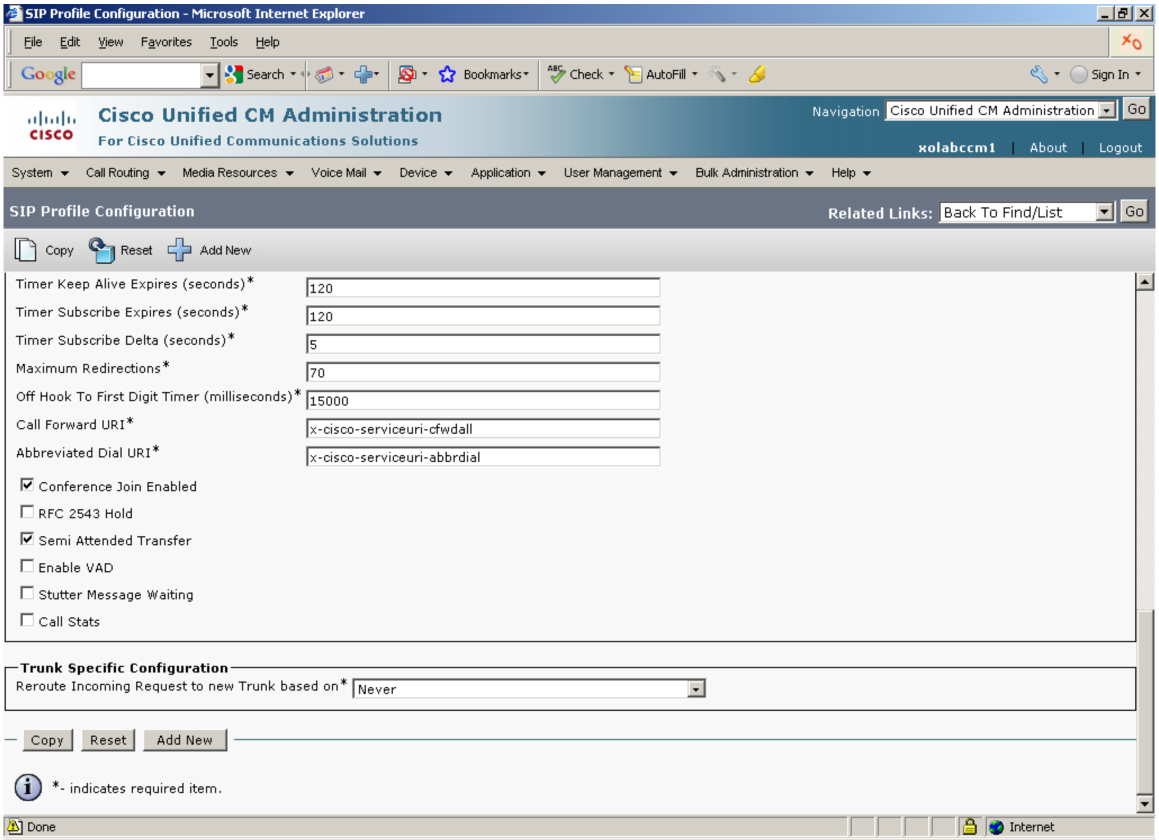
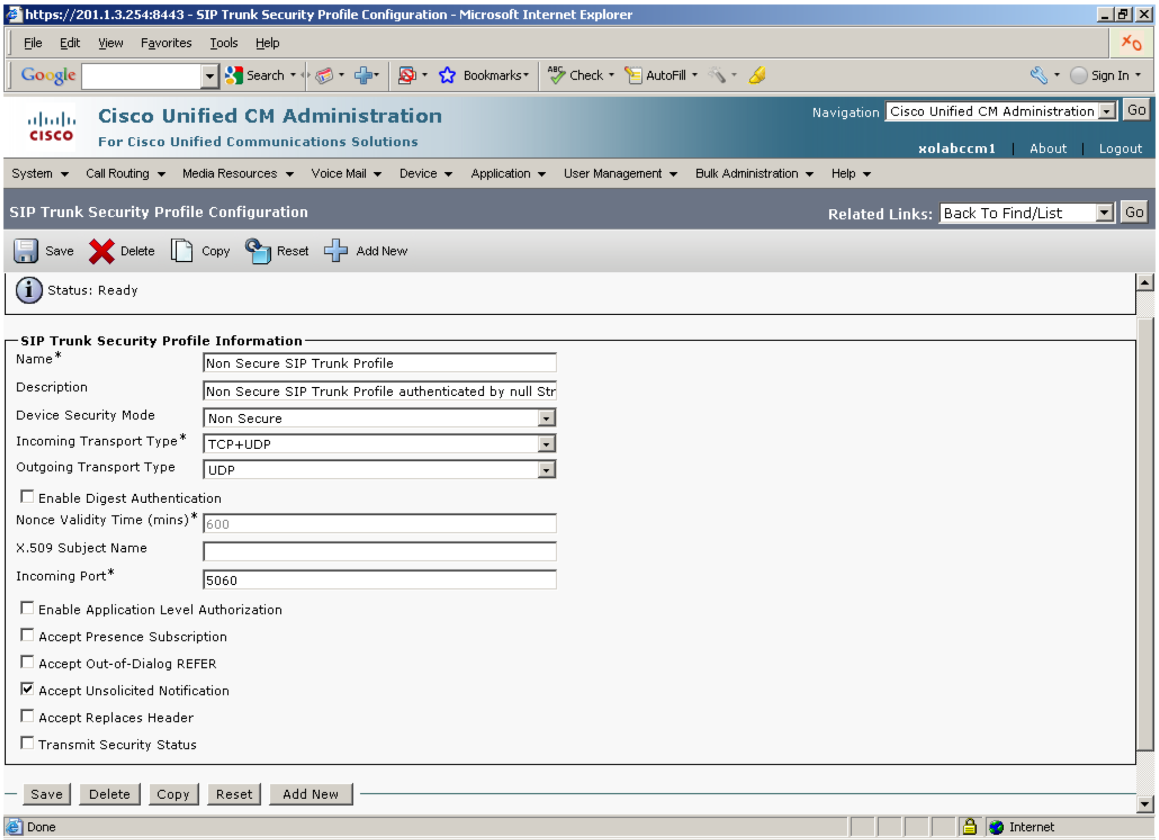


Figure 7, SIP Profile Screen Capture Part 3



3.5 CUCM SIP Trunk Security Profile

Figure 8, SIP Trunk Security Profile Screen Capture



3.6 Work-Arounds for Calls forwarded Off-net

As explained in the Known Issues in section 1, when incoming PSTN calls are delivered to desk phone with Call Forwarding enabled to an off-net PSTN phone, the calls will be forwarded but the originating caller ID will not be passed. This section provides screen captures for the optional 4-digit workaround and the optional 10-digit workaround *The limitations of each noted in the Known Issues section should be taken into consideration prior to implementation.*

4-digit workaround—This SIP trunk configuration scenario allows the transfer of PSTN-to-PSTN calls for Call Forward Always (CFA), Call Forward On Busy (CFOB) and Call Forward No Answer (CFNA) using a four digit extension on the CUCM phones.

Configure “First Redirect Number (External)” in the Calling Party Selection field and then optionally apply a phone number mask in the “Caller ID DN” field such that the resulting phone number is recognized by the XO network. A number is concatenated with the local 4-digit extensions to create valid DIDs that will not be rejected as a calling number. This is possible when the local extensions can be mapped to a contiguous range of DIDs assigned by XO. Alternately, this field may be populated with a single “main” number that reflects the main Operator number of the CUCM.

In the "SIP Trunk Screen Capture Using a Four Digit Phone Extension Part 2" 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 First Redirect Number (External) and the Caller ID DN is set to 469387XXXX. The option for Redirecting Diversion Header Delivery - Outbound is checked.

Figure 9, SIP Trunk Screen Capture Using a Four Digit Phone Extension Part 1

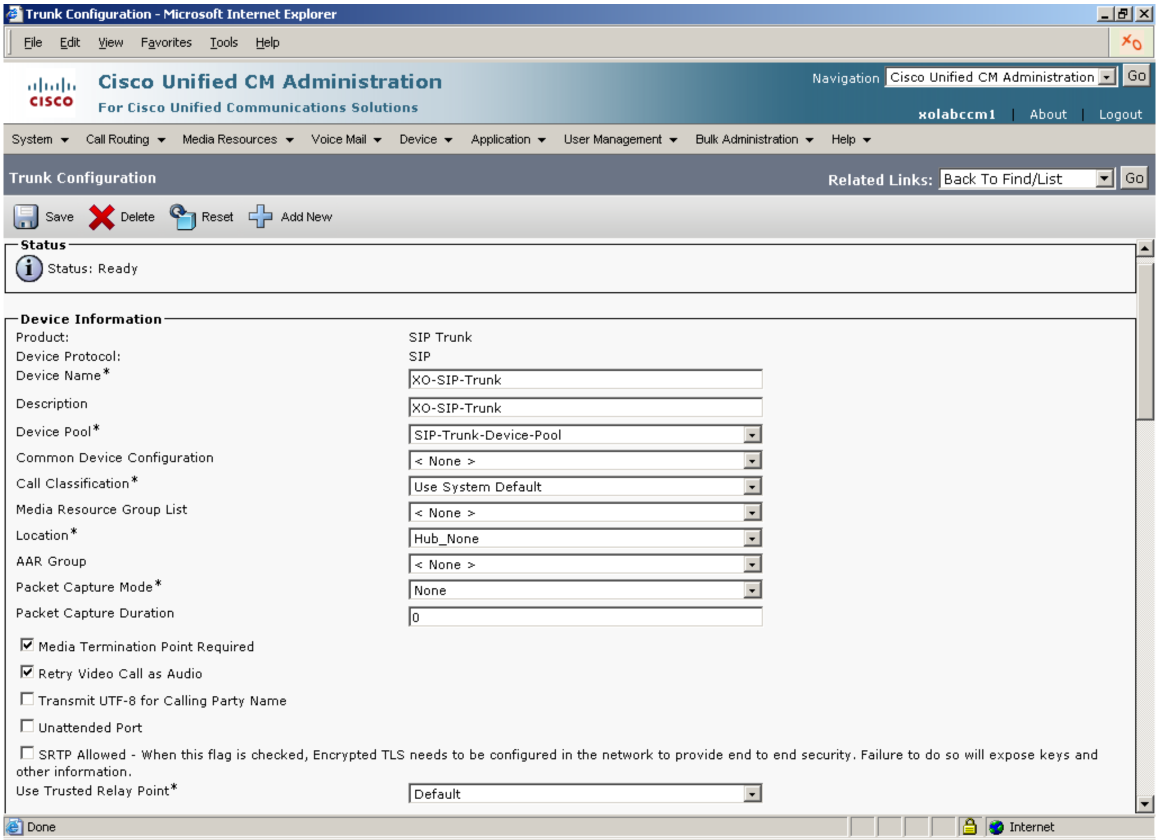


Figure 10, SIP Trunk Screen Capture Using a Four Digit Phone Extension Part 2

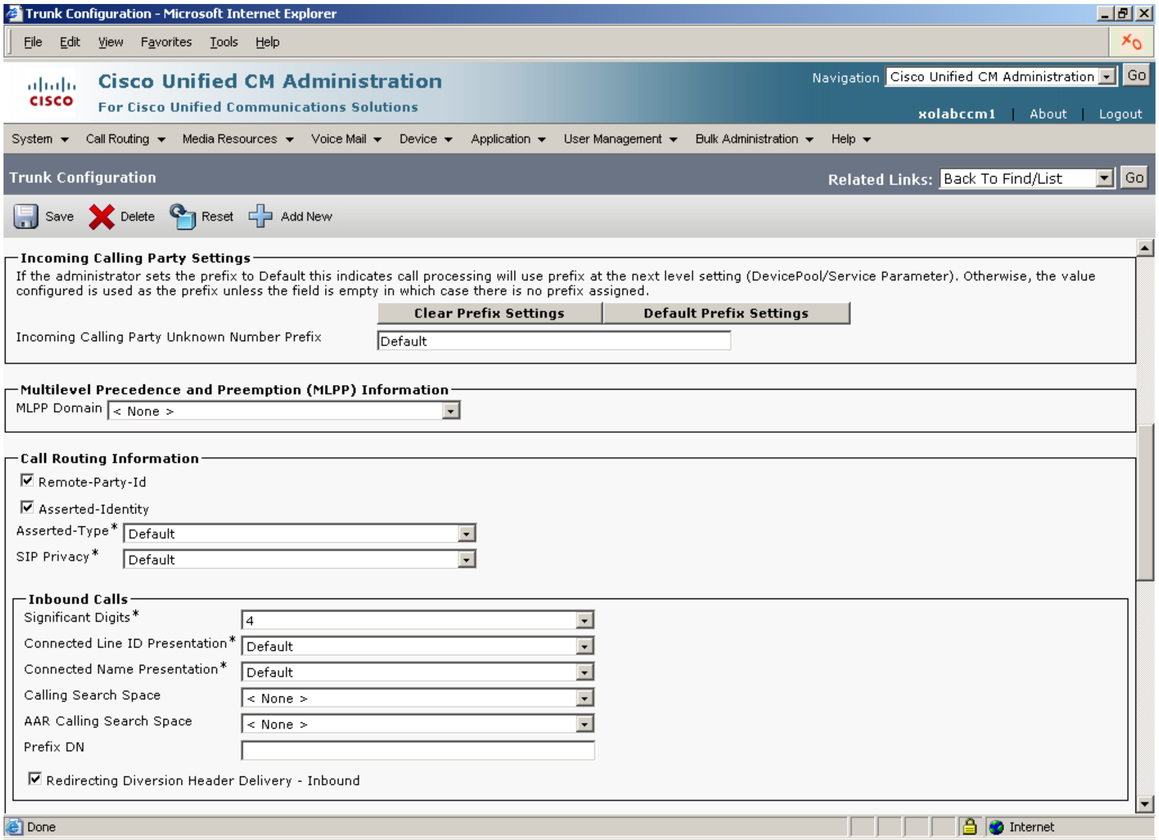


Figure 11, SIP Trunk Screen Capture Using a Four Digit Phone Extension Part 3

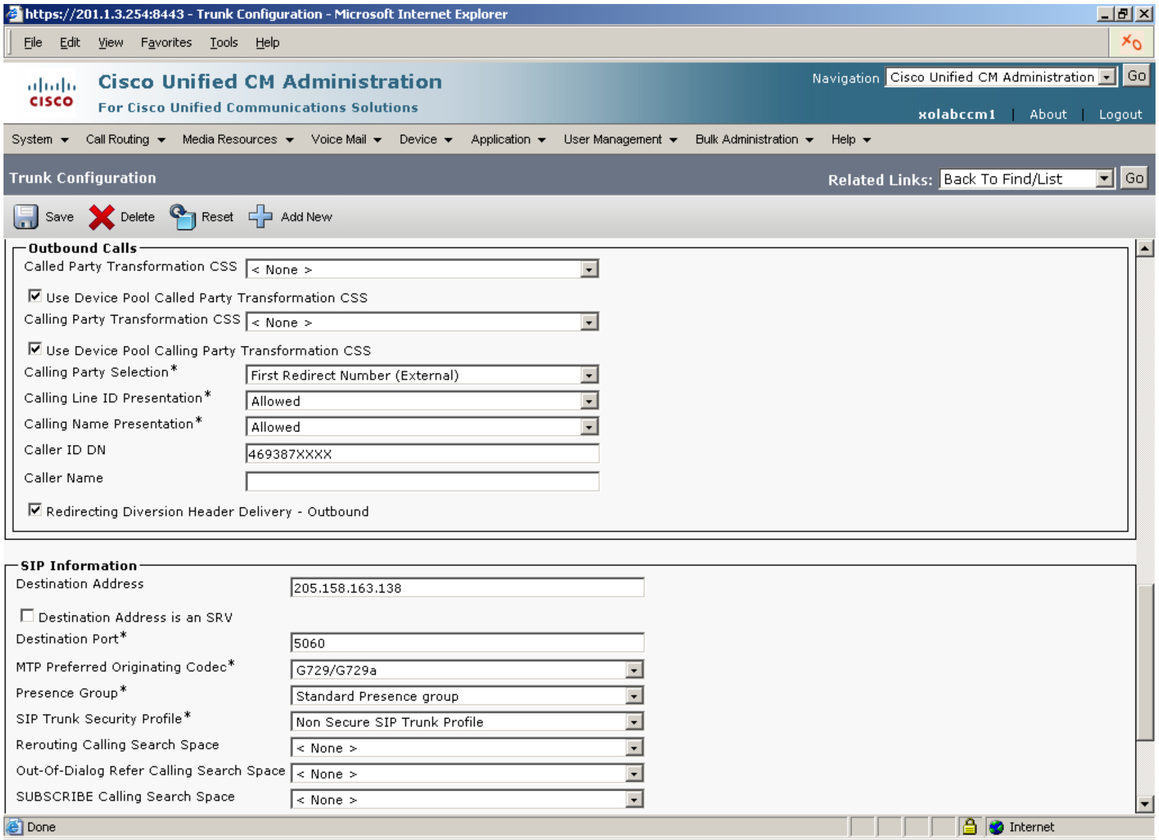


Figure 12, SIP Trunk Screen Capture Using a Four Digit Phone Extension Part 4

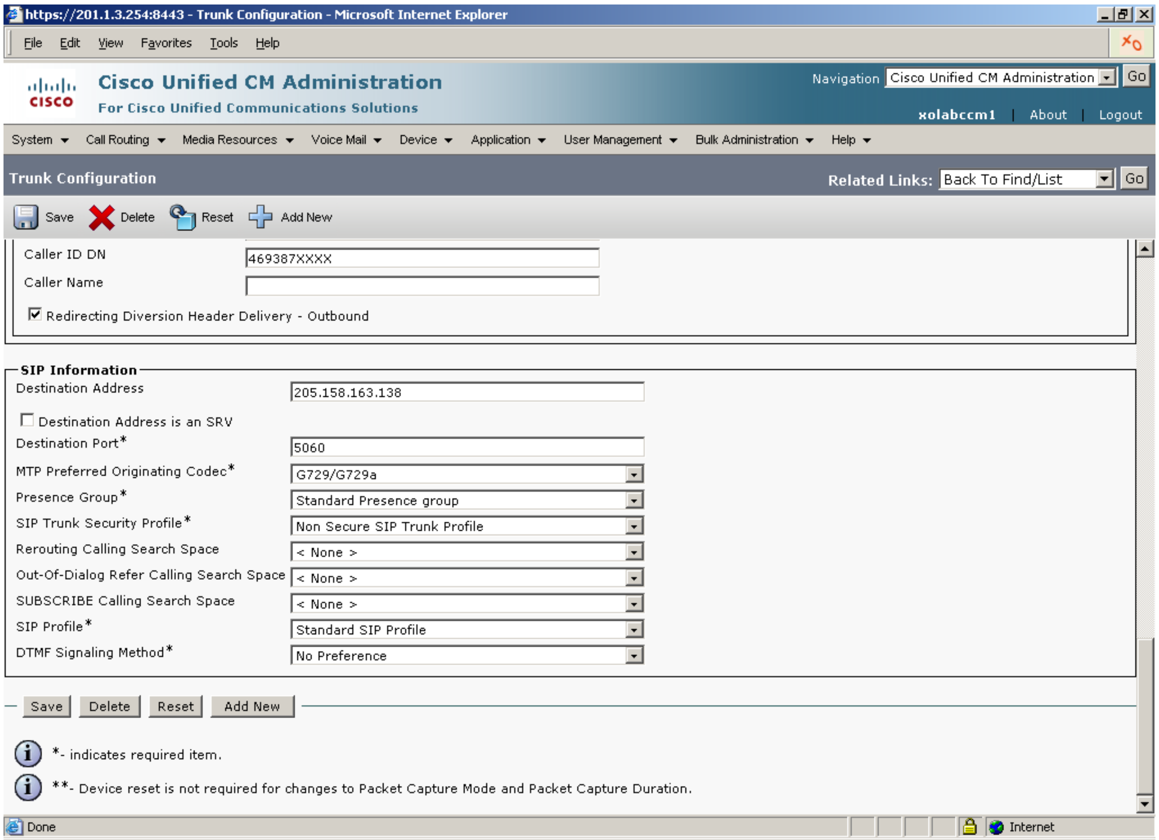
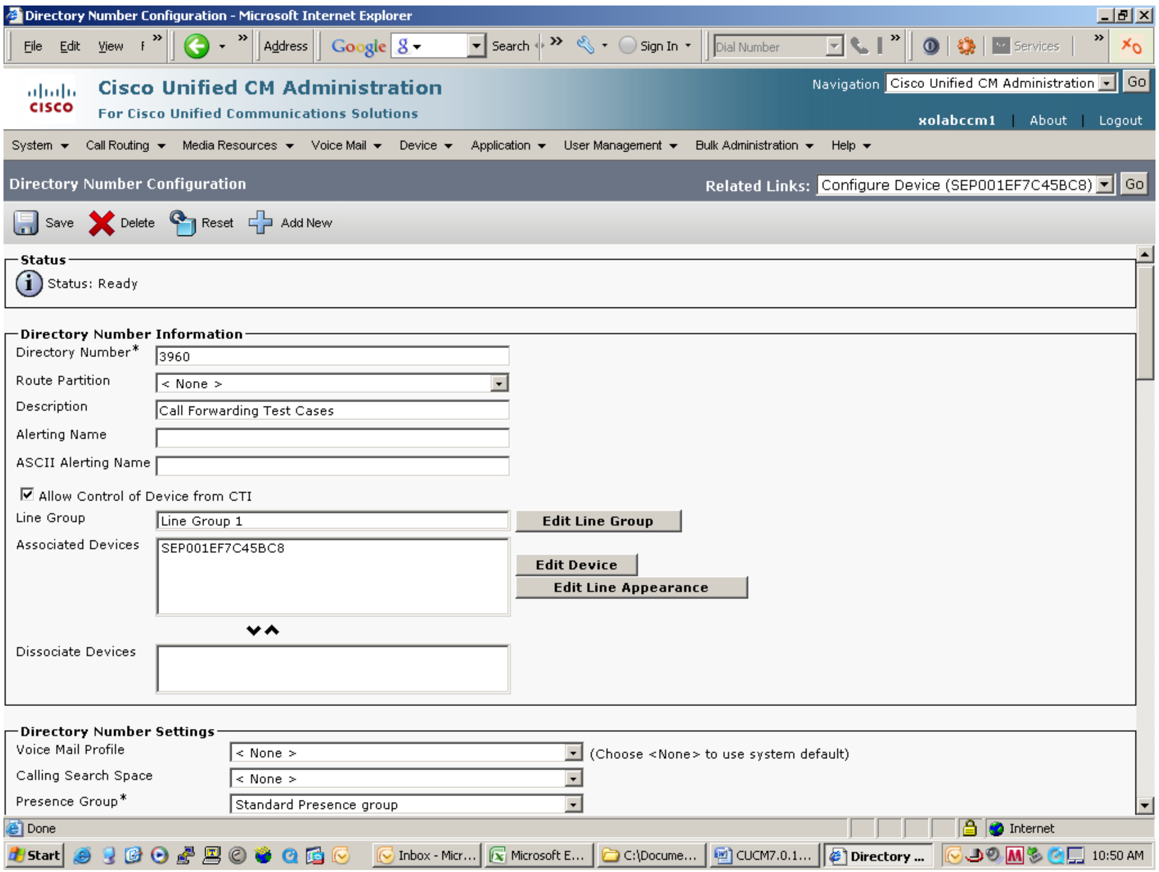


Figure 13, CUCM Phone Screen Capture Using a Four Digit Extension



10-digit workaround—In "SIP Trunk Screen Capture Using a Ten Digit Phone Extension Part 2" within the Call Routing Information, the Inbound Calls section has the "Significant Digits*" set to 10 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. The option for Redirecting Diversion Header Delivery - Outbound is checked. When using a 10-digit phone extension with XO SIP Service Pack 2 (G.729a), please see the additional configuration required, shown in section 3.8.

Figure 14, SIP Trunk Screen Capture Using a Ten Digit Phone Extension Part 1

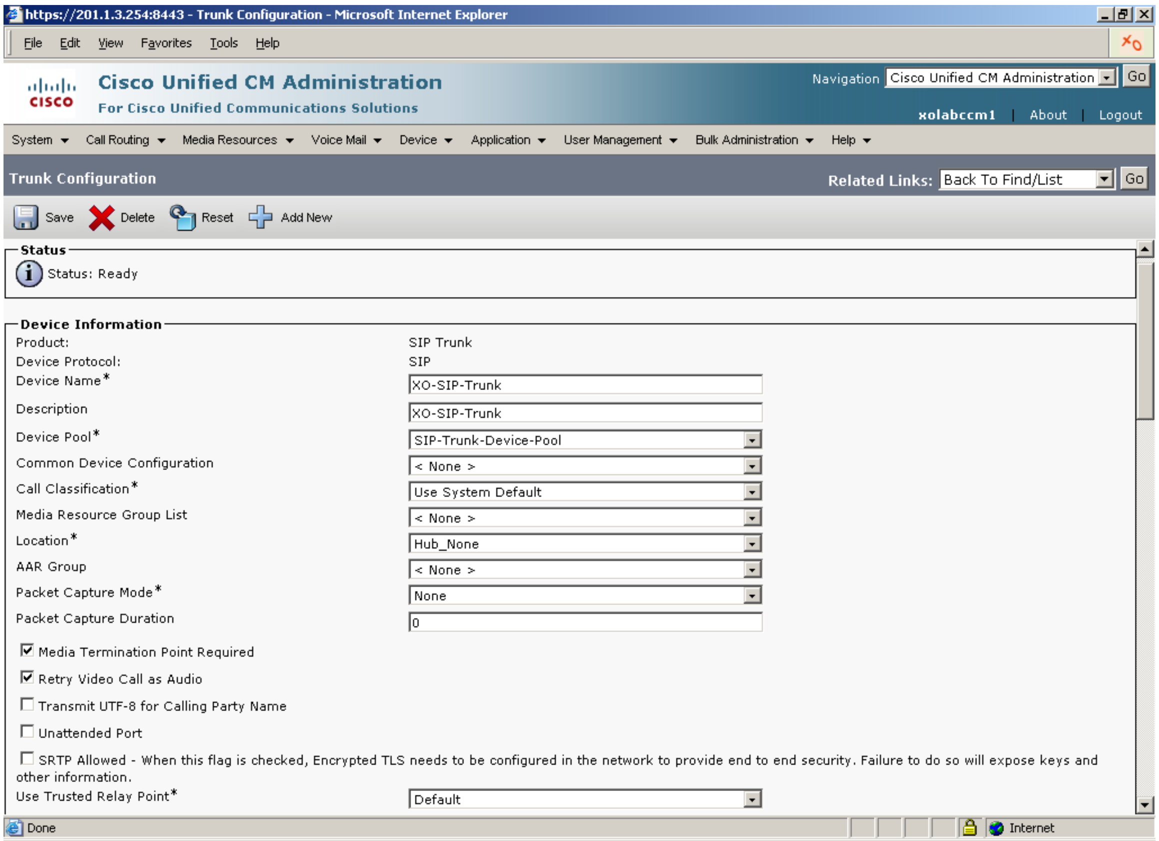


Figure 15, SIP Trunk Screen Capture Using a Ten Digit Phone Extension Part 2

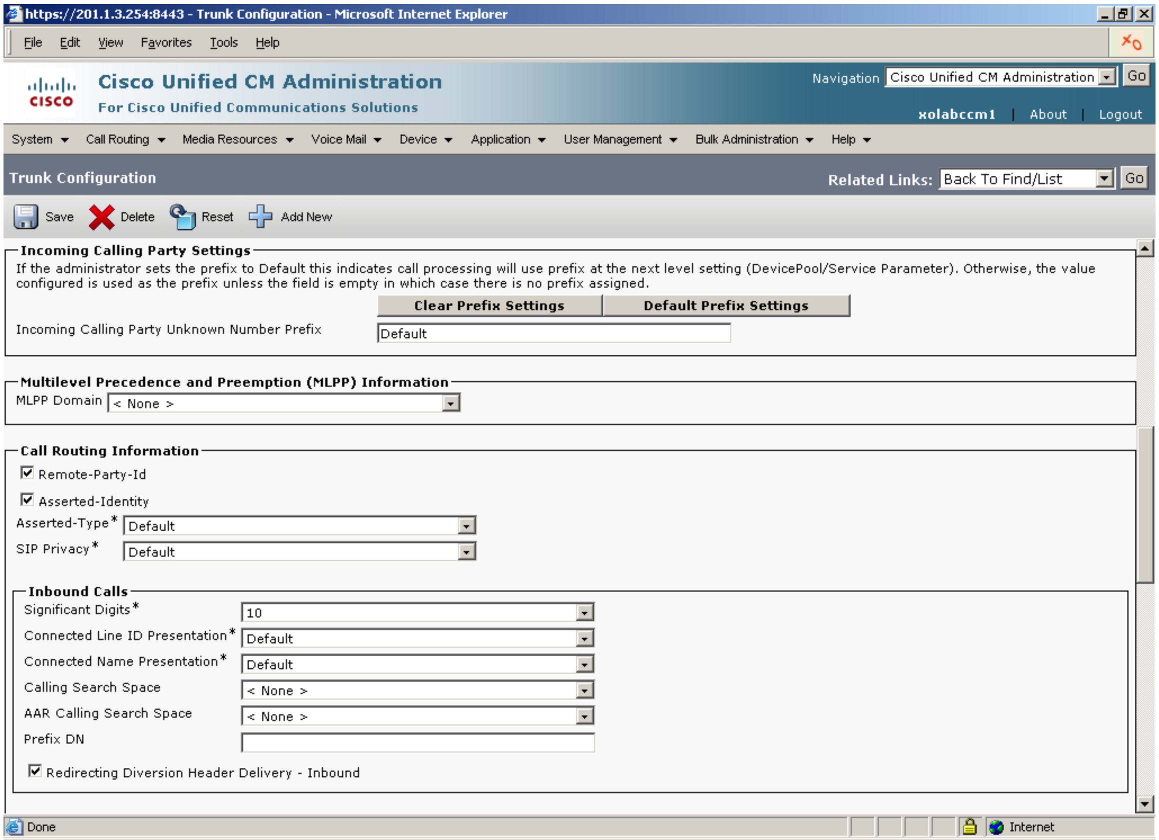


Figure 16, SIP Trunk Screen Capture Using a Ten Digit Phone Extension Part 3

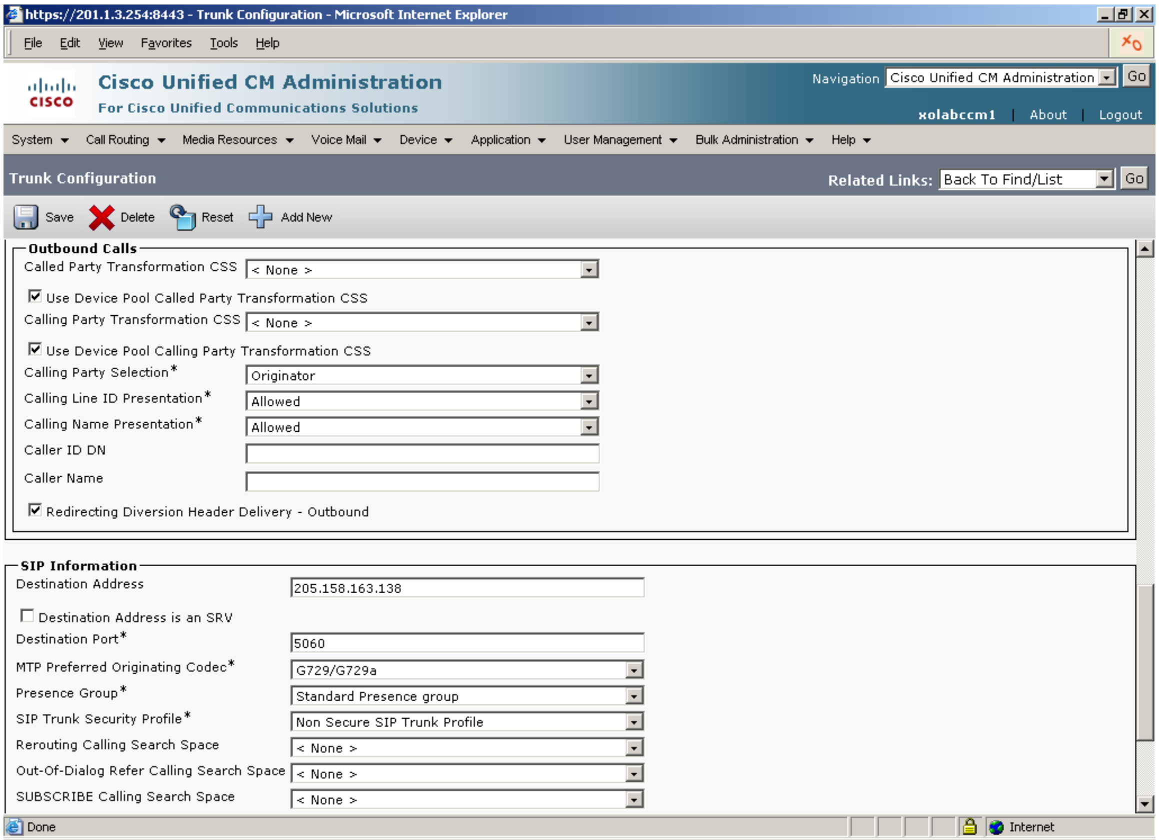
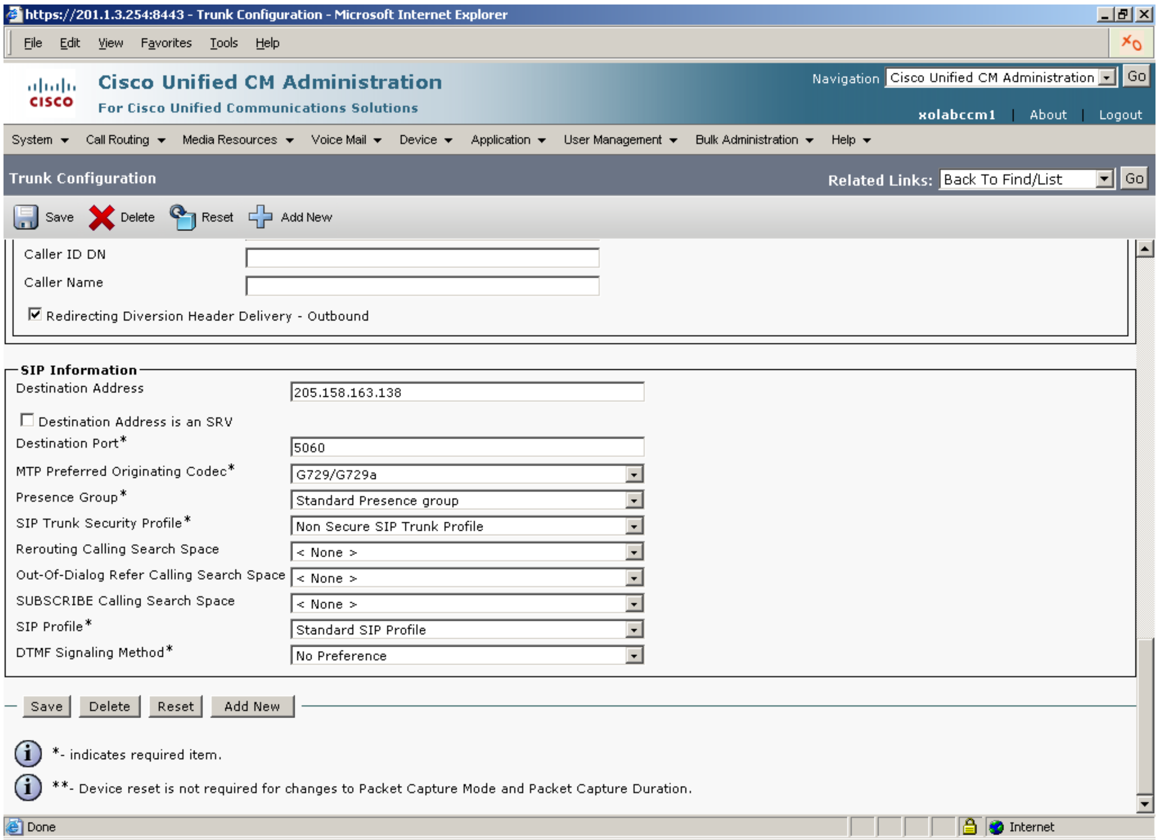


Figure 17, SIP Trunk Screen Capture Using a Ten Digit Phone Extension Part 4

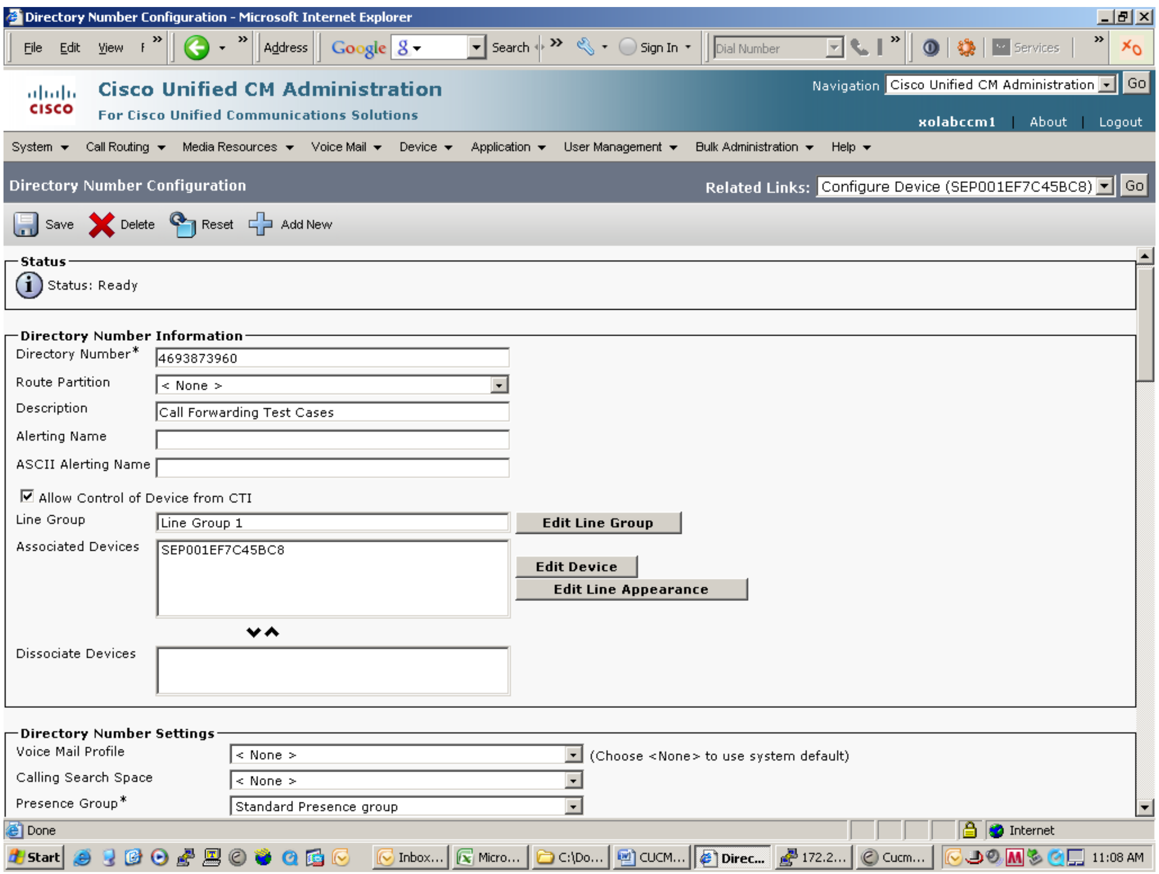


3.8 Additional Configurations for XO SIP Package 2 (G.729 compression) with CUCM 7.0 without CUBE

Configuring SP2 Using a Ten digit Phone Extension.

The following screen shows CUCM SIP Trunk Additional Information for Configuring SP2 Using a Ten digit Phone Extension. The CUCM phones must have a 10-digit extension.

Figure 18, CUCM Phone Screen Capture Using a Ten Digit Extension



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 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.

Figure 19, CUCM CLID Blocked SIP Trunk Screen Capture Part 1

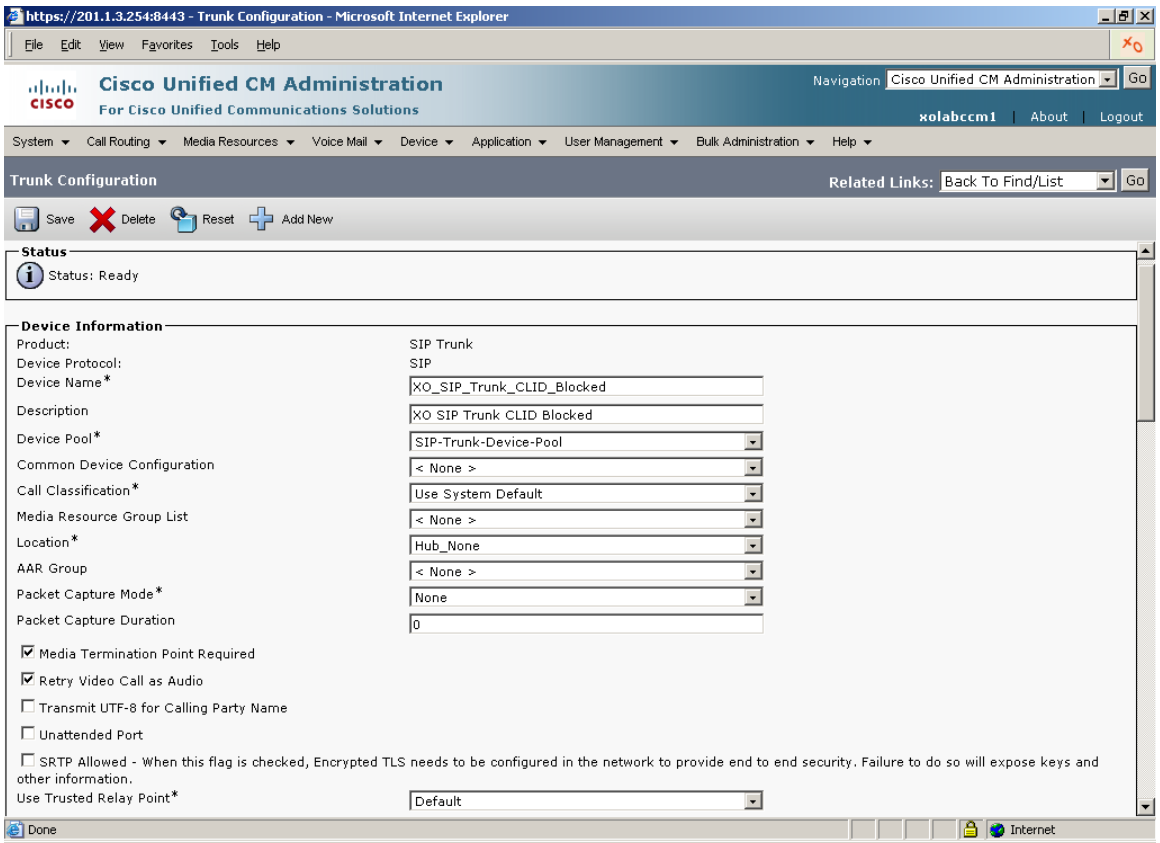


Figure 20, CUCM CLID Blocked SIP Trunk Screen Capture Part 2

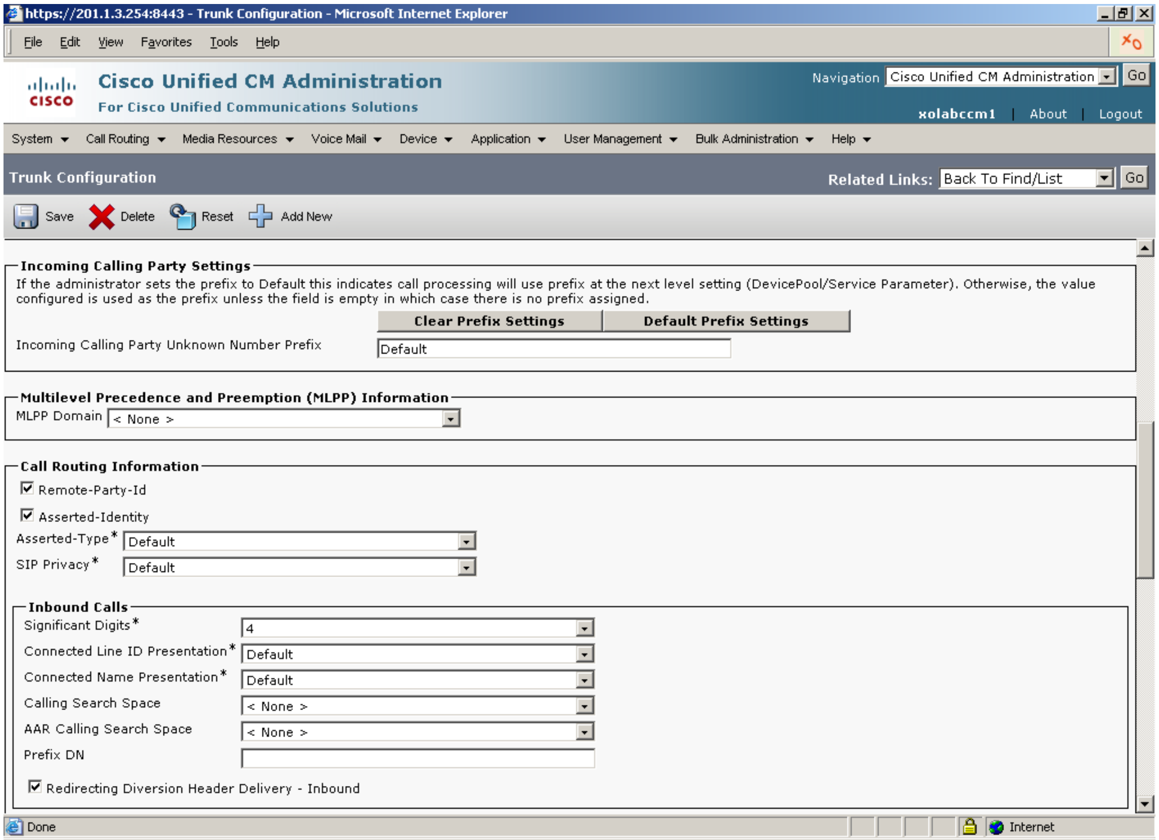


Figure 21, CUCM CLID Blocked SIP Trunk Screen Capture Part 3

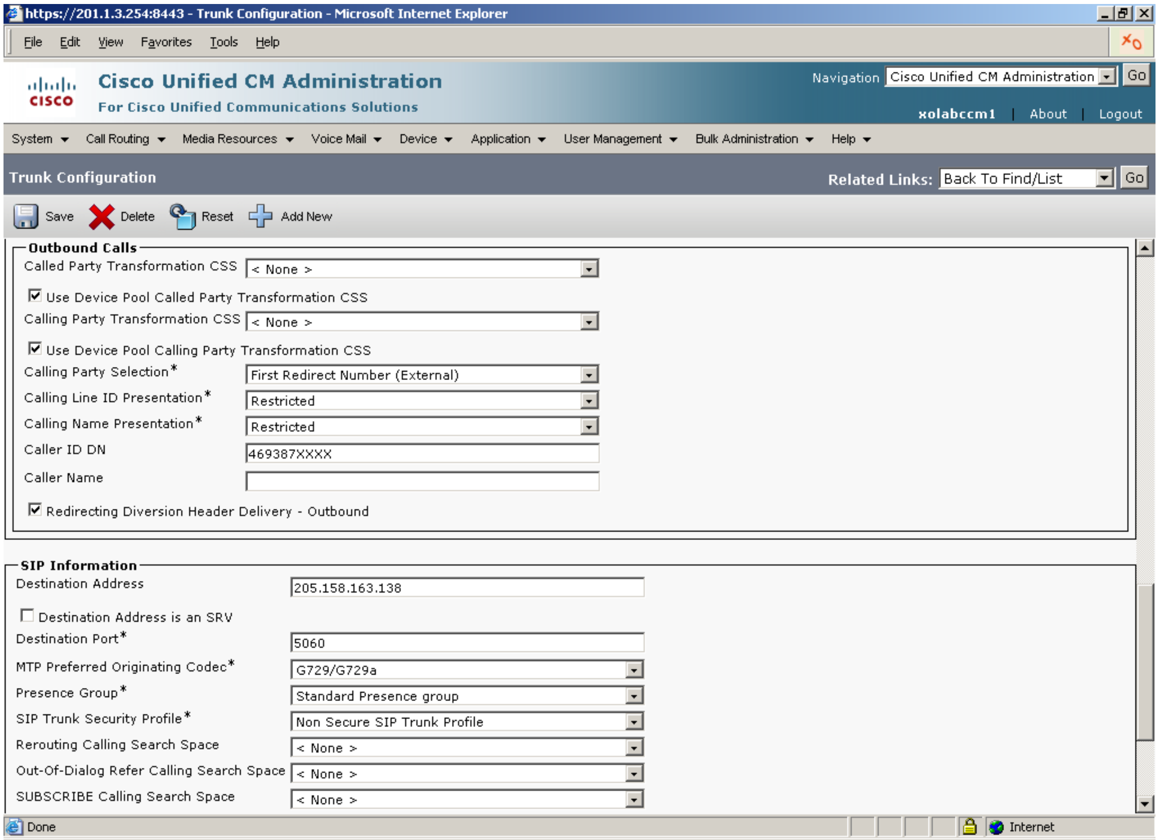
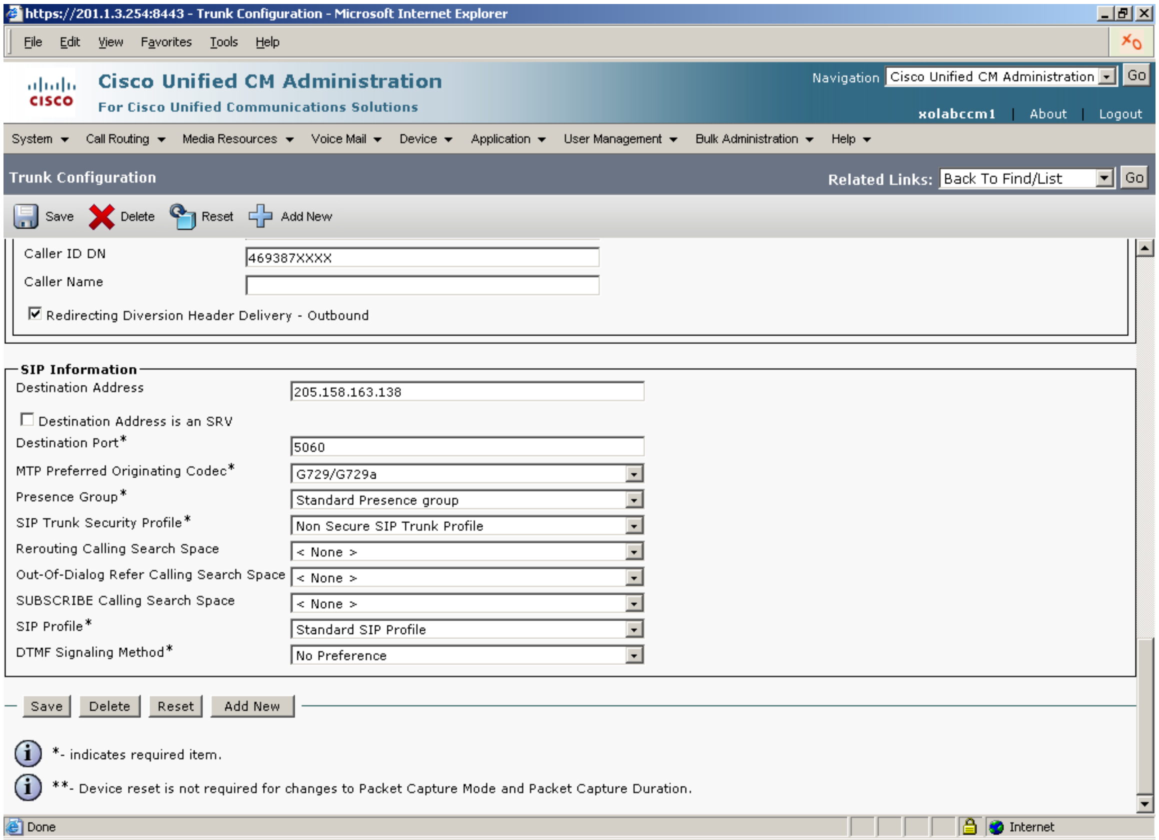


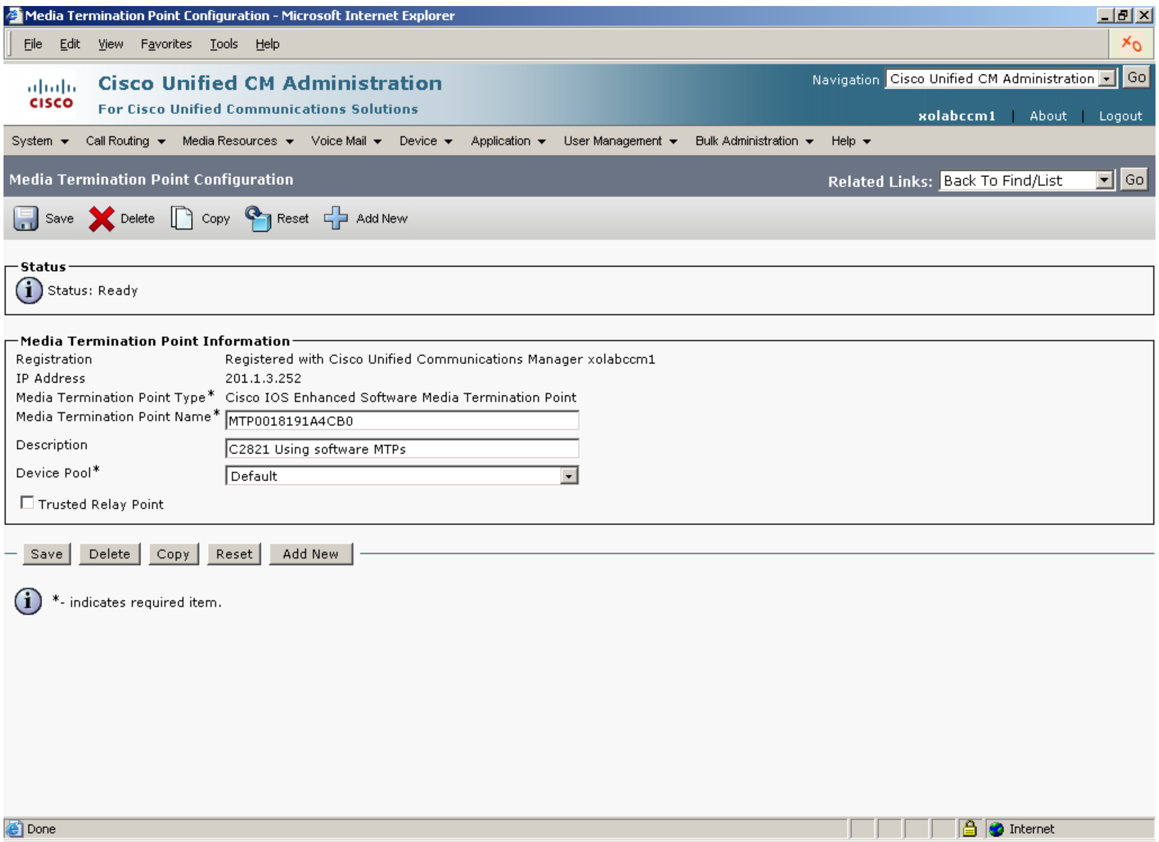
Figure 22, CUCM CLID Blocked SIP Trunk Screen Capture Part 4



CUCM MTP Configuration Screen Capture for SP2

The screen capture below shows the MTP resource parameters listed under CUCM Administration, Media Resources, Media Termination Point. An external MTP resource is configured using a Cisco 2821 ISR which registers with the CUCM. This MTP resource was used in testing all SP2 test calls. When the CUCM is configured for SP1 which uses the G.711 codec, the CUCM uses its own software MTP to process calls. However, when the CUCM is configured for SP2 which uses the G.729 codec, the MTP must be supported on an external device such as a 2800 or 3800 series ISR. This screen can also be used to verify that the Cisco 2821 ISR MTP resource is registered with the CUCM by checking the registration state.

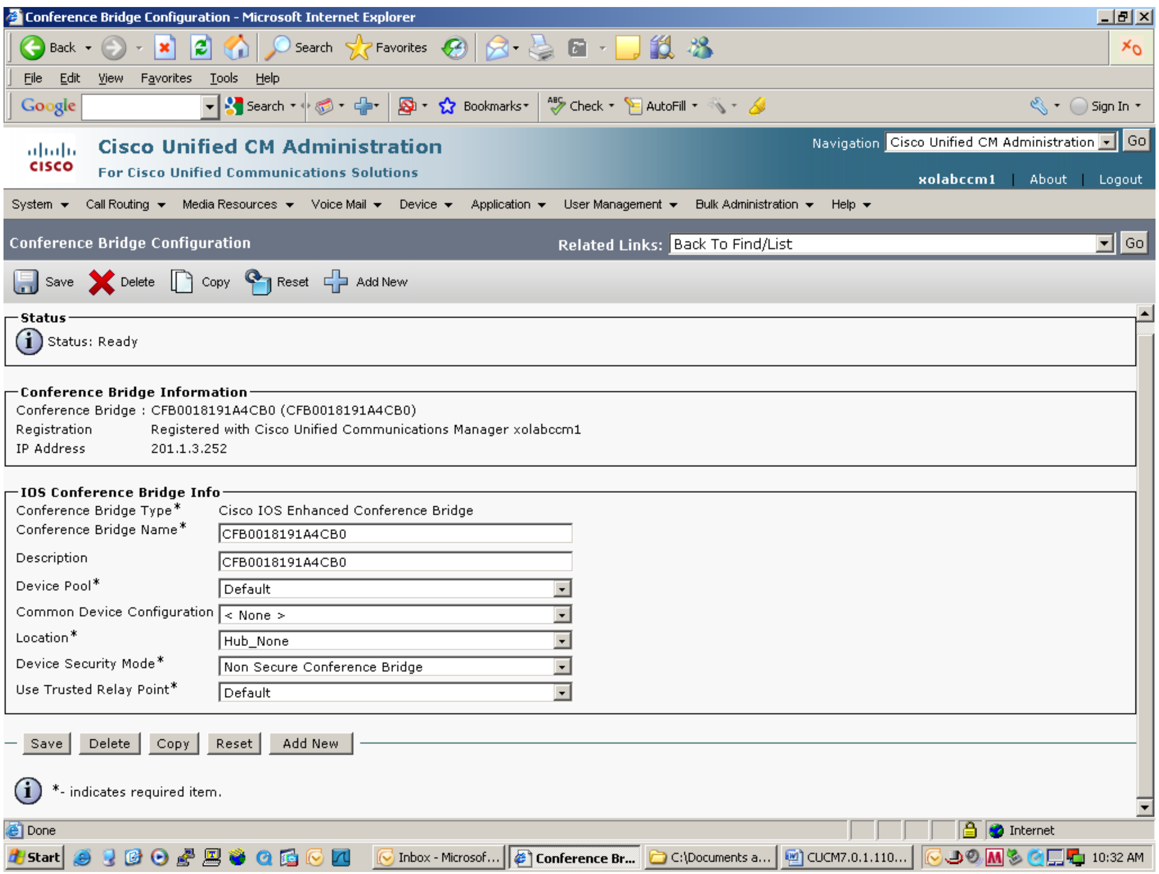
Figure 23, CUCM MTP Configuration Screen Capture



CUCM Conference Bridge Screen Capture for SP2

The screen capture below shows the Conference Bridge resource parameters listed under CUCM Administration, Media Resources, Conference Bridge. An external conference bridge resource is configured using a Cisco 2821 ISR 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 a 2800 or 3800 series ISR. This screen can also be used to verify that the Cisco 2821 ISR conference bridge resource is registered with the CUCM by checking the registration state.

Figure 24, CUCM Conference Bridge Configuration Screen Capture



CUCM Music on Hold Server Codec Selection Screen Captures

When configuring the customer for 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.729 for SP2. The system default for the Music on Hold server codec setting is G.711ulaw. 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 25, Music on Hold Server Codec Screen Capture Part 1

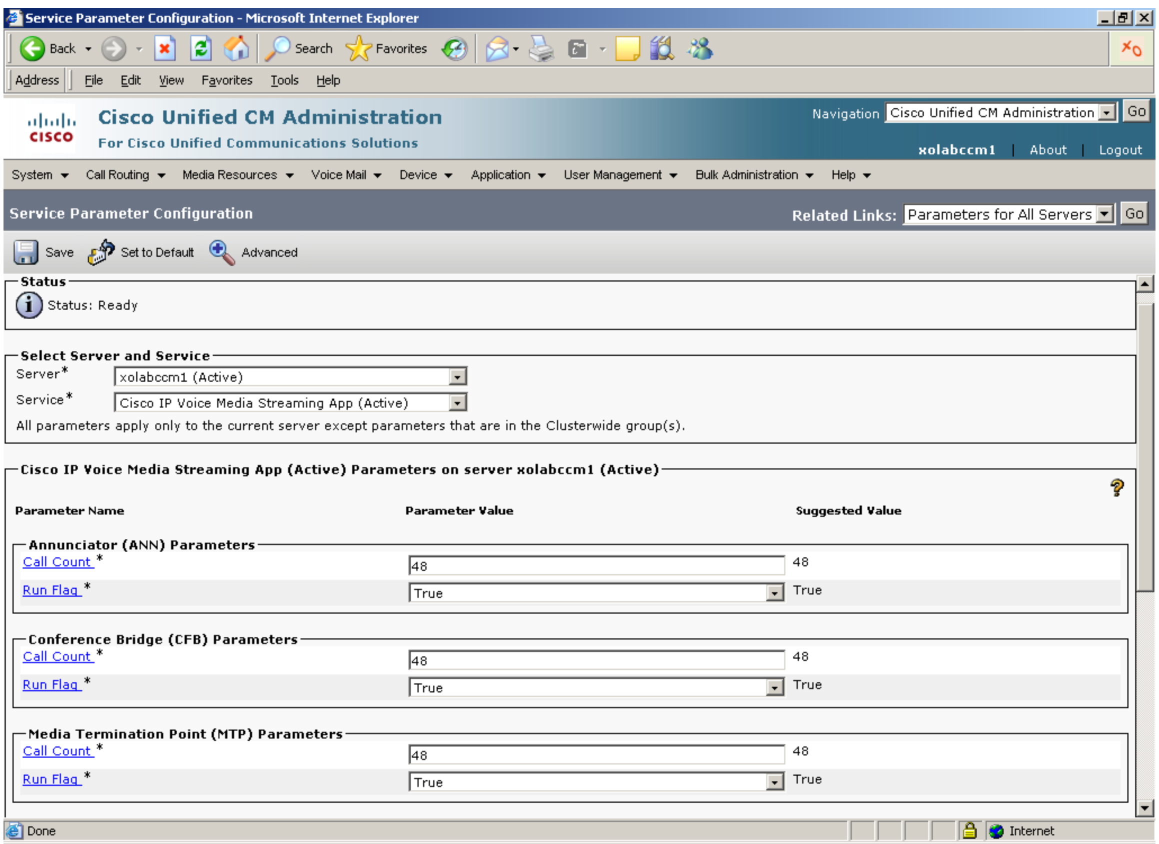
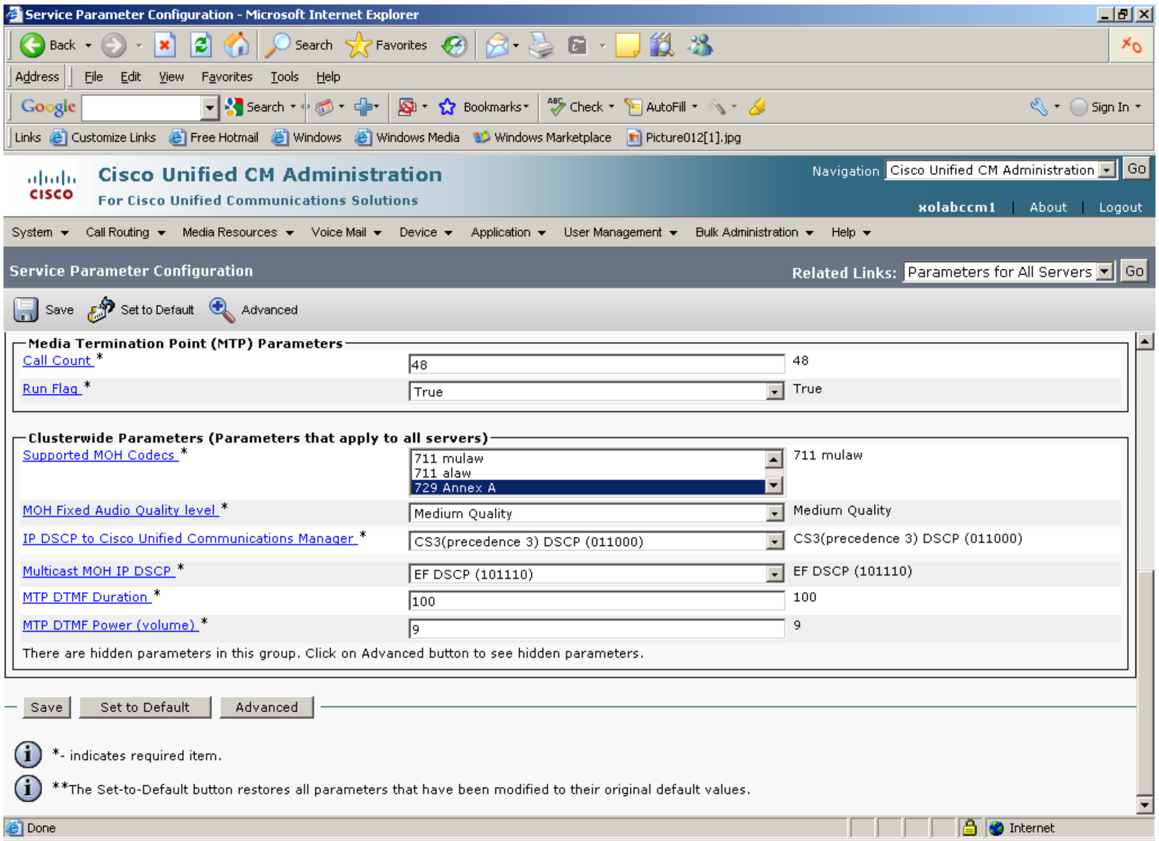


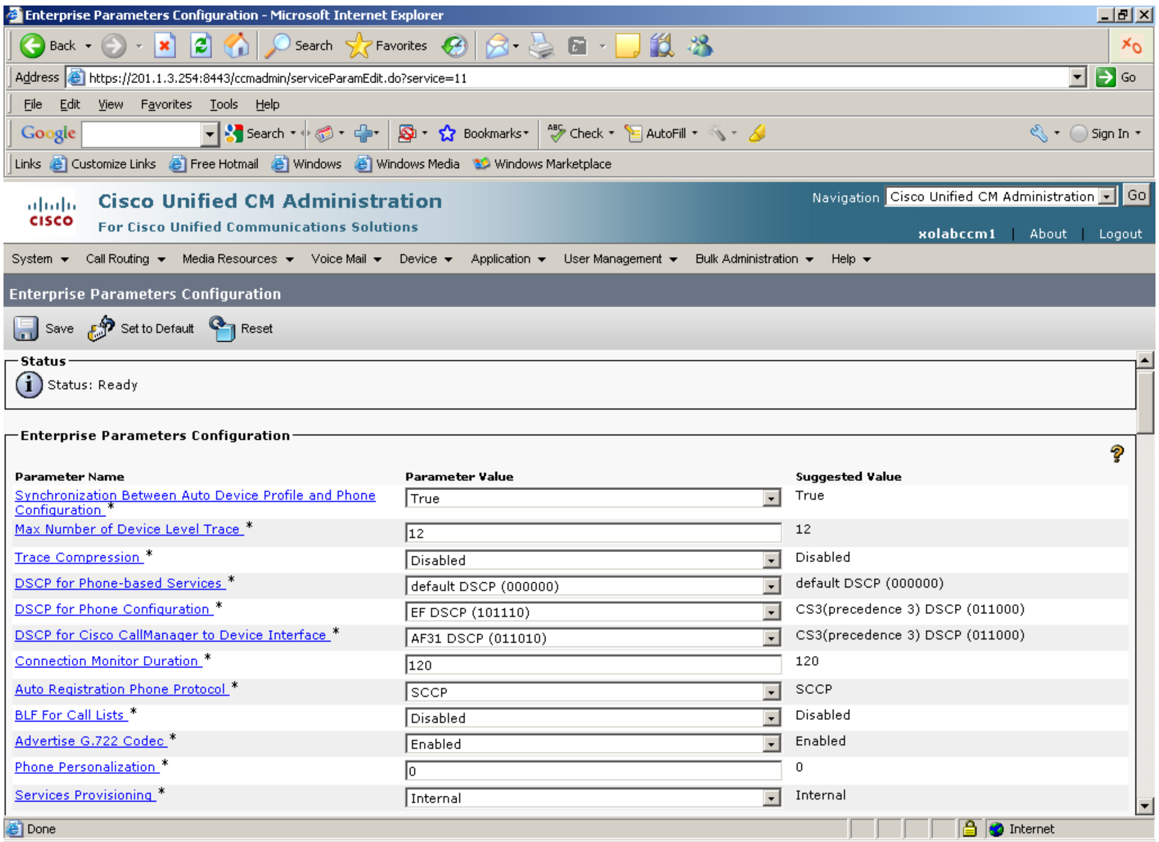
Figure 26, Music on Hold Server Codec Screen Capture Part 2



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 Cisco CallManager 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.

Figure 27, Enterprise Parameters: DSCP Bit Settings for Signaling and RTP



4. Other CUCM configurations

Other CUCM Configurations

Information on more CUCM configuration options is available on Cisco.com at http://www.cisco.com/en/US/products/sw/voicesw/index.html.