

Cisco Call Manager 6.1 Configuration Guide

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1.0 Document Overview

Introduction

XO SIP offers communications, local and long distance voice, Internet access and web hosting via a native IP handoff to a customer's IP PBX. XO has engaged Cisco to help qualify the interoperability of Cisco's Unified Communications Manager (CUCM) with XO SIP. To accomplish that, XO engineers brought CUCM into their lab and performed the configuration tasks necessary to connect to the XO SIP service and to validate key customer requirements. This document presents the CUCM configuration that is required to connect to XO SIP service successfully and documents any caveats to CUCM capabilities with XO's offerings.

Note that this document presents the first phase of XO's interoperability testing with CUCM. The configuration described in this first version of documentation does not address using a companion product in Cisco's Unified Communications offerings, the Cisco Unified Border Element, or "CUBE". CUBE is a Session Border Controller (SBC) implementation on Cisco access router platforms including the Integrated Service Routers (ISR) 2800 and 3800 series devices. It is a component that is recommended by Cisco when CUCM SIP trunking is configured, but that is optional, depending on the features that are configured or expected from CUCM. This document will refer to CUBE occasionally, and CUBE is scheduled for future testing phases with XO's service.

Scope

This document is intended for systems integrators responsible for configuring and deploying Cisco Unified Communications Manager (CUCM) for XO SIP customers. It is also intended for use by XO and Cisco personnel for troubleshooting issues associated with the integration of Cisco CUCM and XO SIP.

CUCM configuration is flexible, with many options. XO's testing and this document are concerned only with the essential configuration necessary for interoperability testing with XO's SIP service. Additional configuration tasks are documented on Cisco's web site Cisco.com at <http://www.cisco.com/en/US/products/sw/voicesw/ps556/index.html>.

Known Issues

While XO certifies interoperability between XO SIP service and the 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:

****67 for Outgoing Calling Line ID Delivery Blocking Per Call is not supported**

Caller ID Failure on Calls Forwarded Off-Net –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.

The following workarounds are suggested; however the limitations of each should be taken into consideration prior to implementation.

- **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. In the example shown in Figure 3, “469387” 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 customer.

Limitation of 4-digit workaround for calls made 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.

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 “408551212,” 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.

**Registration
Method**

CUCM utilizes static registration between IP phones and the IP PBX.

**XO SIP
Service
Packages
Supported**

According to Cisco, CUCM 6.1 implemented without CUBE supports G.711 code only, therefore it supports XO SIP Service Package 1 only:

Pkg	Codec	DTMF	Fax
1	G.711	RFC2833 (in-band RTP DTMF failback)	T.38; G.711 passthrough

XO SIP package 2 (G.729a compression) is not supported

2.0 Cisco Unified Communications Manager for XO SIP Overview

Product Description Cisco Unified Communications Manager (CUCM, formerly Cisco Unified Call Manager) is a powerful call-processing component of the Cisco Unified Communications Solution. It enhances business productivity and facilitates agility by creating a unified workspace encompassing every combination of applications, devices, networks, and operating systems for up to 30,000 users.

Cisco Unified Communications Manager is a scalable, distributable, and highly available enterprise-class IP telephony call-processing system that delivers voice, video, mobility, and presence services to IP phones, media processing devices, VoIP gateways, mobile devices, and multimedia applications.

Since version 4.0, CUCM has supported SIP trunking as a method for IP peering, an alternative to tradition TDM connectivity through gateways. Providers such as XO have begun offering PSTN connectivity to Enterprise customers over SIP trunks. XO's SIP product is one such service.

CUCM XO SIP Qualification Engineers from XO and Cisco worked together to qualify interoperability between CUCM and XO SIP. This included documenting the lab topology and the CUCM configuration supporting the topology. Additional elements in the testing were IP phones, an IP router, and a Cisco Unity Connection voicemail system, which were including for the sake of completeness. The intent of the qualification process was to ensure both that CUCM would function correctly in a XO SIP environment and that XO will be able to support the platform if any issues arise after a customer installation, by referring back to a working configuration.

XO's demarcation point with a network customer is at the managed Cisco router that XO deploys, which provides network access services for IP voice and data traffic. Any equipment on the customer premise, including PBXs supported by the XO SIP service, are the responsibility of XO's customer and any supporting Value-Added Resellers (VARs). With that in mind, it should be understood that the main focus in developing CUCM configuration templates was to validate a configuration that works and that enables communication between the CPE-based equipment and XO's SIP call agent. Basic voice features including on-net and off-net calling, call transfers and forwarding, voicemail, and any other network-based voice services upon which a PBX system depends, fell into the testing and qualification effort by XO and Cisco. More customer-specific features such as hunt-group definitions, paging groups, and the like, have been tested by Cisco but are left to VARs and end-customers to configure.

The LAN topologies into which CUCM and supporting services can be deployed are more than can be addressed in this document, and the configuration options that are used depend entirely upon customer requirements. The intent of this testing effort and documentation was simply to establish a working SIP trunk configuration that supports basic calls and that verifies DTMF-relay functionality over this trunk.

**Supported
Line-side
Protocols**

Although CUCM supports SCCP (Skinny) and SIP on the line side to IP phones, this document focuses on configurations tested with SCCP phones only. Future CUCM testing efforts that address SIP lines side support will be developed and documented in revisions to this guide. In this configuration, CUCM acts as a protocol converter and SIP user-agent between the SIP trunk to XO's call agent and Cisco IP phones running SCCP ("Skinny") images.

**Lab LAN
Topology
for a CUCM
Testing**

In this test bed the dial-plan, IP phones and Unity Connection server configurations were minimal and established just to verify call flows and DTMF-relay. A SIP trunk was employed between the CUCM and the Unity servers.

In deference to a common customer practice, 4-digit extensions were used to permit simple dialing within the Enterprise being simulated in the lab.

Full E.164 DIDs were obtained from the switch and calls were placed to and from the real PSTN.

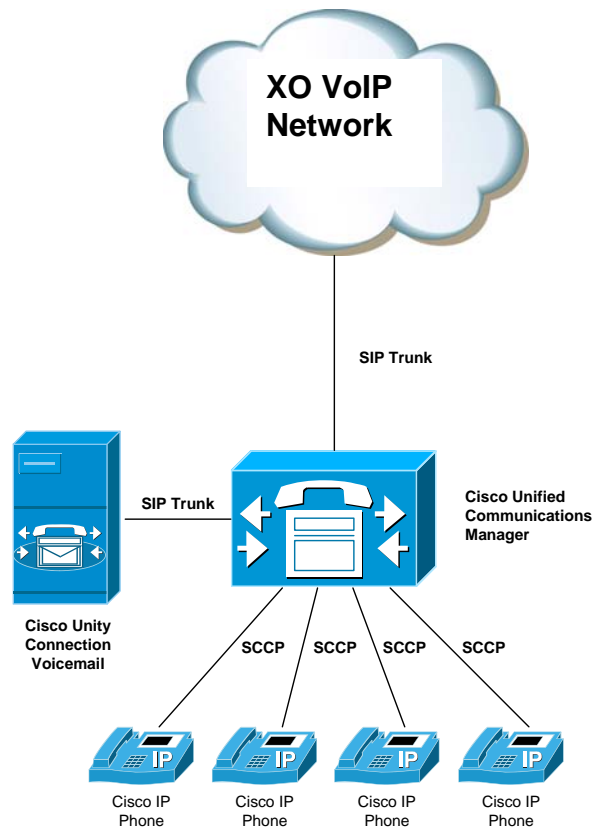


Figure 1, Lab Test LAN Topology

**Requirements
and What was
Tested**

Hardware requirements for Cisco Unified Communications Manager and its supporting servers, like Unity Connection, may be found on Cisco.com at <http://www.cisco.com/en/US/products/sw/voicesw/ps556/index.html>. The specific servers that are chosen for production depend entirely upon customer sizing and feature requirements. In general, Cisco MCS 7800 series servers or approved 3rd party servers from HP and IBM may be used for CUCM and other Cisco Unified Communications platforms.

XO CUCM testing was conducted on a single MCS 7825 server.

This testing effort validated the following software versions:

Component	Supported
CUCM	6.1(2) – Build 6.1.2.-1000-13
Unity Connection ¹	7.0.1.11,000-2

¹ A basic configuration was used on Unity Connection, mainly to validate DTMF-relay after a call was forwarded to a mailbox.

**CUCM
Configuration**

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 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, which may be found 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

The following screenshots detail the SIP trunk configuration for the test lab. Multiple images are shown because the page for the SIP trunk configuration is lengthy.

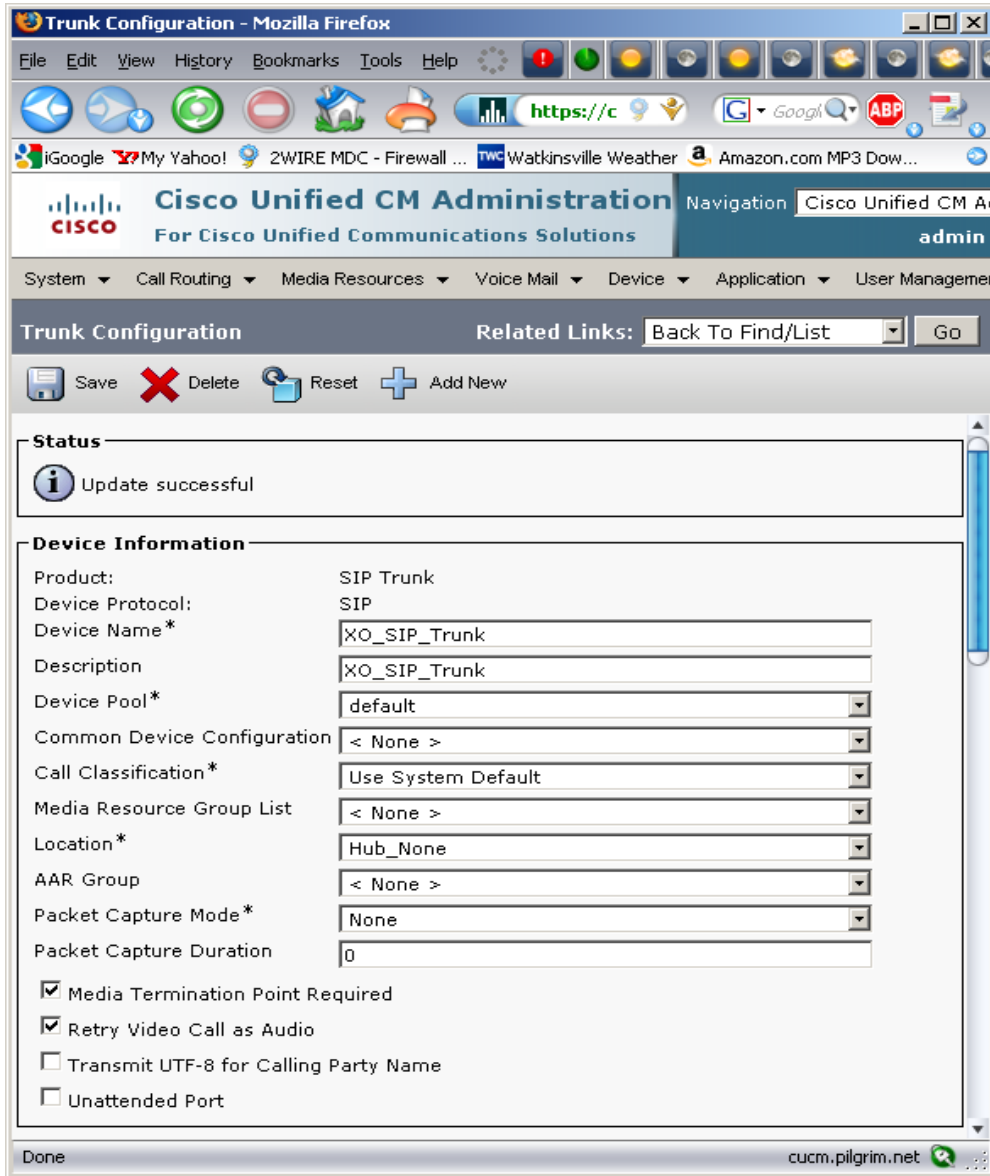


Figure 2 SIP Trunk Device Information

For the most part 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. For XO's testing, software MTPs were used. If MTP is not checked, call hold and music on hold will not work.

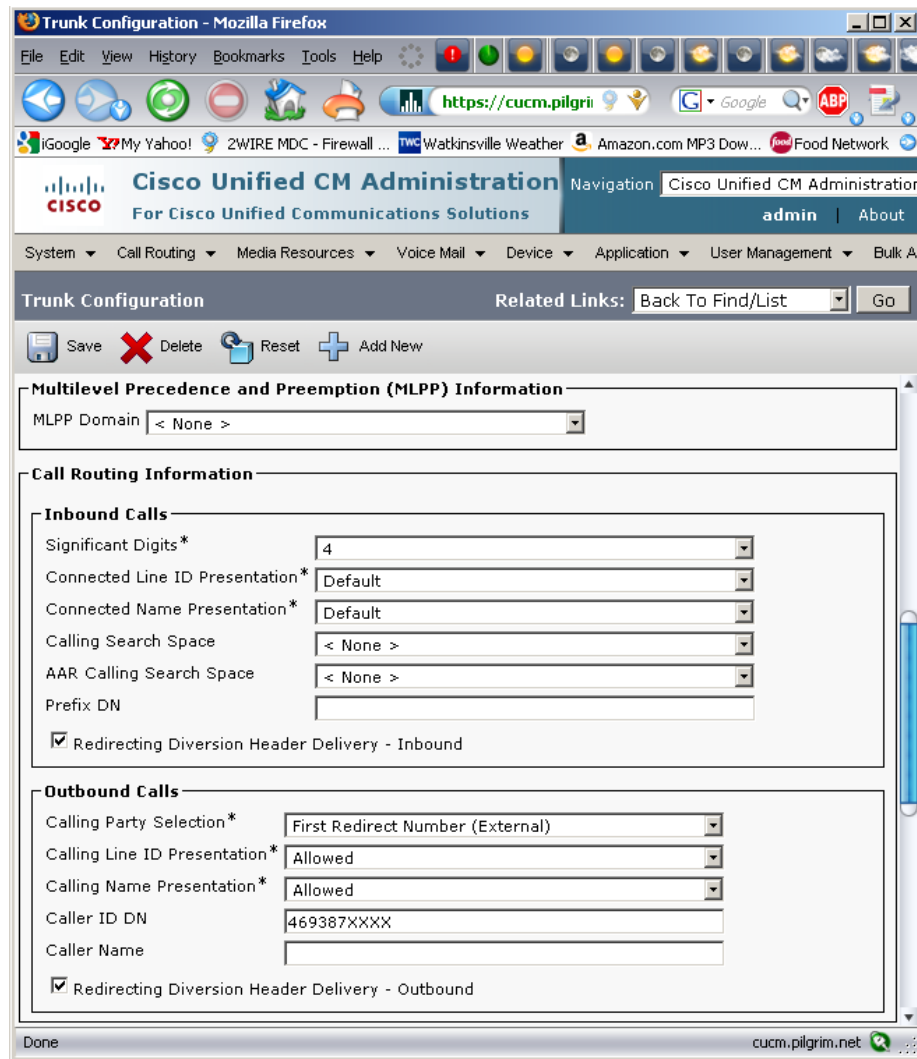


Figure 3 SIP Trunk Inbound and Outbound Call Options

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 1.3 for more details.

The final screenshot shows that the destination endpoint of the CUCM SIP trunk is to the IP address of XO's SBC, to port 5060. Default settings were used for non-secure SIP trunking, and RFC2833 DTMF-relay was specified:

- 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.

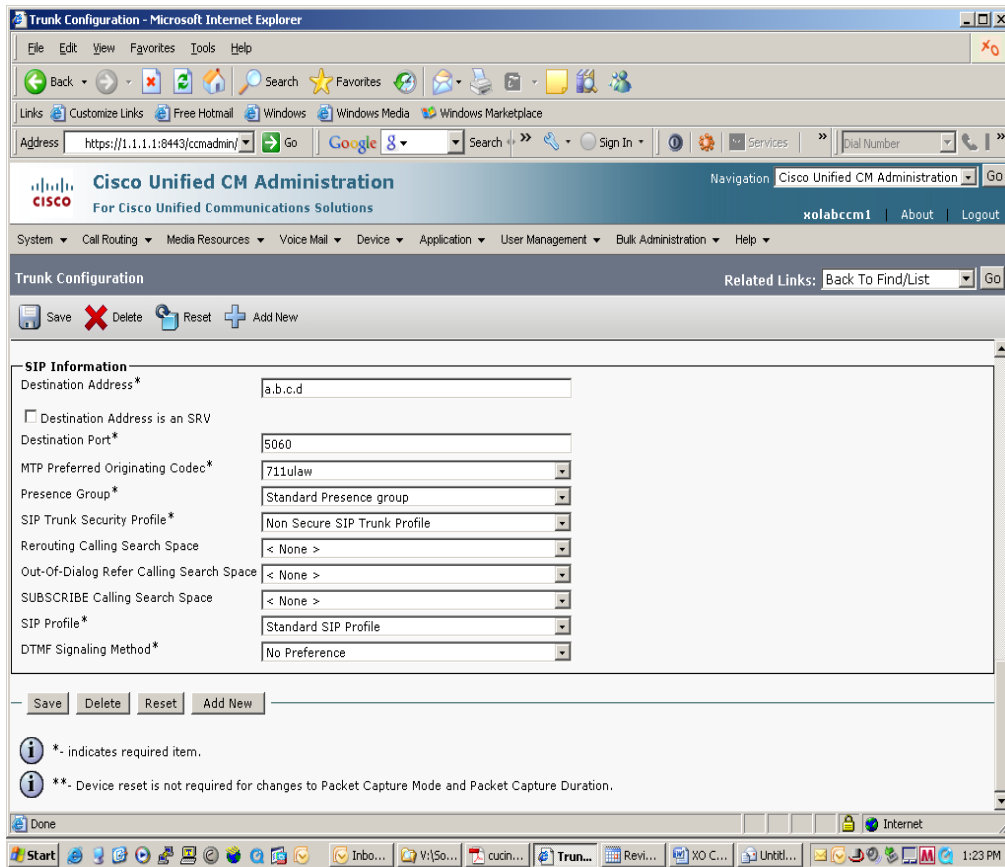


Figure 4 SIP Trunk SIP Information

**Other CUCM
Configuration
Options**

Given the wide variety of the possible CUCM configurations available to XO SIP customers, Cisco and XO customers are encouraged to explore the CUCM options available on Cisco.com at <http://www.cisco.com/en/US/products/sw/voicesw/index.html>.