Cisco Unified Border Element for Contact Center Solutions

First Published: December 17, 2007, OL-14676-02
Last Updated: April 21, 2010

Contents

Americas Headquarters:
Cisco Systems, Inc., 170 West Tasman Drive, San Jose, CA 95134-1706 USA

Introduction

This document provides a sample configuration for contact center solutions using Cisco Unified Border Element (formerly Cisco Multiservice IP-to-IP Gateway software). Cisco Unified Border Element facilitates connectivity among independent Cisco Unified Communications, VoIP, and video networks. Cisco Unified Border Element also provides session border controller (SBC) functions, such as services demarcation, call admission control (CAC), protocol translation, and security.

This document describes the components used in the network, and provides sample device configurations that have been tested for the described features. Use this document for deploying a Cisco Unified Border Element contact center solution.

Prerequisites

Requirements

Ensure that you are using the Cisco IOS Release 15.1(1)T software before you attempt this configuration. The contact center solution using Cisco Unified Border Element described in this document was tested using the Cisco IOS Release 15.1(1)T software. See the "Caveats" section on page 8 for more information.

Components Used

The information in this guide is based on the software and hardware versions identified and described in the following sections:

The information in this document was created through the use of the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, make sure that you understand the potential impact of any command.

Cisco IOS VoiceXML Browser

The Cisco Integrated Services Router G2 (3945 and 3945E) are used in this solution for Cisco IOS VoiceXML browser services.

Cisco Unified SIP Proxy

Cisco Unified Session Initiation Protocol (SIP) Proxy was used in this solution as an alternative to Cisco Unified Presence Server. Cisco Unified SIP Proxy runs on a network module that deploys on the Cisco 3800 Series ISR platform. Cisco Unified SIP Proxy eliminates the need for a standalone SIP proxy server. It is highly configurable, has flexible routing and normalization policies, is highly scalable, and also supports features like percentage routing and time-of-day routing. This solution uses Cisco Unified SIP Proxy version 1.1.4. The Cisco 3845 Series ISR router hosting Cisco Unified SIP Proxy was running Cisco IOS Release 12.4(22)T.

Cisco Unified Communications Manager

Cisco Unified Communications Manager is an integral part of IP-based contact centers where agents use Cisco Unified IP Phones. Cisco Unified Communications Manager is not required if the contact center is time-division multiplexing (TDM) based. The solution discussed in this document uses SIP trunks and Cisco Unified Communications Manager version 7.1(3b).

Cisco Unified Border Element

The Cisco Unified Border Element serves as a feature-rich demarcation point for connecting enterprises to service providers over Unified Communications trunks, including SIP trunks. Cisco Unified Border Element is fully interoperable with the Cisco Unified Communications Manager and the Cisco Unified Communications Manager Express. When interoperating with the Cisco Unified Customer Voice Portal, the Cisco Unified Border Element plays the principal role of receiving calls through provisioned SIP trunks and routing the calls to agent phones. For interactive voice response (IVR) treatment, the Cisco Unified Border Element can also provide a Voice Extensible Markup Language (VoiceXML) gateway. The Cisco Unified Communications Manager-originated calls are routed through Cisco Unified Border Element in SIP-to-SIP deployments. The Cisco IOS Release 15.1(1)T software was used on Cisco Unified Border Element. More features are supported by later Cisco IOS Releases.

Cisco Unified Customer Voice Portal Call Server

The Cisco Unified Customer Voice Portal Call Server uses a SIP back-to-back user agent (B2BUA) and IVR service. It is located logically between the Cisco IOS gateway and Cisco Unified Intelligent Contact Management Enterprise Voice Response Unit (VRU) Peripheral Gateway (PG). The Cisco Unified Customer Voice Portal Call Server is not a required component for Standalone deployment models where call routing is not a desired. Cisco Unified CVP Server 7.0(2) was used.

Cisco Unified Intelligent Contact Management Enterprise

Cisco Unified Intelligent Contact Management Enterprise is a mandatory component for advanced call control mechanism, such as, IP switching and transfer to agents. Cisco Unified Intelligent Contact Management Enterprise provides call center agent management capabilities and call switching capabilities. This solution uses Cisco Unified Intelligent Contact Management Enterprise version 7.5(1).

Cisco Unified Customer Voice Portal VoiceXML Server

The Cisco Unified Customer Voice Portal VoiceXML Server (VXML Server) delivers external VoiceXML documents to the Cisco IOS gateway. Voice applications are written using the VXML Server Studio and then deployed to the VXML Server.

When calls invoke external VoiceXML, the VXML Server builds VoiceXML pages dynamically according to the contents of the deployed application. The VXML Server interacts only with the Cisco IOS gateway. The gateway requests the VoiceXML document from the VXML Server upon command from the Cisco Unified Customer Voice Portal Call Server.

This solution uses the VXML Server version 7.0(2).

Nuance Automatic Speech Recognition/Text-To-Speech Server

The Nuance Automatic Speech Recognition/Text-To-Speech (ASR)/(TTS) Server provides speech recognition services and text-to-speech services for the Cisco IOS VoiceXML gateway. Communication between the ASR/TTS server(s) and the Cisco IOS VoiceXML gateway uses Media Resource Control Protocol (MRCP). The Cisco Unified Border Element–Cisco Unified Customer Voice Portal interoperability scenarios used MRCP version 1 and the following Nuance versions:

- Nuance Recognizer 9.0(7)
- Nuance RealSpeak 4.5
- Nuance SpeechServer 5.0(6)

Conventions

Refer to *Cisco Technical Tips Conventions* for information on document conventions.

Background Information

This document describes two models of contact center solutions that include Cisco Unified Border Element: centralized and distributed. In the centralized model, VoiceXML runs on a separate Cisco Integrated Services Router G2 (3945, 3945E) independent of Cisco Unified Border Element. In the distributed model, VoiceXML runs on the same Cisco IOS gateway on which Cisco Unified Border Element runs.

To configure a contact center solution with Cisco Unified Border Element, you should understand the following concepts:

- Call Flows, page 5
- Feature Summary, page 6
- Caveats, page 8

Call Flows

This section describes the call flow for calls originated by Cisco Unified Communications Manager.

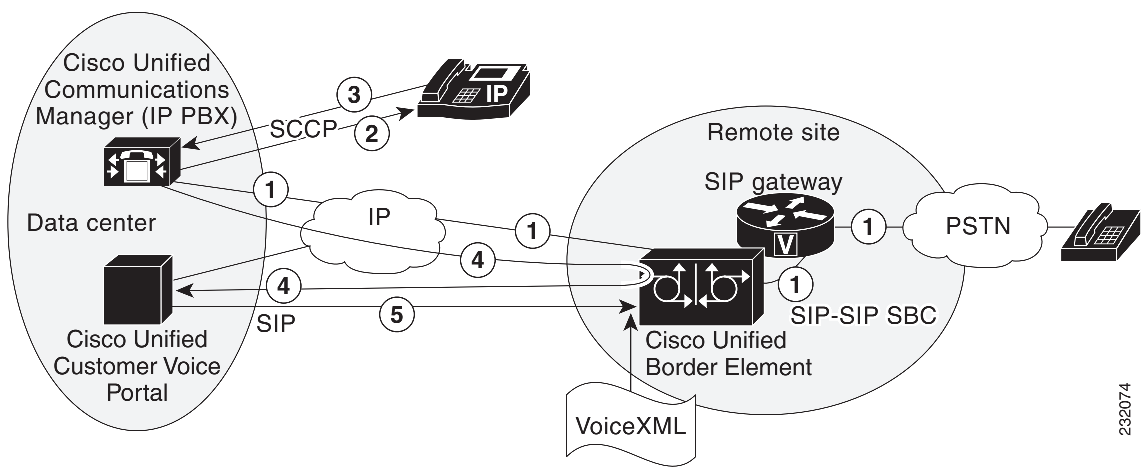
- Cisco Unified Communications Manager–Originated Calls, page 5

Cisco Unified Communications Manager–Originated Calls

Figure 1 shows the call flow for calls originated by Cisco Unified Communications Manager:

1. The incoming call enters Cisco Unified Customer Voice Portal through Cisco Unified Border Element.
2. The call is directly routed to the agent phone via Cisco Unified Communications Manager.
3. The call is transferred by means of Skinny Client Control Protocol (SCCP) from the agent to Cisco Unified Communications Manager. In one possible scenario, a warm transfer is executed, that is, a second agent is not available, so the first agent needs to be queued.
4. Cisco Unified Communications Manager originates a SIP call to the Cisco Unified Border Element. Cisco Unified Border Element originates a SIP call to the Cisco Unified Customer Voice Portal.
5. The call is being anchored at the Cisco Unified Border Element. Cisco Unified Customer Voice Portal invokes the VoiceXML browser on the Cisco Unified Border Element, which is the originating gateway.

Figure 1 Cisco Unified Communications Manager–Originated Calls: SIP-to-SIP



Note these points about the call flow in Figure 1:

- When a PSTN call is transferred, sometimes referred to as a *warm transfer*, to Cisco Unified Customer Voice Portal by an IP Phone, the originating gateway information is lost.
- Cisco Unified Customer Voice Portal cannot queue the call at the correct gateway for VoiceXML control.
- Cisco Unified Border Element provides an anchor point for the call at the originating gateway.
- Because of the IP Phone transfer, Cisco Unified Communications Manager establishes the SIP call to Cisco Unified Customer Voice Portal through the Cisco Unified Border Element.
  - Cisco Unified Border Element and the SIP gateway are co-resident.
  - Cisco Unified Border Element anchors the SIP Cisco Unified Communications Manager–to–Cisco Unified Customer Voice Portal call so that Cisco Unified Customer Voice Portal sees the Cisco IOS gateway IP address and not the Cisco Unified Communications Manager IP address.
  - Cisco Unified Customer Voice Portal is centralized and cannot pull the call back to the correct gateway because of a routing problem in defining the originating gateway versus the closest gateway.

Feature Summary

The features described in this section were tested as part of the solution configuration.

- Cisco IOS VoiceXML Browser, page 6
- DTMF Interworking, page 7
- Transcoding, page 7
- GTD Pass-Through, page 7
- HTTPS, page 8
- Standalone Model, page 8

Cisco IOS VoiceXML Browser

Applications written in VoiceXML provide access through a voice browser to content and services over the telephone, just as HTML provides access through a web browser running on a PC. The universal accessibility of the telephone and its ease of use make VoiceXML applications a powerful alternative to HTML for accessing the information and services of the World Wide Web.

The Cisco IOS VoiceXML feature provides a platform for interpreting VoiceXML documents. When a telephone call is made to a Cisco IOS VoiceXML–enabled gateway, VoiceXML documents are downloaded from the Cisco Unified Customer Voice Portal servers, providing content and services to the caller, typically in the form of prerecorded audio in an IVR application. You can access online business applications over the telephone that provide, for example, stock quotes, sports scores, or bank account balances.

VoiceXML brings the advantages of web-based development and content delivery to voice applications. VoiceXML is similar to HTML in its simplicity and in its presentation of information. The Cisco IOS VoiceXML feature is based on the VoiceXML 2.1 W3C Candidate Recommendation (June 13, 2005) and is designed to provide web developers great flexibility and ease in implementing VoiceXML applications.

The Cisco IOS VoiceXML browser is central to this solution. Depending on the model being used, the VoiceXML content is interpreted by the Cisco IOS VoiceXML browser on the Cisco Unified Border Element, in the distributed model, or on a separate Cisco IOS VoiceXML gateway, in the centralized model. Both models were tested.

DTMF Interworking

Dual tone multifrequency (DTMF) is important in the contact center because many interactions require the caller to enter DTMF input using the phone keypad. DTMF can be carried out-of-band in the signaling path or in-band in the bearer path. Various DTMF methods exist, and it is important for any solution to interwork reliably among these methods.

The following DTMF relay methods were tested:

- In-band voice to RFC 2833—Requires transcoding resources on the Cisco Unified Border Element.
- RFC 2833 to RFC 2833—Does not require transcoding resources on the Cisco Unified Border Element.

Transcoding

The transcoding feature enables the Cisco Unified Border Element to link two networks using dissimilar codecs. Transcoding may be required to conserve bandwidth across a WAN link. For example, G.711 u-law packets consume more bandwidth than G.729r8 packets, and you may need to perform this conversion before traversing a bandwidth-limited WAN link. Transcoding may be required also in cases in which certain applications support only a certain codec type.

In the contact center space, the agents are typically G.711 u-law or G.729r8 based. However, calls that are inbound via the SIP trunk may be using a different codec. The G.711 u-law, G.729r8, and internet Low Bitrate Codec (iLBC) are the common types. Calls using different codecs than the agents use would necessitate transcoding on the Cisco Unified Border Element.

Transcoding of G.711 u-law to G.729r8 codecs has been tested and is supported. Transcoding of inbound iLBC calls over SIP Trunk to G711 and G729 calls to the agent is supported and has been tested.

GTD Pass-Through

Generic Transparency Descriptor (GTD) is a mechanism for representing telephony signaling messages and parameters in a generic fashion, in particular ISDN User Part (ISUP) messages and parameters. Mapping of every individual ISUP message and parameter to an equivalent SIP message is not viable. The GTD offers a solution in which the ISUP messages and parameters can be represented in a generic fashion and transported by the underlying call signaling messages to each node transited by the call. GTD is supported only across SIP networks that support Multipurpose Internet Mail Extension (MIME) header encapsulation.

In the contact center space, the ISDN user-to-user information (UUI) element has special significance and needs to be passed to the Intelligent Contact Manager (ICM) Enterprise script. The UUI data is passed via the Cisco IOS gateway in the SIP INVITE when signaling forwarding is configured. The INVITE contains the extra section of GTD ISDN variables, and the Cisco Unified Customer Voice Portal SIP subsystem parses the inbound body to send these across the GED-125 API in the NEW CALL message in the “UserToUser” field.

The testing described in this guide focused only on ensuring that the GTD information received on the inbound leg on the Cisco Unified Border Element was passed to the outbound leg via the Session Description Protocol (SDP) of the outbound INVITE. No attempt was made to verify that the UUI data actually reached the ICM script, and no attempt was made to determine how it was used by the script.

HTTPS

With the Secure HTTP (HTTPS) feature, you can use a secure socket interface to load VoiceXML application documents and scripts and play audio prompt files from the HTTP server. In data exchanges between the HTTP server and the client, data is encrypted by one party before it is sent over the network and then decrypted by the other party when it is received. The encryption protection makes eavesdropping much more difficult to decipher, if not impossible.

HTTPS does affect the gateway performance because of the additional overhead involved in processing secure transactions.

HTTPS was tested for two standard models:

- Secure connections between the Cisco IOS gateway and the Cisco Unified Customer Voice Portal VoiceXML Server (standalone model)
- Secure connections between the Cisco IOS gateway and the Cisco Unified Customer Voice Portal Call Server (comprehensive model)

Both the Cisco Unified Customer Voice Portal VoiceXML Server and the Cisco Unified Customer Voice Portal Call Server support HTTPS connections.

Standalone Model

The Cisco Unified Customer Voice Portal standalone deployment model provides IVR services only. Intelligent contact center integration is not included, although it is possible to transfer the call to an arbitrary destination, without call context and without queuing. The standalone model requires Cisco Unified Customer Voice Portal VoiceXML servers and Cisco IOS gateways, but does not include the Cisco Unified Customer Voice Portal Call Server or any Cisco Unified ICM Enterprise components. IVR applications support either DTMF only or a combination of DTMF and ASR and TTS.

Caveats

The tested configuration has the following caveats:

- Supplementary services, such as Call Hold, Call Resume, and Call Transfer, are not supported in media flow-around mode. Basic calls are supported.
- Midcall insertion and release of transcoder on Cisco Unified Border Element is supported in Cisco IOS Release 15.1(2)T.
- MRCP version 2 was not tested with Cisco Unified ICM Enterprise-based microapps scripts.

Configure

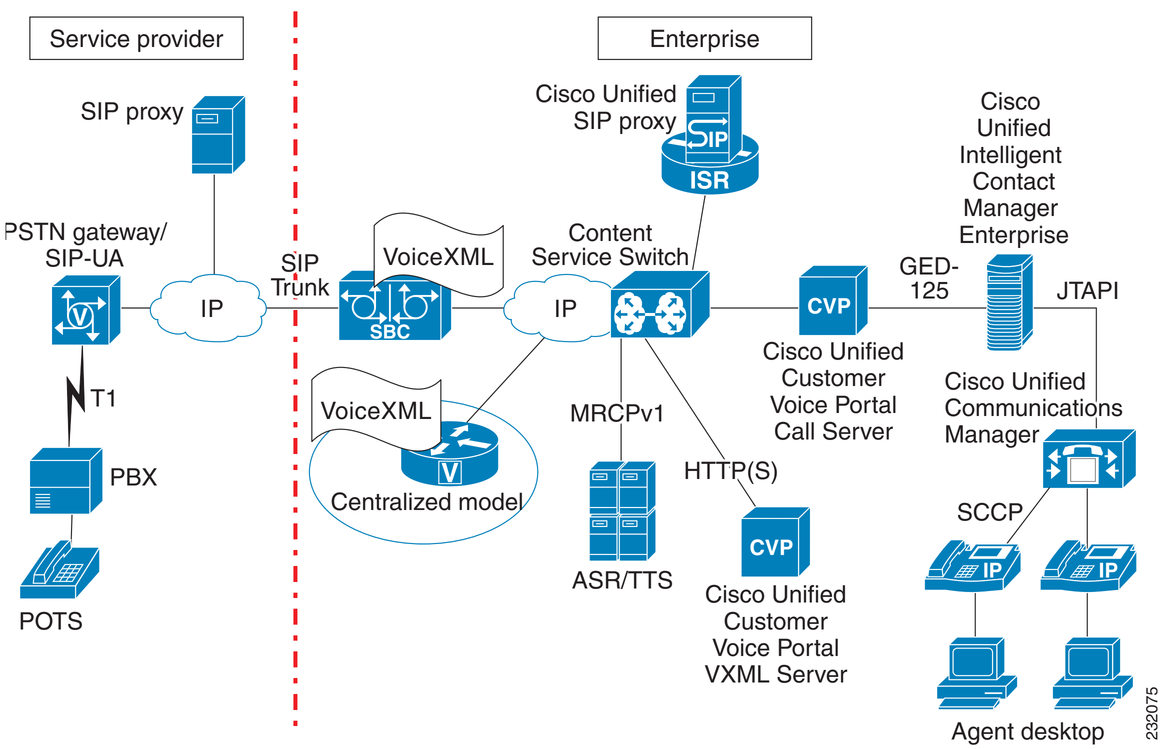
This section provides the information you need to configure the features described in this document.

Note Use *Command Lookup Tool* for more information on the commands used in this guide.

Network Diagram

Figure 2 shows the network setup for the contact center solution with the Cisco Unified Border Element network topology.

Figure 2 Contact Center Solution with Cisco Unified Border Element Network Topology



Configurations

This document uses these configurations:

- Distributed Model, page 10
- Centralized Model, page 19

Distributed Model

The following sections provide configurations of the key devices in the contact center solution using Cisco Unified Border Element distributed model in which VoiceXML runs on the same Cisco IOS gateway as Cisco Unified Border Element.

- Cisco Unified Border Element, page 10
- Cisco Unified Communications Manager Configuration, page 18

Cisco Unified Border Element

The following example shows the configuration of the Cisco Unified Border Element in the distributed model. Significant sections of this output are shown in bold type for emphasis.

```
version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router
!
boot-start-marker
boot-end-marker
!
card type e1 0 0
card type e1 0 1
card type e1 0 2
card type e1 0 3
card type e1 1 0
card type e1 1 1
card type e1 2 0
card type e1 2 1
card type e1 3 0
card type e1 3 1
card type e1 4 0
card type e1 4 1
logging buffered 1000000
no logging console
!
no aaa new-model
!
clock timezone PDT -7
clock summer-time PDT recurring
network-clock-participate slot 1
network-clock-participate slot 2
network-clock-participate slot 3
network-clock-participate slot 4
network-clock-participate wic 0
network-clock-participate wic 1
network-clock-participate wic 2
network-clock-participate wic 3
!
!
crypto pki trustpoint myCA
 enrollment terminal
 revocation-check crl none
!
crypto pki trustpoint myCallServer
 enrollment terminal
 revocation-check crl none
!
```

```
!
crypto pki certificate chain myCA
 certificate ca 264FF556F87B84BD45DE6B44E5C346D7
  3082036A 30820252 A0030201 02021026 4FF556F8 7B84BD45 DE6B44E5 C346D730
    .
    .
    .
  E77E202F E7DCA228 AB3C4C2B 9FBE2315 E3C19030 EED039A3 918ACE7B 3F9B11DF
  0098FC98 AFB8B3AF BFD1B06C CD44B346 CCEF9E57 D6F829B0 2AA91091 6845F3D4
  7697E31D 58F0A313 511483E5 A927
    quit
crypto pki certificate chain myCallServer
 certificate ca 23B9BF01000000000012
  3082043F 30820327 A0030201 02020A23 B9BF0100 00000000 12300D06 092A8648
  86F70D01 01050500 30133111 300F0603 55040313 08506572 664F5344 4D301E17
  0D303831 30303631 37313031 315A170D 30393130 30363137 32303131 5A306631
  4F0583A5 69BA9F19 82ED576D 22E02BFD 0570F3D8 7BB7F784 D0B1A175 62731D11
  6311340D 2B77BEC8 852B14EC 46848B30 9862FA7A C61D2338 117734C0 D24D975D
  AE83901D 8EC7F6E0 0F1F1AE2 4B62EDDA 754B3A39 DCD6B1C2 2E240068 CB62A52E 8846AA
    quit
no ipv6 cef
ip source-route
ip cef
!
!
!
!
no ip domain lookup
ip domain name cisco.com
ip host asr-en-us 10.1.174.51
ip host asr-en-tts 10.1.174.53
ip host tts-en-us 10.1.174.53
!
multilink bundle-name authenticated
!
!
!
!
isdn switch-type primary-5ess
isdn voice-call-failure 0
!
voice-card 0
!
voice-card 1
!
voice-card 2
 dspfarm
 dsp services dspfarm
!
voice-card 3
!
voice-card 4
!
!
!
voice service voip
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
! This command enables the Cisco Unified Border Element
! to proces calls between SIP end points
 allow-connections sip to sip
! This command is required if GTD parameters are to be passed between in and out logs
 signaling forward unconditional
```

```
 h323
  emptycapability
  h245 passthru tcsnonstd-passthru
 sip
  rel1xx disable
  header-passing
  midcall-signaling passthru
!
!
voice class uri PerfTTS1 sip
 pattern PerfTTS1@10.1.174.53
!
voice class uri PerfASR1 sip
 pattern PerfASR1@10.1.174.51
!
!
!
voice translation-rule 71083
 rule 1 /671083/ /71083/
!
!
voice translation-profile CMO
 translate called 71083
!
!
http client cache memory file 10000
ivr asr-server rtsp://10.1.174.51/recognizer
ivr tts-server rtsp://10.1.174.53/synthesizer
!
application
 service new-call flash0:bootstrap.vxml
  paramspace english language en
  paramspace english index 0
  paramspace english location flash0:
  paramspace english prefix en
 !
 service ASRTTS-Transfer flash0:CVPSelfService.tcl
  param CVPPrimaryVXMLServer 10.1.174.58
  paramspace english index 0
  paramspace english language en
  paramspace english location flash0:
  paramspace english prefix en
  param CVPSelfService-app AudAsrTTS-Transfer
  param CVPSelfService-port 7000
 !
 service CVPSelfService flash0:CVPSelfServiceBootstrap.vxml
  paramspace english language en
  paramspace english index 0
  paramspace english location flash0:
  paramspace english prefix en
 !
 service ringtone flash0:ringtone.tcl
  paramspace english language en
  paramspace english index 0
  paramspace english location flash0:
  paramspace english prefix en
 !
 service cvperror flash0:cvperror.tcl
  paramspace english index 0
  paramspace english language en
  paramspace english location flash0:
  paramspace english prefix en
 !
 service bootstrapssl flash0:bootstrap.tcl
```

```
  paramspace english index 0
  paramspace english language en
  param cvpserverssl 1
  paramspace english location flash0:
  paramspace english prefix en
  param cvpserverport 8443
 !
 service handoff flash0:handoff.tcl
  paramspace english language en
  paramspace english index 0
  paramspace english location flash0:
  paramspace english prefix en
 !
 service bootstrap flash0:bootstrap.tcl
  paramspace english index 0
  paramspace english language en
  paramspace english location flash0:
  paramspace english prefix en
  param cvpserverssl 0
 !
 monitor
  interface stats
  interface event-log
  interface event-log aaa
  interface event-log asr
  interface event-log tts
  interface event-log http
  interface event-log tftp
  interface event-log rtsp
  interface event-log ram
  interface event-log flash
  interface event-log smtp
  interface max-server-records 100
  stats
  event-log
  event-log max-buffer-size 40
  history session max-records 500
  history session retain-timer 3660
 !
!
mrcp client rtpsetup enable
vxml version 2.0
license udi pid C3900-SPE150/K9 sn FHH123000HU
hw-module pvdm 0/0
!
hw-module pvdm 0/1
!
hw-module pvdm 0/2
!
hw-module pvdm 0/3
!
!
!
archive
 log config
  hidekeys
!
redundancy
!
!
!
!
controller E1 0/0/0
!
```

```
controller E1 0/0/1
!
controller E1 0/1/0
!
controller E1 0/1/1
!
controller E1 0/2/0
!
controller E1 0/2/1
!
controller E1 0/3/0
!
controller E1 0/3/1
!
controller E1 1/0
!
controller E1 1/1
!
controller E1 1/0/0
!
controller E1 1/0/1
!
controller E1 2/0
!
controller E1 2/1
!
controller E1 2/0/0
!
controller E1 2/0/1
!
controller E1 3/0
!
controller E1 3/1
!
controller E1 3/0/0
!
controller E1 3/0/1
!
!
!
!
!
!
!
!
interface GigabitEthernet0/0
 ip address 10.1.175.6 255.255.0.0
 duplex auto
 speed auto
!
interface GigabitEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
 media-type rj45
!
interface GigabitEthernet0/2
 no ip address
 shutdown
 duplex auto
 speed auto
 media-type rj45
!
```

```
!
ip forward-protocol nd
!
no ip http server
no ip http secure-server
ip http client secure-trustpoint PerfOSDM-CA
!
ip route 1.1.0.0 255.255.0.0 10.1.0.1
ip route 223.255.254.0 255.255.255.0 10.1.0.1
!
!
!
!
!
nls resp-timeout 1
cpd cr-id 1
!
!
control-plane
!
call treatment on
call threshold global cpu-5sec low 100 high 100
call threshold global cpu-avg low 100 high 100
!
!
!
sccp local GigabitEthernet0/0
sccp ccm 10.1.175.6 identifier 1 version 6.0
sccp
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 2 register MTP000bbecadcf2
 associate profile 1 register MTP001b547ef9b0
 keepalive retries 5
 switchover method immediate
 switchback method immediate
!
dspfarm profile 1 transcode
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 codec g729r8
 codec g729br8
 codec ilbc
 maximum sessions 1
 associate application SCCP
 shutdown
!
dspfarm profile 2 transcode universal
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 codec ilbc
 maximum sessions 6
 associate application SCCP
!
dial-peer voice 4 voip
 description Incoming dial-peer for all calls
 incoming called-number 710..
 dtmf-relay rtp-nte
 codec g711ulaw
```

```
 no vad
!
dial-peer voice 5 voip
 description Send Call to Cisco Unified SIP Proxy
 destination-pattern 710..
 session protocol sipv2
 session target ipv4:10.1.174.221
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
dial-peer voice 6 voip
 service bootstrap
 incoming called-number 888999T
! The dial-peer tag "888999" corresponds to the LABEL configured on the ICM. This
! label is known as the "VRU transfer label" and is returned to the CUCVP call server
! (and eventually to the CUBE via the CUPS) by the ICM for VRU treatment. codec g711ulaw
 no vad
!
dial-peer voice 9191 voip
 service ringtone
 incoming called-number 91T
 codec g711ulaw
!
dial-peer voice 9292 voip
 service cvperror
 incoming called-number 92T
 codec g711ulaw
!
dial-peer voice 9 voip
 translation-profile incoming CMO
 incoming called-number 671083
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
dial-peer voice 50 voip
 destination-pattern 333....
 session protocol sipv2
 session target ipv4:10.1.174.125
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
dial-peer voice 40 voip
 incoming called-number 333....
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
dial-peer voice 1800888 voip
 service asrtts-transfer
 incoming called-number 1800888
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
!
gateway
 timer receive-rtp 1200
!
!
!
gatekeeper
```

```
 shutdown
!
!
telephony-service
 sdspfarm units 5
 sdspfarm transcode sessions 128
 sdspfarm tag 1 MTP001b547ef9b0
 sdspfarm tag 2 MTP000bbecadcf2
 max-ephones 6
 max-dn 1
 ip source-address 10.1.175.6 port 2000
 max-conferences 4 gain -6
 call-forward system redirecting-expanded
 transfer-system full-consult
 create cnf-files version-stamp Jan 01 2002 00:00:00
!
alias exec log show logg
alias exec rtp show voip rtp connections
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
 exec-timeout 0 0
 password lab
 login
 transport input all
!
scheduler allocate 20000 1000
end
```

Cisco Unified Communications Manager Configuration

On Cisco Unified Communications Manager, configure the SIP trunks to the Cisco Unified SIP ProxyServer, as shown in Figure 3. The agent phones should be associated with the appropriate configured application user.

Figure 3 Cisco Unified Communications Manager Express Configuration

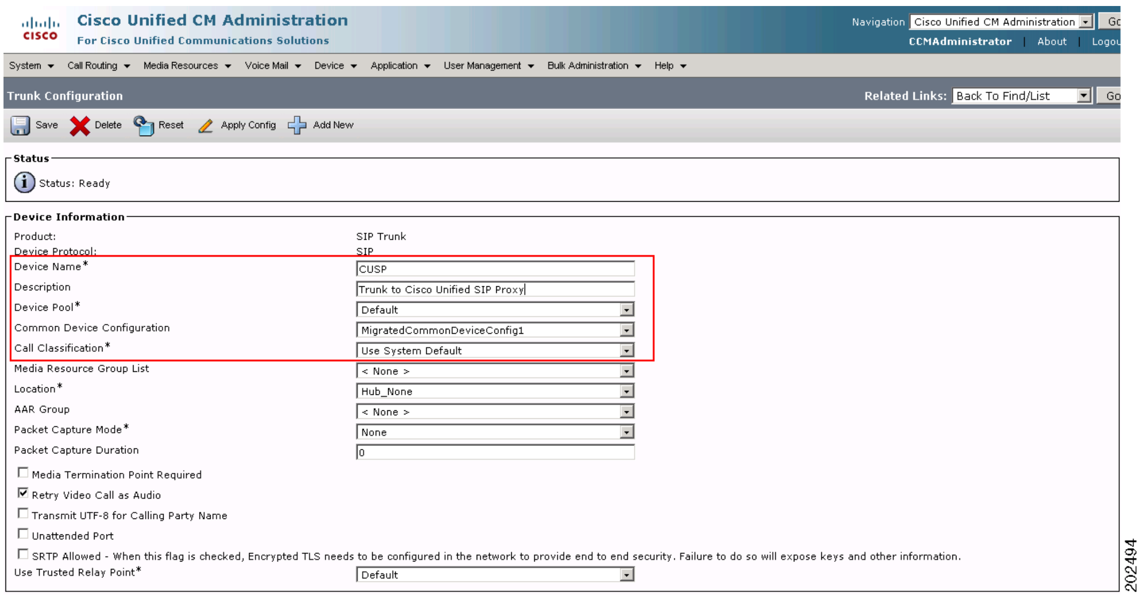
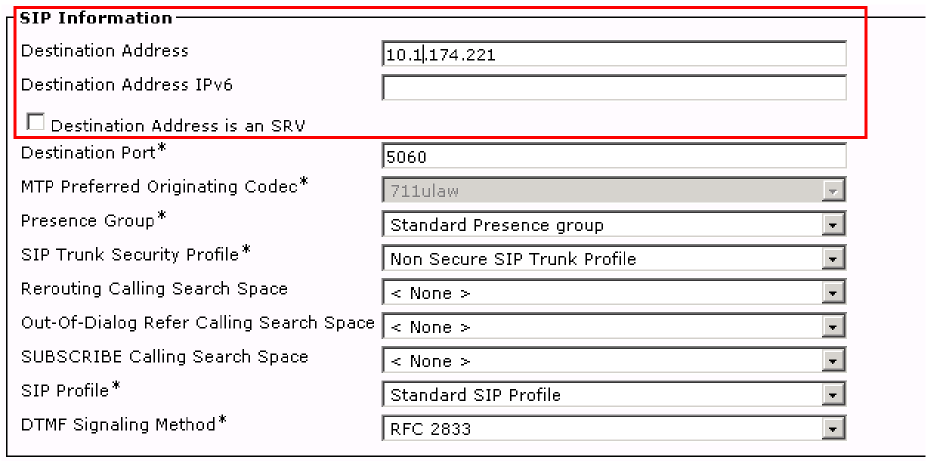


Figure 4 shows the SIP information configuration on Cisco Unified Communications Manager.

Figure 4 Cisco Unified Communications Manager Express Configuration



Centralized Model

The following are the configurations of the key devices in the contact center solution using the Cisco Unified Border Element centralized model in which VoiceXML runs on a separate Cisco IOS gateway that is independent of Cisco Unified Border Element.

- Cisco Unified Border Element, page 19
- Static Routes on Cisco Unified SIP Proxy, page 25
- Cisco Unified Communications Manager Configuration, page 28

Cisco Unified Border Element

The following example shows the configuration of the Cisco Unified Border Element in the centralized model. Significant sections of this output are shown in bold type for emphasis.

```
version 15.1
service timestamps debug datetime msec localtime show-timezone
service timestamps log datetime msec localtime show-timezone
no service password-encryption
!
hostname VoiceXMLGateway
!
boot-start-marker
boot-end-marker
!
card type e1 0 0
card type e1 0 1
card type e1 0 2
card type e1 0 3
card type e1 1 0
card type e1 1 1
card type e1 2 0
card type e1 2 1
card type e1 3 0
card type e1 3 1
card type e1 4 0
```

```
card type e1 4 1
logging buffered 500000
no logging console
enable password lab
!
no aaa new-model
!
clock timezone PST -8
clock summer-time PDT recurring
network-clock-participate slot 1
network-clock-participate slot 2
network-clock-participate slot 3
network-clock-participate slot 4
network-clock-participate wic 0
network-clock-participate wic 1
network-clock-participate wic 2
network-clock-participate wic 3
!
!
crypto pki trustpoint CallServer
 enrollment terminal
 revocation-check none
!
crypto pki trustpoint MediaServer
 enrollment terminal
 revocation-check none
!
!
crypto pki certificate chain CallServer
 certificate ca 272DAA83000000000017
  3082041E 30820306 A0030201 02020A27 2DAA8300 00000000 17300D06 092A8648
  86F70D01 01050500 30133111 300F0603 55040313 08506572 664F5344 4D301E17
  0D313030 33333131 37353230 375A170D 31313033 33313138 30323037 5A304531
<snip>
  083FAE6A C12647B2 82CB7DFD 600EDFE0 42F29D45 E44E9EAE E546A381 154A7879
  C904B986 E93D876A C4A139CD F1ED6EB4 F0609941 89844C29 796BACDF 515DAEDA
  D91BA2AC 85C81104 BA6D48B2 B05D6551 A7AA0FAC 131BF8D2 7B2B97AD 7A07D2E8 40AF
    quit
crypto pki certificate chain MediaServer
 certificate ca 272882B0000000000016
  3082043F 30820327 A0030201 02020A27 2882B000 00000000 16300D06 092A8648
  86F70D01 01050500 30133111 300F0603 55040313 08506572 664F5344 4D301E17
  0D313030 33333131 37343632 395A170D 31313033 33313137 35363239 5A306631
<snip>
  69E5690F 654B692E 6D31943C 9DE87E14 C57CC72E 91A1A309 DA131201 CE60DF42
  CAC778BD D2EA7AE9 9E83933B 403ECBFD 77767E67 A8CC7F32 DA4A41E1 06B4011B
  D88553D0 54EB7349 FC33CA2C 272283A5 16BE4FE7 61355383 E05AB0FC 008787D7 780063
    quit
no ipv6 cef
ip source-route
ip cef
!
!
!
!
no ip domain lookup
ip domain name cisco.com
ip host asr-en-us 10.1.174.51
ip host asr-en-tts 10.1.174.53
ip host tts-en-us 10.1.174.53
!
multilink bundle-name authenticated
!
!
```

```
!
!
isdn switch-type primary-5ess
isdn voice-call-failure 0
!
voice-card 0
 dspfarm
 dsp services dspfarm
!
voice-card 1
 dspfarm
 dsp services dspfarm
!
voice-card 2
 dspfarm
 dsp services dspfarm
!
voice-card 3
 dspfarm
 dsp services dspfarm
!
voice-card 4
 dspfarm
 dsp services dspfarm
!
!
!
voice service voip
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 signaling forward unconditional
 sip
  rel1xx disable
  header-passing
!
!
voice class uri PerfTTS1 sip
 pattern PerfTTS1@10.1.174.53
!
voice class uri PerfASR1 sip
 pattern PerfASR1@10.1.174.51
!
!
!
!
http client cache memory file 10000
http client secure-trustpoint CallServer
ivr asr-server rtsp://10.1.174.51/recognizer
ivr tts-server rtsp://10.1.174.53/synthesizer
!
application
 service new-call flash0:bootstrap.vxml
  paramspace english language en
  paramspace english index 0
  paramspace english location flash0:
  paramspace english prefix en
 !
 service ASRTTS-Transfer flash0:CVPSelfService.tcl
  param CVPPrimaryVXMLServer 10.1.174.58
  paramspace english index 0
  paramspace english language en
  paramspace english location flash0:
```

```
 paramspace english prefix en
 param CVPSelfService-app AudAsrTTS-Transfer
 param CVPSelfService-port 7000
!
service CVPSelfService flash0:CVPSelfServiceBootstrap.vxml
 paramspace english language en
 paramspace english index 0
 paramspace english location flash0:
 paramspace english prefix en
!
service ringtone flash0:ringtone.tcl
 paramspace english language en
 paramspace english index 0
 paramspace english location flash0:
 paramspace english prefix en
!
service cvperror flash0:cvperror.tcl
 paramspace english index 0
 paramspace english language en
 paramspace english location flash0:
 paramspace english prefix en
!
service bootstrapssl flash0:bootstrap.tcl
 param cvpserverssl 1
 paramspace english language en
 paramspace english index 0
 paramspace english location flash0:
 paramspace english prefix en
 param cvpserverport 8443
!
service handoff flash0:handoff.tcl
 paramspace english language en
 paramspace english index 0
 paramspace english location flash0:
 paramspace english prefix en
!
service bootstrap flash0:bootstrap.tcl
 paramspace english index 0
 paramspace english language en
 paramspace english location flash0:
 paramspace english prefix en
 param cvpserverssl 0
!
monitor
  interface stats
  interface event-log
  interface event-log aaa
  interface event-log asr
  interface event-log tts
  interface event-log http
  interface event-log tftp
  interface event-log rtsp
  interface event-log ram
 !
!
mrcp client rtpsetup enable
vxml version 2.0
license udi pid C3900-SPE150/K9 sn FHH123000HU
hw-module pvdm 0/0
!
hw-module pvdm 0/1
!
hw-module pvdm 0/2
!
```

```
hw-module pvdm 0/3
!
!
!
archive
 log config
  hidekeys
!
redundancy
!
!
!
!
controller E1 0/0/0
!
controller E1 0/0/1
!
controller E1 0/1/0
!
controller E1 0/1/1
!
controller E1 0/2/0
!
controller E1 0/2/1
!
controller E1 0/3/0
!
controller E1 0/3/1
!
controller E1 1/0
!
controller E1 1/1
!
controller E1 1/0/0
!
controller E1 1/0/1
!
controller E1 2/0
!
controller E1 2/1
!
controller E1 2/0/0
!
controller E1 2/0/1
!
controller E1 3/0
!
controller E1 3/1
!
controller E1 3/0/0
!
controller E1 3/0/1
!
controller E1 4/0
!
controller E1 4/1
!
controller E1 4/0/0
!
controller E1 4/0/1
!
ip ftp username administrator
ip ftp password roZes
ip ssh version 1
```

```
!
!
!
!
!
!
!
interface GigabitEthernet0/0
 ip address 10.1.175.6 255.255.0.0
 no ip redirects
 ip route-cache same-interface
 duplex auto
 speed auto
 no keepalive
!
interface GigabitEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
 media-type rj45
 no keepalive
!
interface GigabitEthernet0/2
 no ip address
 shutdown
 duplex auto
 speed auto
!
!
ip forward-protocol nd
!
ip http server
no ip http secure-server
!
ip route 1.1.0.0 255.255.0.0 10.1.0.1
ip route 223.255.254.254 255.255.255.255 10.1.0.1
!
!
!
!
!
nls resp-timeout 1
cpd cr-id 1
!
!
control-plane
!
call treatment on
call threshold global cpu-5sec low 100 high 100
call threshold global cpu-avg low 100 high 100
!
!
!
!
dial-peer voice 6 voip
 service bootstrap
 incoming called-number 888999T
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
dial-peer voice 9191 voip
 service ringtone
```

```
 incoming called-number 91T
 dtmf-relay rtp-nte
 codec g711ulaw
!
dial-peer voice 9292 voip
 service cvperror
 incoming called-number 92T
 dtmf-relay rtp-nte
 codec g711ulaw
!
!
!
!
!
gatekeeper
 shutdown
!
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
 exec-timeout 0 0
 password lab
 login
 transport input all
!

scheduler allocate 20000 1000
end
```

Static Routes on Cisco Unified SIP Proxy

The following is the configuration of the Cisco Unified SIP Proxy. Significant sections of this output are shown in bold type for emphasis.

```
tover-cusp-module(cusp)> sh config active
Building CUSP configuration...
!
server-group sip global-load-balance call-id
server-group sip retry-after 0
server-group sip element-retries udp 2
server-group sip element-retries tls 1
server-group sip element-retries tcp 1
sip dns-srv
 enable
 no naptr
 end dns
!
no sip header-compaction
no sip logging
!
sip max-forwards 70
sip network netDEST standard
 no non-invite-provisional
 allow-connections
 retransmit-count invite-client-transaction 5
 retransmit-count invite-server-transaction 9
 retransmit-count non-invite-client-transaction 9
 retransmit-timer T1 500
 retransmit-timer T2 4000
 retransmit-timer T4 5000
```

```
 retransmit-timer TU1 5000
 retransmit-timer TU2 32000
 retransmit-timer clientTn 64000
 retransmit-timer serverTn 64000
 udp max-datagram-size 1500
 end network
!
sip overload reject retry-after 0
!
no sip peg-counting
!
sip privacy service
sip queue message
 drop-policy head
 low-threshold 80
 size 2000
 thread-count 20
 end queue
!
sip queue radius
 drop-policy head
 low-threshold 80
 size 2000
 thread-count 20
 end queue
!
sip queue request
 drop-policy head
 low-threshold 80
 size 2000
 thread-count 20
 end queue
!
sip queue response
 drop-policy head
 low-threshold 80
 size 2000
 thread-count 20
 end queue
!
sip queue st-callback
 drop-policy head
 low-threshold 80
 size 2000
 thread-count 10
 end queue
!
sip queue timer
 drop-policy none
 low-threshold 80
 size 2500
 thread-count 8
 end queue
!
sip queue xcl
 drop-policy head
 low-threshold 80
 size 2000
 thread-count 2
 end queue
!
route recursion
!
sip tcp connection-timeout 30
```

```
sip tcp max-connections 256
!
no sip tls
!
trigger condition in-netDEST
 sequence 1
  in-network netDEST
  end sequence
 end trigger condition
!
trigger condition mid-dialog
 sequence 1
  mid-dialog
  end sequence
 end trigger condition
!
accounting
 no enable
 no client-side
 no server-side
 end accounting
!
route table CVPtoCUCM
 key 1 target-destination 10.1.174.125 netDEST
 key 2 target-destination 10.1.174.125 netDEST
 key 3 target-destination 10.1.174.125 netDEST
 key 5 target-destination 10.1.174.125 netDEST
 key 65 target-destination 10.1.175.7 netDEST
 key 71 target-destination 10.1.174.210 netDEST
 key 888999 target-destination 10.1.175.6 netDEST
 key 91 target-destination 10.1.175.6 netDEST
 key 92 target-destination 10.1.175.6 netDEST
 end route table
!
policy lookup CVPtoCUCM
 sequence 1 CVPtoCUCM request-uri uri-component user
  rule prefix
  end sequence
 end policy
!
trigger routing sequence 1 by-pass condition mid-dialog
trigger routing sequence 2 policy CVPtoCUCM condition in-netDEST
!
no server-group sip global-ping
!
sip record-route netDEST udp 10.1.174.221
sip listen netDEST udp 10.1.174.221 5060
!
end
```

The 888999* entry points to the VoiceXML gateway whose IP address is 10.1.175.6 and not to the Cisco Unified Border Element. The VRU leg is established with the Cisco IOS VoiceXML gateway, that is, VoiceXML runs on a separate Cisco IOS gateway.

Cisco Unified Communications Manager Configuration

The Cisco Unified Communications Manager configuration for the centralized model is the same as that described for the distributed model in the "Cisco Unified Communications Manager Configuration" section on page 18.

Calls originated by Cisco Unified Communications Manager can use Cisco Unified Border Element and needs the configuration on Cisco Unified Communications Manager shown in Figure 5, Figure 6, and Figure 7.

In Figure 5, Cisco Unified Border Element is added as a SIP Trunk to the Cisco Unified Communications Manager for SIP-to-SIP calls.

Figure 5 Cisco Unified Communications Manager

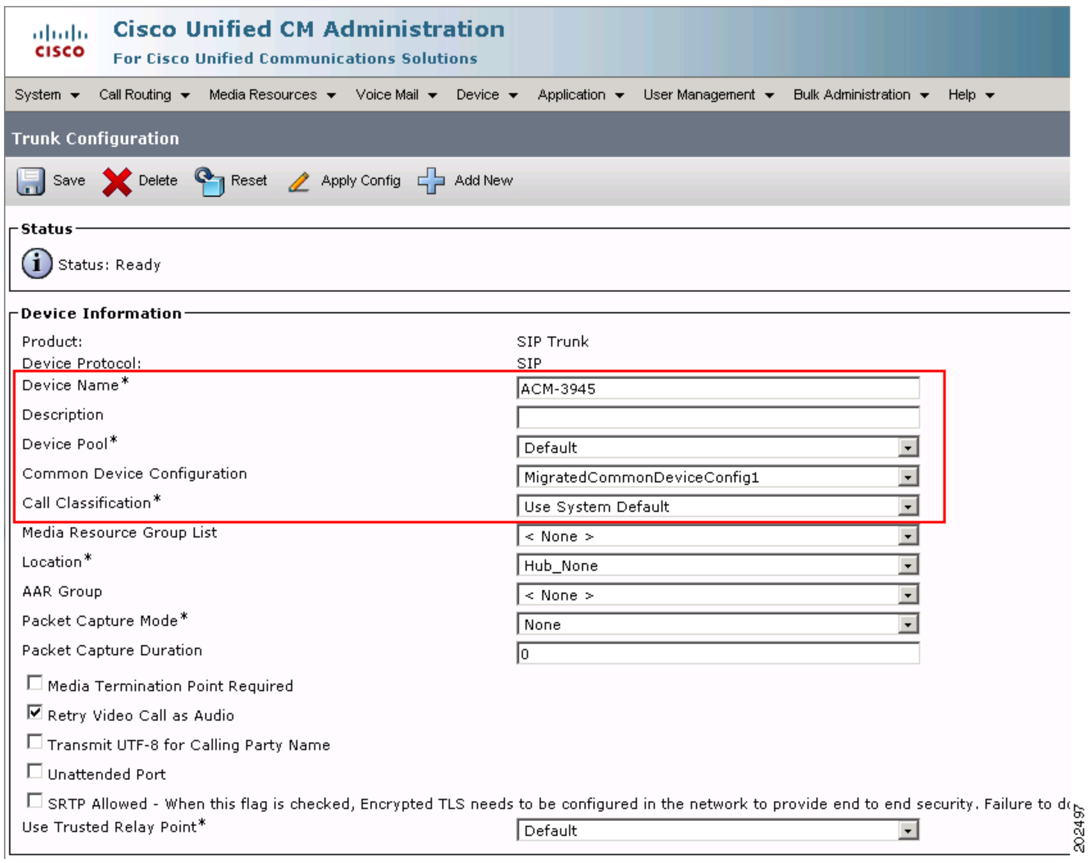


Figure 6 shows the SIP information configuration on the Cisco Unified Communications Manager.

Figure 6 Cisco Unified Communications Manager

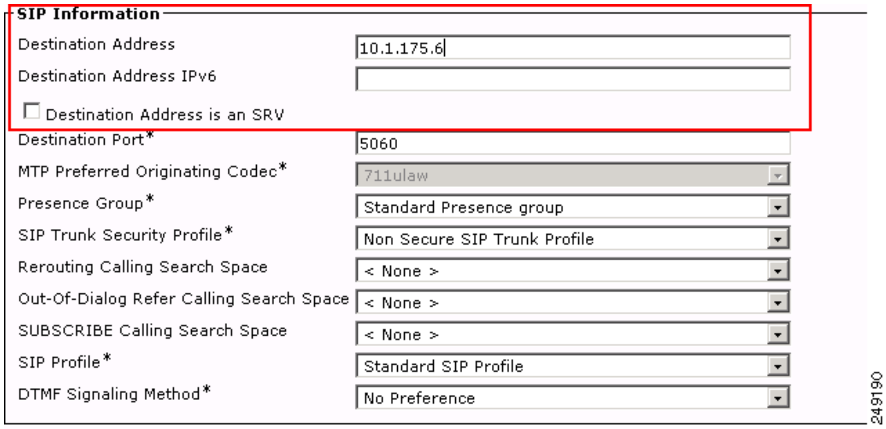
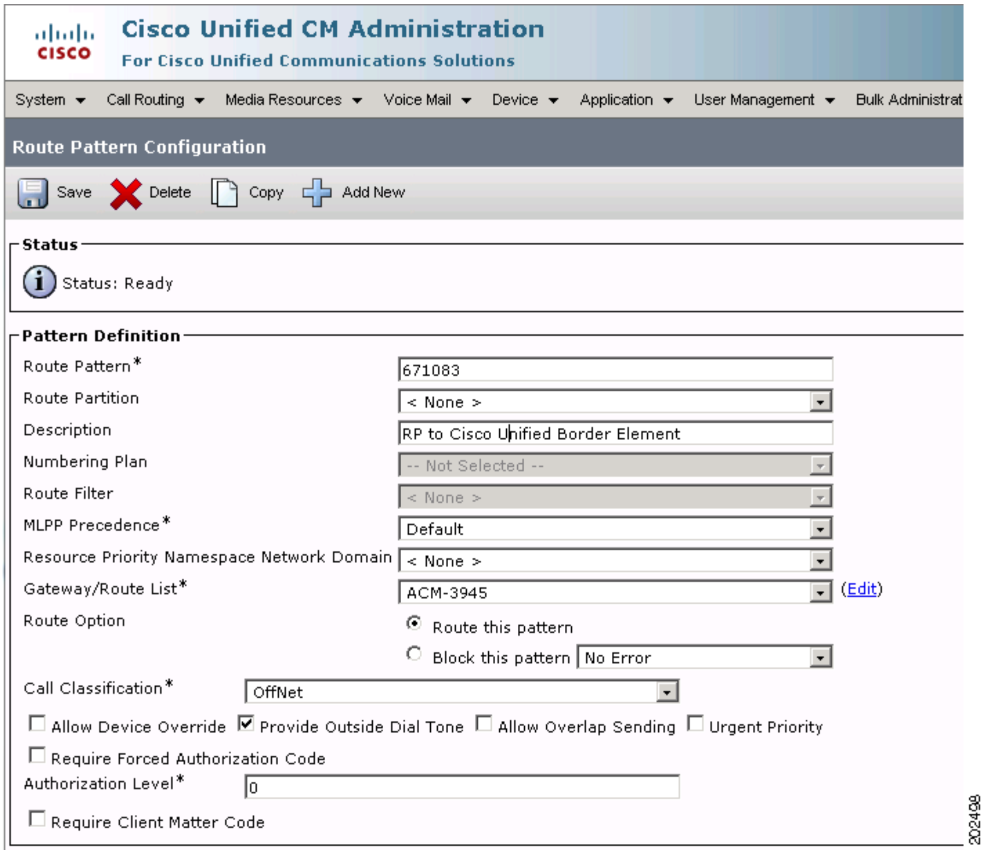


Figure 7 shows the route that is required on the Cisco Unified Communications Manager to route the calls through Cisco Unified Border Element.

Figure 7 Cisco Unified Communications Manager



Verify

Use this section to confirm that your configuration works properly. To display and verify your configuration, use the following **show** commands:

- **show call active voice brief**
- **show mrcp client session active detail**
- **show voip rtp connections**
- **show http client cache**

See *Cisco IOS Voice Command Reference* for more information.

Troubleshoot

Use this section to troubleshoot your configuration.

Note Refer to *Important Information on Debug Commands* before you use **debug** commands.

Use the following **debug** commands to troubleshoot your configuration:

- **debug ccsip messages—**This command displays all the SIP service provider interface (SPI) messages. It traces the SIP messages exchanged between the gateway and other User Agents (UAs) and Cisco Unified SIP Proxy.
- **debug http client all—**This command displays all debugging messages for the HTTP client.
- **debug mrcp all—**This command displays all debugging messages for MRCP operations.
- **debug ssl openssl error, debug ssl openssl msg, debug ssl openssl state—**These commands monitor HTTPS connections only. They do not produce any output for HTTP connections.
- **debug voip application vxml—**Use this command to troubleshoot a VoiceXML application. The output from this command can be verbose.
- **debug voice ccapi inout**—The command traces the execution path through the call control application programming interface (CCAPI), which serves as the interface between the call session application and the underlying network-specific software. Use the output from this command to understand how calls are being handled by the voice gateway. This command shows how a call flows through the system. At this debug level, you can see the call setup and teardown operations that are performed on the telephony and network call legs.

Related Information

The following information is referenced in this guide:

- *Cisco Unified Customer Voice Portal (CVP) 4.x Solution Reference Network Design (SRND)*
- *Cisco Unified Border Element (CUBE) White Papers*
- *Cisco Unified Communications SRND Based on Cisco Unified Communications Manager 5.x*