Cisco Unified Communications Manager with Cisco VCS (SIP Trunk)

Deployment Guide

Cisco VCS X8.6
Unified CM 8.6.x, 9.x, 10.x

July 2015

Contents

Introduction

This deployment guide provides guidelines on how to configure the Cisco TelePresence Video Communication Server (VCS) version X8.6 and Cisco Unified Communications Manager (Unified CM) versions 8.6.x and 9.x to interwork via a SIP trunk.

Deployment scenario

A company already has Unified CM running their telephone/video system. They want to integrate this with a VCS Control which connects their existing (or new) video conferencing systems, so that voice and video terminals can communicate with one another across one unified network.

The existing telephone system uses telephone (digit-only) numbers to specify who to call. This functionality is to be extended into the video system, so that all endpoints will be contactable by telephone numbers.

For the purposes of this example, endpoints connected to the Unified CM are identified by 3xxx extension numbers and endpoints connected to the VCS Control are identified by 4xxx extension numbers. (Note that more complicated dial plans can also be supported, including alphanumeric dialing; they would require additional transforms/routing configuration).

Unified CM and the VCS Control are connected together using a SIP trunk across an IP network; the VCS Control domain is vcs.domain. Calls sent to Unified CM will have the domain portion set to the VCS domain; calls from Unified CM to VCS will arrive with the domain portion set as <FQDN of VCS>:5060 for TCP and <FQDN of VCS>:5061 for TLS.

It is assumed that the VCS Control is running version X7 or later code and has at least the following option keys installed:

- Traversal calls
- Non-traversal calls

Summary of configuration process

This document specifies how to configure both the Unified CM and the VCS Control so that calls can be made:

- from video endpoints connected to the VCS to other video endpoints connected to that same VCS
- from IP handsets or other devices connected to Unified CM to other IP handsets or devices connected to that same Unified CM
- from video endpoints connected to the VCS to IP handsets or other devices connected to Unified CM
- from IP handsets or other devices connected to Unified CM to video endpoints connected to the VCS

The configuration process describes each of these stages separately, so that individual stages can be implemented and tested before moving on to the next.

Initially the configuration use non-secure TCP connections, as this allows for easier troubleshooting. It then describes how to secure the video network over TLS.

Prerequisites for system configuration

Before using this document to configure the VCS Control and Unified CM to interwork, make sure that:

- Unified CM contains a basic configuration and has already set up at least:
  - System > Server
  - System > Cisco Unified CM
  - System > Cisco Unified CM Group
  - System > Date / Time Group
  - System > Presence Group
  - System > Region Information
  - System > Device Pool
  - System > DHCP
  - System > Location
  - System > Physical location
  - System > Enterprise parameters
  - System > Licensing
- The VCS Control must be configured with IP address, DNS and NTP information, and is accessible for management via its web interface (see *VCS Basic Configuration (Single VCS Control) Deployment Guide*).

Enabling calls between endpoints registered on the VCS Control

VCS Control configuration

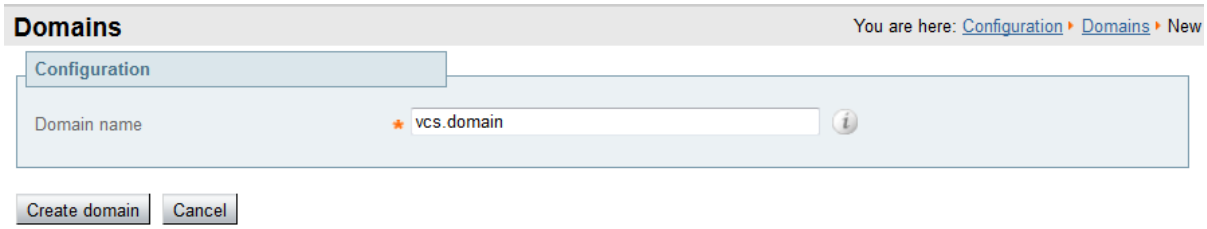
Configuration of the VCS Control to enable calls to be made between devices that register to it can be broken down into the following steps:

- Setting up the SIP domain of the VCS Control. This is needed for SIP registration.
- Creating a transform to ensure that domain information is added to dialed numbers that do not include a domain. This forces dialed number information from SIP and H.323 endpoints into a common format: number@domain.

Setting up the SIP domain of the VCS Control

SIP endpoints register with the VCS with an AOR (Address Of Record) in the format 4_digit_ number@vcs.domain. The VCS must be configured with the SIP domain information so that it will accept these registrations.

1. Go to **Configuration > Domains**.
2. Click **New**.
3. Enter the domain **Name**, for example vcs.domain.
4. Click **Create domain**.



Creating transforms

In this deployment scenario, users want to be able to dial other endpoints registered to the VCS using a 4xxx extension number. Unified CM endpoints are to be dialed using a 3xxx number. This dialing model can be supported by H.323 (if the endpoint registers the 4-digit E.164 alias), however, SIP does not support dialing by numbers alone. If a number (without a domain appended) is dialed from a SIP endpoint the endpoint will automatically append its own domain.

For consistency with both SIP and H.323 dialing, this deployment scenario always uses the URI form for routing calls (that is, dialed_digits@domain). When the VCS receives a call request, the dialed number:

- will contain the 4 digit extension number that identifies the specific endpoint to route to
- may or may not include a domain (only included when a SIP endpoint is making the call)

Thus, a transform is needed to ensure that the dialed number is transformed into a consistent form, in this case to add the domain (vcs.domain) if required. To achieve this, a regex is used: `([^@]*)` transforms to `\1@vcs.domain` (any dialed information which does not contain a domain – does not contain an '@' – has the '@vcs.domain' added.)

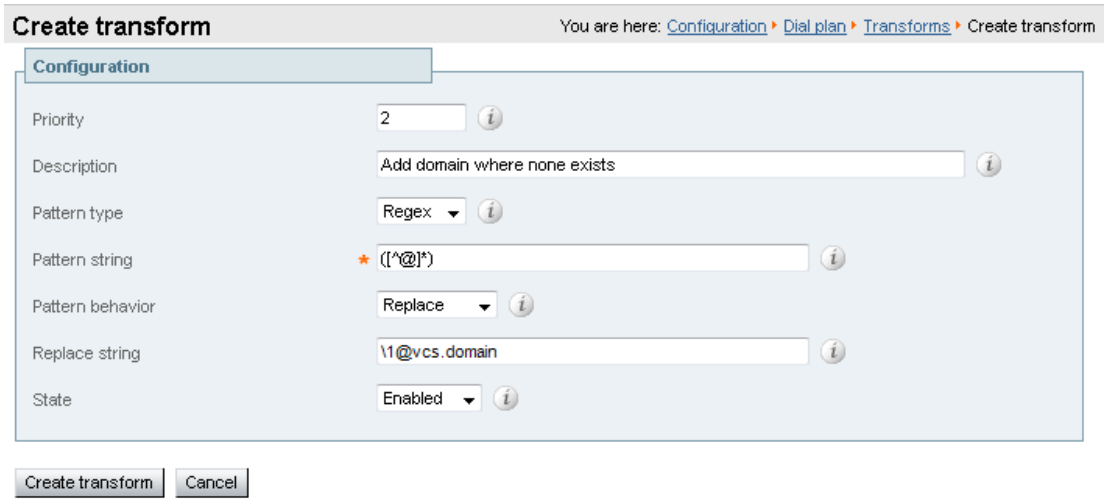
See the Regular Expression Reference in the Appendices section of *VCS Administrator Guide* for further details, or alternatively search the internet for the term "Regular Expression".

To create the transform:

1. Go to **Configuration > Dial plan > Transforms**.
2. Click **New**.
3. Configure the fields as follows:

Priority	2
Description	"Add domain where none exists" for example
Pattern type	*Regex*
Pattern string	([^@]*)
Pattern behavior	*Replace*
Replace string	\1@vcs.domain
State	*Enabled*

4. Click **Create transform**.



Unified CM configuration

No configuration is required on Unified CM for the VCS to route calls between endpoints registered locally to the VCS.

Registering endpoints to the VCS Control

Endpoint configuration

For H.323, configure the endpoints as follows:

- H.323 ID (for example 4000@vcs.domain, 4001@vcs.domain and so on)
- H.323 Call Setup = Gatekeeper
- Gatekeeper IP address = IP address of the VCS

For SIP, configure the endpoints as follows:

- SIP Address (URI) (for example 4000@vcs.domain, 4001@vcs.domain and so on)
- Server Address (Proxy address) = IP address of the VCS

Confirming registrations

Registration status can be confirmed by checking the VCS via **Status > Registrations**.

By default the VCS will accept all H.323 registrations and all SIP registrations within the specified SIP domain. You can limit registrations by explicitly allowing or denying individual registrations. See the "VCS Configuration" section of *VCS Administrator Guide* for further details.

Test calls

Make some test calls.

Your call history can be seen on the VCS via **Status > Calls > History**.

Enabling calls between endpoints registered on Unified CM

VCS Control configuration

No configuration is required on the VCS for Unified CM to route calls between endpoints registered locally to the Unified CM.

Unified CM configuration

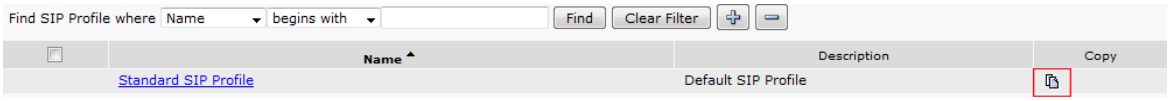
The configuration of Unified CM and Cisco phones to enable calls to be made between the phones consists of setting up a SIP Profile, specifying the phones on Unified CM, giving the phones phone numbers and getting the phones to load their configuration. This comprises the following steps:

- Configuring the SIP Profile for VCS (already exists if using version 9.x)
- Configuring the region with an appropriate session bit rate for video calls
- Configuring a SIP Profile for phone devices
- Adding a phone device: add the new phone device to the list of supported endpoints on Unified CM
- Configuring the device directory number: specify the telephone number that will cause this phone to ring
- Configure the phone endpoint to pick up its configuration from Unified CM.

Configuring the SIP Profile for VCS

Note: This procedure does not apply to Unified CM versions 9.x and later, because the newer versions have a "Standard SIP Profile For Cisco VCS" .

1. On Unified CM, go to **Device > Device Settings > SIP Profile**.
2. Click **Copy** against the **Standard SIP Profile**.



3. Configure the fields as follows (leave other fields as default values):

Name	"Standard SIP Profile For Cisco VCS"
Default MTP Telephony Event Payload Type	101
Redirect by Application	Select the check box
Use Fully Qualified Domain in SIP Requests	Select the check box
Allow Presentation Sharing using BFCP	Select the check box (in Unified CM 8.6.1 or later)
Timer Invite Expires	180
Timer Register Delta	5
Timer Register Expires	3600

Timer T1	500
Timer T2	Leave as default (typically 4000 or 5000)
Retry INVITE	6
Retry non-INVITE	10
Start Media Port	16384
Stop Media Port	32766
Call Pickup URI	x-cisco-serviceuri-pickup
Call Pickup Group Other URI	x-cisco-serviceuri-opickup
Call Pickup Group URI	x-cisco-serviceuri-gpickup
Meet Me Service URI	x-cisco-serviceuri-meetme
Timer Keep Alive Expires	120
Timer Subscribe Expires	120
Timer Subscribe Delta	5
Maximum Redirections	70
Off Hook To First Digit Timer	15000
Call Forward URI	x-cisco-serviceuri-cfwdall
Abbreviated Dial URI	x-cisco-serviceuri-abbrdial
Reroute Incoming Request to new Trunk based on	Never

4. Click **Save**.

Configuring the region with an appropriate session bit rate for video calls

Ensure that your regions have an appropriate session bit rate for video calls:

1. Go to **System > Region Information > Region**.
2. Select the region (for example the **Default** region).
3. Set **Maximum Session Bit Rate for Video Calls** to a suitable upper limit for your system, for example 6000 kbps.
4. Click **Save** and then click **Apply Config**.

Configuring the SIP Profile for phone devices

This creates the SIP Profile that is to be applied to all phone devices.

1. On Unified CM, go to **Device > Device Settings > SIP Profile**.
2. Click **Copy** against the **Standard SIP Profile**.
3. Configure the following fields, leaving everything else as its default value:

Name	Standard SIP Profile – for phone devices
Use Fully Qualified Domain in SIP Requests	Select the check box
Allow Presentation Sharing using BFCP	Select the check box if BFCP (Dual video / presentation sharing) is required.

4. Click **Save**.

Adding a phone device

1. Go to **Device > Phone**.
2. Click **Add New**.
3. Select a **SIP Profile** of *Standard SIP Profile – for phone devices*.
4. Configure the other fields as required.
5. Click **Save** and click **OK**.
6. Click **Apply Config** and click **OK**.

Alternatively, if there is already another phone configured, copy its configuration by selecting “super copy”, entering the new phone’s MAC address and then changing the description (especially correct the MAC address part of the description).

Configuring the device directory number

1. Go to **Device > Phone**.
2. Select the relevant device name.
3. On the left hand side, select a line.
4. Set up the required directory number (for this example use a 3xxx number).

Configuring phone endpoint to pick up its configuration from Unified CM

Cisco phones

1. Press the **settings** button.
2. Select the Network Configuration section, and check whether the **TFTP Server** is the IP address of Unified CM. If not:
   a. Press the **settings** button twice – to return to SETTINGS menu.
   b. Select **Unlock** and enter the appropriate password.
   c. Select the Network Configuration section.
   d. Set **Alternate TFTP** = *YES*.
   e. Set **TFTP Server** = <IP address of Unified CM>.
   f. Select **Accept**.
   g. Select **Save**.

The phone should now indicate that Line 1 is the phone number specified on Unified CM (for example 3001).

EX60 / EX90 using TE6 software

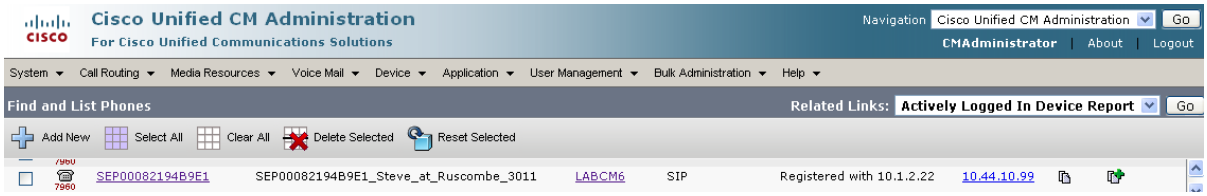
Enter the relevant provisioning details:

- Select a **Provisioning Mode** of *Cisco UCM*.
- Enter the **Address of the provisioning manager**.

Calls can now be made between handsets registered on Unified CM.

Confirming registrations

Registration status of phones connected to Unified CM can be seen on the **Device > Phone** page.



Test calls

Make some test calls by dialing the numbers of the registered phones (for example, 3001).

Enabling endpoints registered on VCS to call endpoints registered on Unified CM

Unified CM configuration

Configuration of Unified CM to enable calls to be made between devices that register to it can be broken down into 3 steps:

- Configuring the SIP Trunk security profile
- Configuring the SIP Trunk device
- Configuring the Cluster Fully Qualified Domain Name

Configuring the SIP Trunk security profile

1. Go to **System > Security > SIP Trunk Security Profile**.
2. (Before version 9.x) Click **Add New** and name the new profile.
3. (9.x onwards) Select **Non Secure SIP Trunk Profile**.
4. Configure the fields as follows:

Name	Non Secure SIP Trunk Profile
Device Security Mode	Non Secure
Incoming Transport Type	TCP+UDP
Outgoing Transport Type	TCP
Incoming Port	5060
Accept Unsolicited Notification	Select this check box
Accept Replaces Header	Select this check box

5. Click **Save**.

Configuring the SIP Trunk device

1. On Unified CM, go to **Device > Trunk**.
2. Click **Add New**.
3. Select a **Trunk Type** of *SIP Trunk*.
   - **Device Protocol** displays *SIP*.
   - If asked for a **Trunk Service Type**, select *None (Default)*.
4. Click **Next**.
5. Configure the **Device Information** fields as follows:

Device Name	As required, such as VCS_system
Device Pool	(As set up in System > Device Pool)

Call classification	OnNet
Location	(As set up in System > Location)
Packet Capture Mode	None
Media Termination Point Required	Clear this check box if any video phones registered to Unified CM are to make or receive video calls with endpoints registered to VCS. Select this check box if audio devices only are registered to Unified CM.
SRTP Allowed	Select this check box. For background, read Secure RTP between CUCM and VCS or Expressway Configuration Example
Run On All Active Unified CM Nodes	Select this check box

6. Configure the **Call Routing Information > Inbound Calls** fields as follows:

Significant digits	All
Connected Line ID Presentation	Default
Connected Name Presentation	Default
Calling Search Space	(As set up in **Call Routing > Class of Control > Calling Search Space**)
Prefix DN	<blank>
Redirecting Diversion Header Delivery – Inbound	Select this check box

7. Configure the **Call Routing Information > Outbound Calls** fields as follows:

Calling Party Selection	Originator
Calling Line ID Presentation	Default
Calling Name Presentation	Default
Caller ID DN	<blank>
Caller Name	<blank>

8. Configure the **SIP Information** fields as follows:

Destination address is an SRV	Select this check box if a domain is specified for the destination address, and the DNS server uses DNS SRV records to direct the domain to a cluster of VCSs. Do not select this check box if an IP address is specified as the **Destination address**.

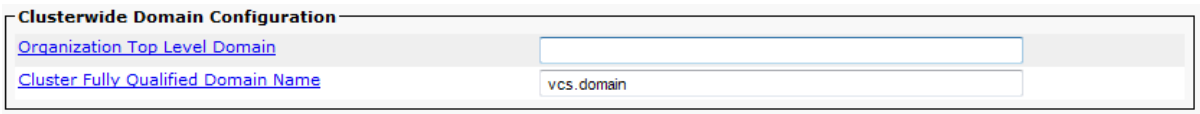
Destination address	<FQDN of VCS / VCS cluster>. Alternatively you can enter the <IP address of VCS>. If you are not using SRV records and need to specify multiple peers, click  to add extra **Destination address** rows. The content of the VCS transform configured at Creating a transform to convert Unified CM supplied domain information to the VCS SIP domain [p.21] depends on which form of address is entered here.
Destination port	5060 (this displays as zero if you are using SRV records)
Presence Group	Standard Presence Group (or whichever presence group has been configured in **System > Presence Group**)
SIP Trunk Security Profile	Non Secure SIP Trunk Profile
SIP Profile	Standard SIP Profile for Cisco VCS
DTMF Signaling Method	RFC 2833
Normalization Script	vcs-interop (if available)

9. Click **Save**.
10. Click **Reset**.
11. Click **Reset**.

Configuring the Cluster Fully Qualified Domain Name

Unified CM must be configured with a **Cluster Fully Qualified Domain Name** so that it can receive calls to addresses in the format <address>@domain. (It is also required when Unified CM is clustered so that VCS can send the call to any Unified CM node.)

1. Go to **System > Enterprise parameters**, and find the **Clusterwide Domain Configuration** section.
2. Set the **Cluster Fully Qualified Domain Name** to the same domain as the video network, for example vcs.domain.
   This parameter defines one or more Fully Qualified Domain Names (FQDNs) for this cluster. Multiple FQDNs must be separated by a space. Requests containing URLs (for example, SIP calls) whose host portion matches any of the FQDNs in this parameter will be recognized as a request destined for this cluster and/or devices attached to it.
3. Click **Save**.



VCS Control configuration

The configuration of the VCS Control has 3 steps:

- Configuring a neighbor zone that contains the Unified CM
- Configuring a search rule to route calls to that zone
- Configuring a transform that converts number@<IP address of cucm> to number@vcs.domain

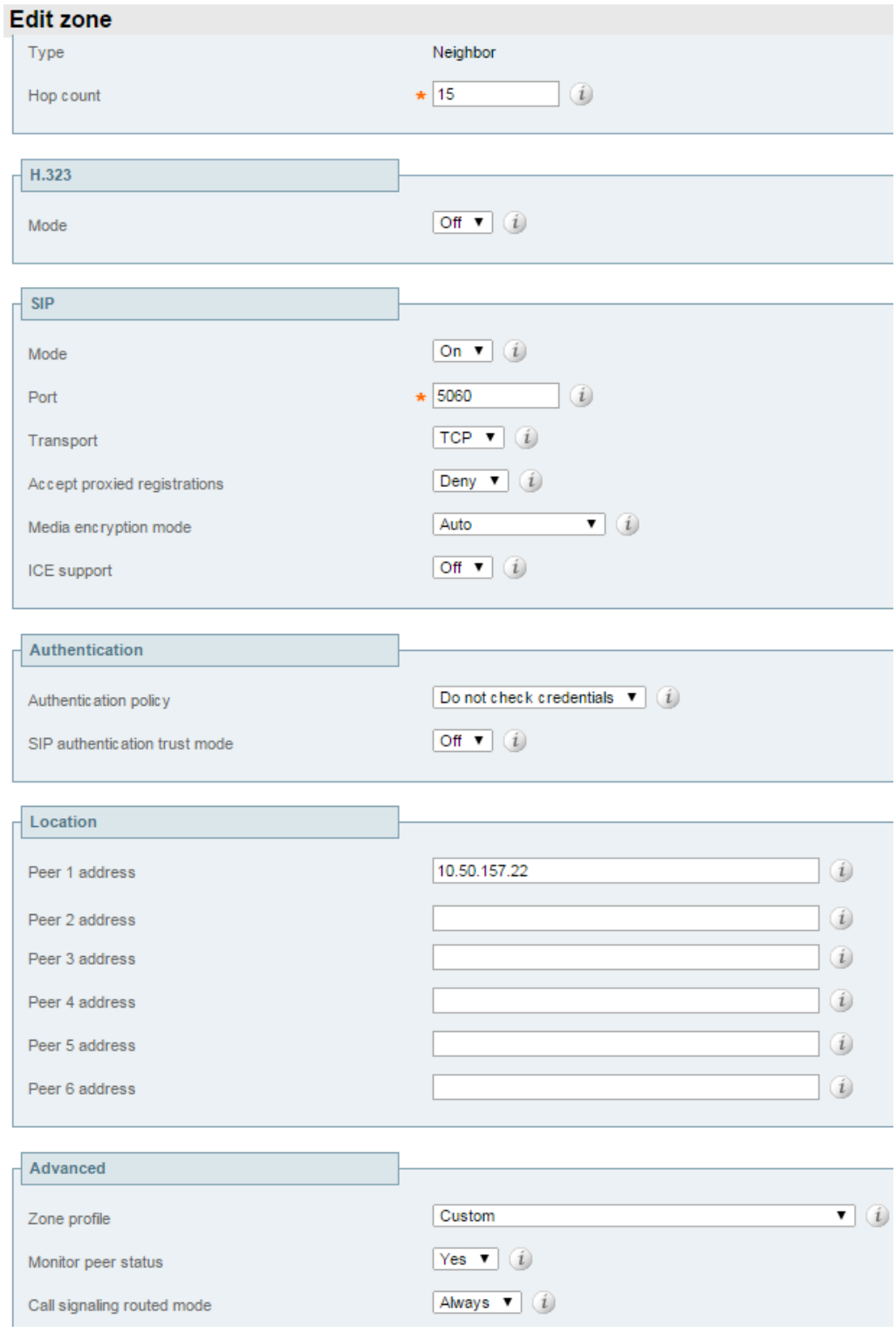
Creating a neighbor zone for Unified CM

1. Go to **Configuration > Zones > Zones**.
2. Click **New**.
3. Configure the fields as follows (leave all other fields with default values):

Name	CUCM Neighbor
Type	*Neighbor*
Hop count	15
H.323 mode	*Off* (H.323 is not supported between VCS and Unified CM)
SIP mode	*On*
SIP port	5060 for TCP or 5061 for TLS (must match the port set on the SIP trunk)
Transport	*TCP* or *TLS*. Choose *TLS* if you want secure transport and encrypted media
Accept proxied registrations	*Deny*
Media encryption mode	*Auto*
SIP authentication trust mode	*Off*
Peer 1 address	IP address of Unified CM, or the FQDN of Unified CM. If you are planning to ultimately use a TLS connection, then typically you will need to specify the FQDN of Unified CM here as this is the name that will be used to authenticate the certificate presented by Unified CM.
Zone profile (Advanced section)	This depends upon your version of Unified CM: • Select *Cisco Unified Communications Manager* for versions prior to 8.6.1 • Select *Cisco Unified Communications Manager (8.6.1 or later)* for 8.6.1 or 8.6.2 • Select *Custom* for 9.x or later and: ◦ Set **Call signaling routed mode** to *Always* ◦ Leave all the other fields as their default values Note that Unified CM 8.6.1 or later is required for BFCP (dual video / presentation sharing).

This configures the VCS to use SIP over TCP to communicate with the Unified CM. To use TLS, complete the configuration as described here for TCP and then see Connecting VCS to Unified CM Using TLS [p.24].

4. Click **Create zone**.



Creating a search rule to route calls to the Unified CM neighbor zone

Search rules specify the range of telephone numbers / URIs to be handled by this neighbor Unified CM. They can also be used to transform URIs before they are sent to the neighbor.

In this example deployment, the transforms set up in Enabling calls between endpoints registered on the VCS Control [p.7] ensure that dial strings are in URI format number@vcs.domain.

1. Go to **Configuration > Dial plan > Search rules**.
2. Click **New**.
3. Configure the fields as follows to route the call to Unified CM:

Rule name	Route to CUCM
Description	For example: Send 3xxx@vcs.domain calls to CUCM
Priority	100
Protocol	*Any*
Source	*Any*
Request must be authenticated	Configure this setting according to your authentication policy
Mode	*Alias pattern match*
Pattern type	*Regex*
Pattern string	(3\d{3})@vcs.domain(.*)
Pattern behavior	*Leave* (@domain formatted addresses will work in Unified CM due to the **Cluster Fully Qualified Domain Name** enterprise parameter)
On successful match	*Stop*
Target zone	*CUCM Neighbor*
State	*Enabled*

4. Click **Create search rule**.

See the "Zones and Neighbors" section of *VCS Administrator Guide* for further details.

Creating a transform that converts number@<IP address of cucm> to number@vcs.domain

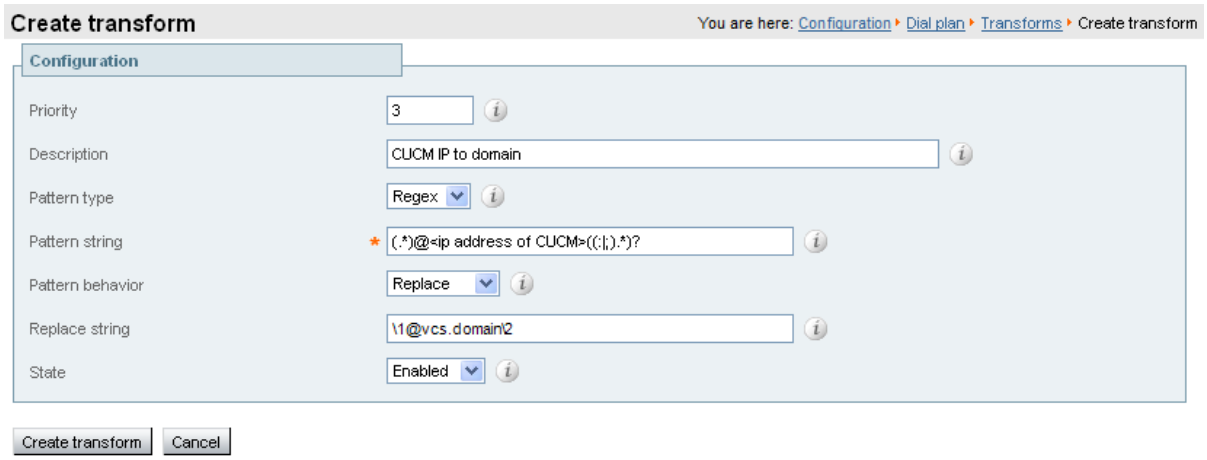
When a call is made from Unified CM to VCS, the callback address is presented as number@<ip address of cucm>. If the VCS-registered endpoint returns the call, the VCS needs to be able to route it back to Unified CM. To enable this, the domain portion of the address must have the IP address removed and the video domain added (so that the existing search rule can route the call to Unified CM). A transform is required:

1. Go to **Configuration > Dial plan > Transforms**.
2. Click **New**.
3. Configure the fields as follows:

Priority	3
Description	"CUCM IP to domain" for example
Pattern type	*Regex*

Pattern string	(.*)@<ip address of Unified CM>((:|;).*)? If a Unified CM cluster is in use, the regex must cater for the IP address of every possible node, for example (.*)@(10\.1\.2\.22|10\.1\.2\.23)((:|;).*)?
Pattern behavior	*Replace*
Replace string	\1@vcs.domain\2
State	*Enabled*

4. Click **Create transform**.



Test calls

Make some test calls from endpoints registered on the VCS Control to endpoints registered on Unified CM by dialing the required Unified CM extension number (3xxx) on the VCS endpoint.

Enabling endpoints registered on Unified CM to call endpoints registered on VCS

VCS Control configuration

Creating a transform to convert Unified CM supplied domain information to the VCS SIP domain

This transform converts URIs received from Unified CM to the format used in the VCS's Local Zone and thus expected within any neighbor zones.

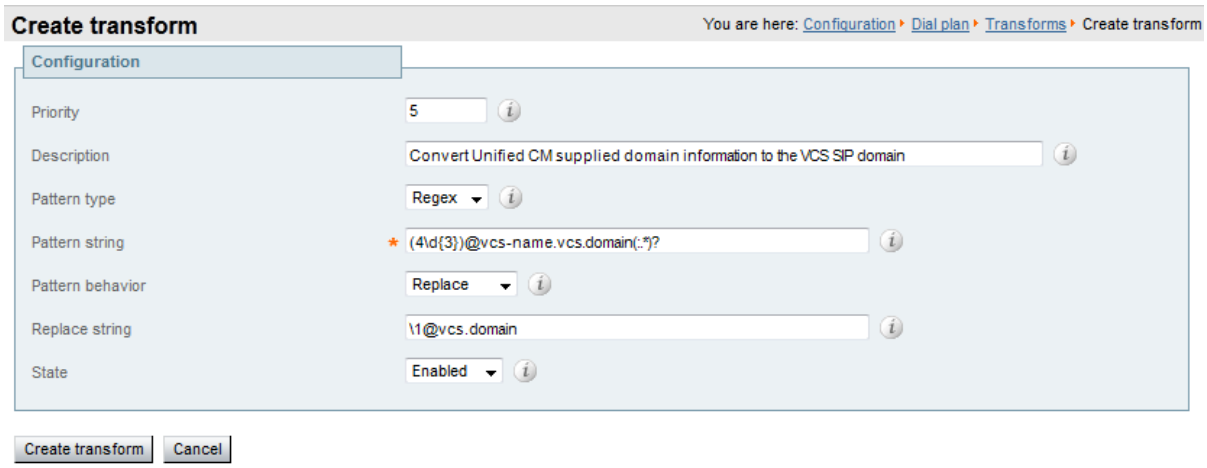
The domain portion of the URI received from Unified CM depends on its SIP Trunk configuration (see Configuring the SIP Trunk device [p.14]). Thus, this could be the IP address:port of the VCS or the FQDN of the VCS or VCS cluster.

In this example, it is matching URIs received from Unified CM in the form 4xxx@vcs-name.vcs.domain:<port> and converting it into 4xxx@vcs.domain.

1. Go to **Configuration > Dial plan > Transforms**.
2. Click **New**.
3. Configure the fields as follows:

Description	Convert Unified CM supplied domain information to the VCS SIP domain
Priority	Enter a high priority such as 5 (the priority of this transform should be before any transforms that need to be applied for searching local and neighbor zones)
Pattern type	*Regex*
Pattern string	For example: (4\d{3})@vcs-name.vcs.domain(:.*)?
Pattern behavior	Replace
Replace string	For example: \1@vcs.domain
State	Enabled

4. Click **Create transform**.



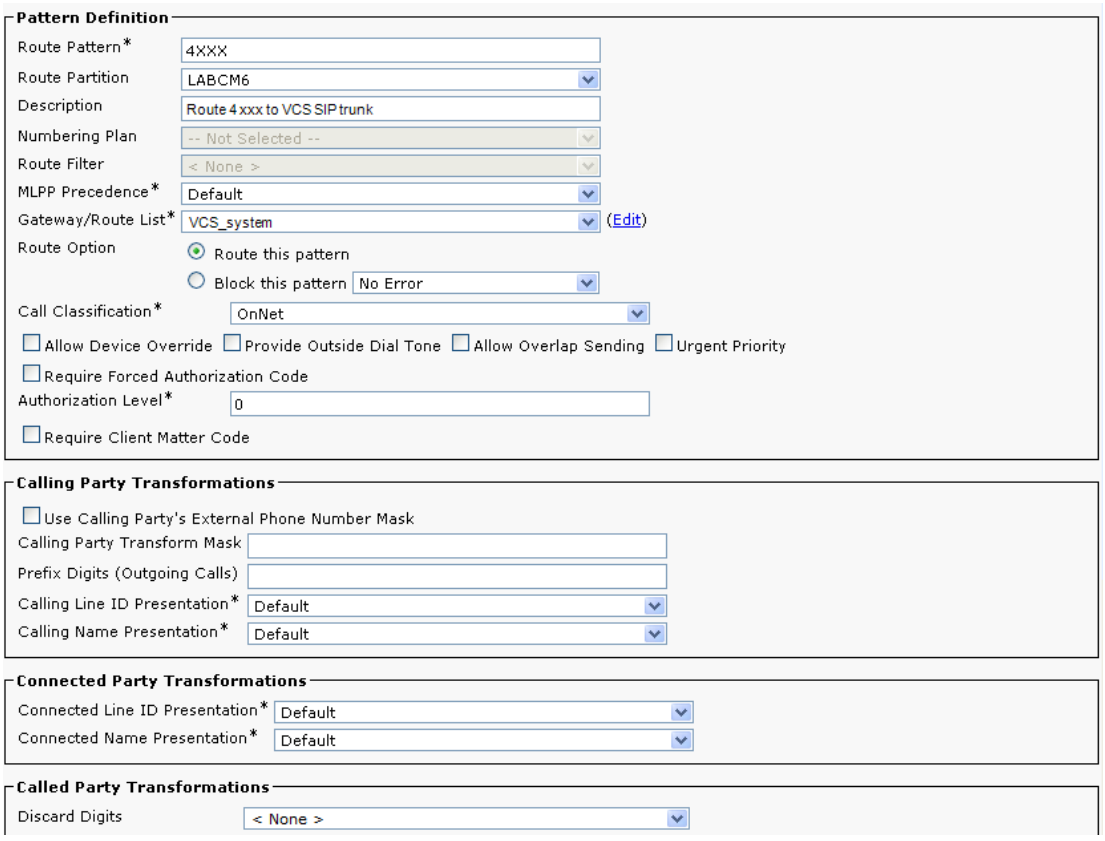
Unified CM configuration

Allowing numeric dialing from Cisco phones to VCS

Unified CM can be configured to take a prefix and route calls to a SIP trunk based on a specific prefix. Configure Unified CM to route calls dialed as 4xxx to the VCS:

1. On Unified CM, go to **Call Routing > Route/Hunt > Route Pattern**.
2. Click **Add New**.
3. Configure a Route Pattern to route calls dialed 4xxx to the VCS trunk (no change to dialed number).

Pattern Definitions	
Route Pattern	4XXX
Route Partition	(As set up in **System > Device Pool**)
Description	As required, for example “Route 4 xxx to VCS SIP trunk”
Gateway/Route List	Required Trunk to route calls to the VCS Control
Call Classification	*OnNet*
Provide Outside Dial Tone	Not selected
Called Party Transformations	
Discard Digits	< None >



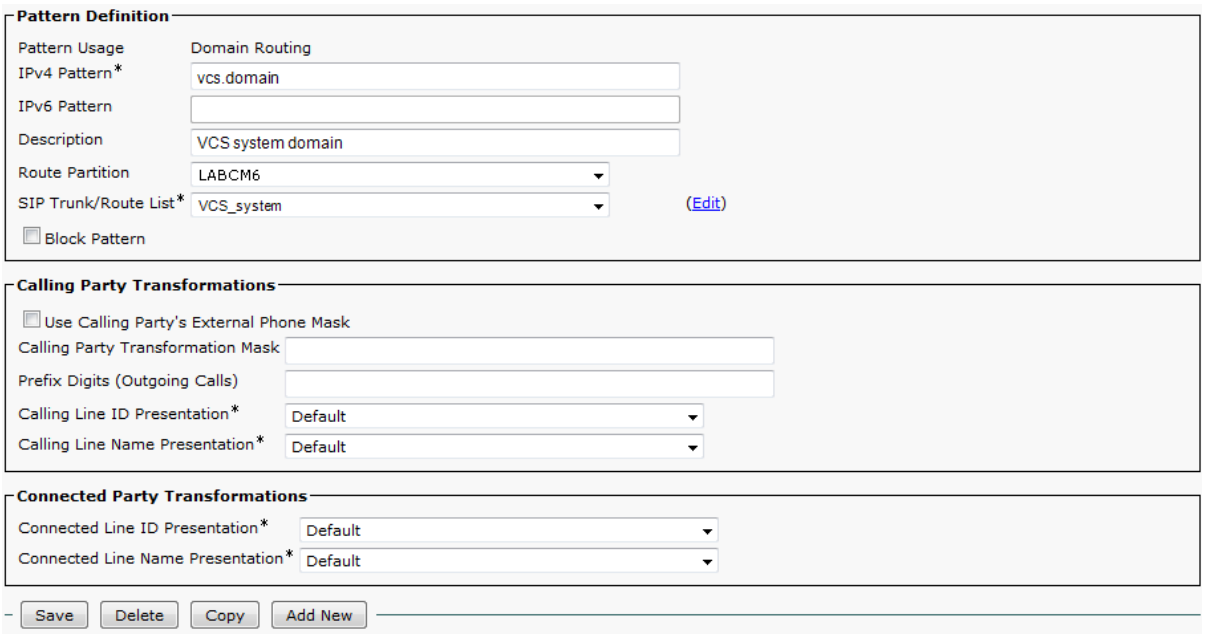
Allowing dialing to VCS domain from Cisco phones

Configure a SIP route pattern that tells Unified CM that anything with, for example, a domain vcs.domain needs to be sent down the VCS SIP trunk. This is required to permit dialing from endpoints that support SIP URIs with domains, and also for enabling the reverse path to the VCS for certain signaling.

1. On Unified CM, go to **Call Routing > SIP Route Pattern**.
2. Click **Add New**.
3. Configure the fields as follows:

Pattern Usage	*Domain Routing*
IPv4 Pattern	Domain for calls, for example vcs.domain
Route Partition	Default is “<None>”; set according to dial plan restrictions
SIP Trunk	Required Trunk to route calls to the VCS Control

4. Click **Save**.



When nnnn@vcs.domain is dialed by an endpoint registered to Unified CM, Unified CM will route the call to the VCS as nnnn@<FQDN of VCS>:5060 (TCP) or nnnn@<FQDN of VCS>:5061 (TLS). (The domain may alternatively be the IP address of VCS, depending on what is configured as the SIP Trunk **Destination Address**.)

Calls can now be made from Unified CM to endpoints on VCS registered as 4xxx@vcs.domain.

Test calls

Make some test calls from endpoints registered on Unified CM to endpoints registered on the VCS Control.

Connecting VCS to Unified CM Using TLS

These instructions explain how to take a system that is already configured and working using a TCP interconnection between VCS and Unified CM, and to convert that connection to use TLS instead. This process involves:

- Ensuring certificate trust between Unified CM and VCS
- Setting the Cluster Security Mode of the Unified CM to 1 (Mixed Mode)
- Configuring a SIP trunk security profile on Unified CM
- Updating the Unified CM trunk to VCS to use TLS
- Updating the VCS neighbor zone to Unified CM to use TLS

Ensuring Certificate Trust Between Unified CM and VCS

For Unified CM and VCS to establish a TLS connection with each other:

- VCS and Unified CM must both have valid server certificates loaded (you must replace the VCS's default server certificate with a valid server certificate)
- VCS must trust Unified CM's server certificate (the root CA of the Unified CM server certificate must be loaded onto VCS)
- Unified CM must trust VCS's server certificate (the root CA of the VCS server certificate must be loaded onto Unified CM)

See *VCS Certificate Creation and Use Deployment Guide* for full details about loading certificates and how to generate CSRs on VCS to acquire certificates from a Certificate Authority (CA).

Note: In a clustered environment, you must install CA and server certificates on each peer/node individually.

We strongly recommend that you do not use self-signed certificates in a production environment.

Loading Server and Trust Certificates on VCS

VCS server certificate

VCS has only one server certificate. By default, this is a certificate signed by a temporary certificate authority. We recommend that it is replaced by a certificate generated by a trusted certificate authority.

To upload a server certificate:

1. Go to **Maintenance > Security certificates > Server certificate**.
2. Use the **Browse** button in the **Upload new certificate** section to select and upload the **server certificate** PEM file.
3. If you used an external system to generate the Certificate Signing Request (CSR) you must also upload the **server private key** PEM file that was used to encrypt the server certificate. (The private key file will have been automatically generated and stored earlier if the VCS was used to produce the CSR for this server certificate.)
   - The **server private key** PEM file must not be password protected.
   - You cannot upload a server private key if a certificate signing request is in progress.
4. Click **Upload server certificate data**.

Note: If you are using Unified CM version 8.5(1) or earlier and are having problems establishing a TLS connection between VCS and Unified CM, we recommend adding the following x509 extended key attributes into the CSR:

- serverAuth (1.3.6.1.5.5.7.3.1) -- TLS Web server authentication
- clientAuth (1.3.6.1.5.5.7.3.2) -- TLS Web client authentication
- ipsecEndSystem (1.3.6.1.5.5.7.3.5) -- IP security end system

VCS trusted CA certificate

The **Trusted CA certificate** page (**Maintenance > Security certificates > Trusted CA certificate**) allows you to manage the list of certificates for the Certificate Authorities (CAs) trusted by this VCS. When a TLS connection to VCS mandates certificate verification, the certificate presented to the VCS must be signed by a trusted CA in this list and there must be a full chain of trust (intermediate CAs) to the root CA.

The root CA of the Unified CM server certificate must be loaded into the VCS's trusted CA certificate list.

To upload a new file containing one or more CA certificates, **Browse** to the required PEM file and click **Append CA certificate**. This will append any new certificates to the existing list of CA certificates. If you are replacing existing certificates for a particular issuer and subject, you have to manually delete the previous certificates.

Repeat this process on every VCS that will communicate with this Unified CM.

Loading Server and Trust Certificates on Unified CM

Certificate management for Unified CM is performed in the **Cisco Unified OS Administration** application.

All existing certificates are listed under **Security > Certificate Management**. Server certificates are of type *certs* and trusted CA certificates are of type *trust-certs*.

Unified CM server certificate

By default, Unified CM has a self-signed server certificate **CallManager.pem** installed. We recommend that this is replaced with a certificate generated from a trusted certificate authority.

Unified CM trusted CA certificate

To load the root CA certificate of the authority that issued the VCS certificate (if it is not already loaded):

1. Click **Upload Certificate/Certificate chain**.
2. Select a **Certificate Name** of *CallManager-trust*.
3. Click **Browse** and select the file containing the root CA certificate of the authority that issued the VCS certificate.
4. Click **Upload File**.

Repeat this process on every Unified CM server that will communicate with VCS. Typically this is every node that is running the CallManager service.

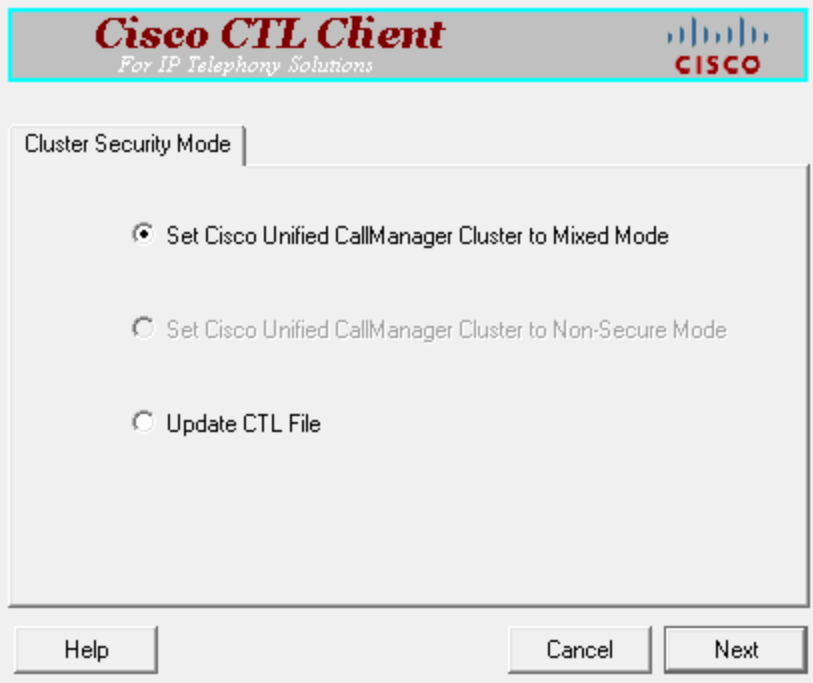
Setting the Cluster Security Mode to Mixed Mode

The Cisco Unified Communications Manager cluster must be in Mixed Mode to allow the registration of both secure devices and non-secure devices. This allows for best effort encryption between the VCS and the Cisco Unified Communications Manager. Read Setting up secure RTP between UCM and VCS for background on best effort encryption between VCS and Unified CM.

As of version 10.0, you can use the CLI to change the cluster security mode. On earlier versions, you must use the Cisco CTL Client plugin to change the cluster security mode. The security mode change updates the CTL file, so you must restart the Cisco CallManager and Cisco Tftp services after the change.

The process is summarized below, but you should refer to the *Cisco Unified Communications Manager Security Guide* for your version, which you can find on the Cisco Unified Communications Manager (CallManager) Maintain and Operate Guides page.

1. Obtain access to the Unified CM publisher node, including hardware security tokens (if using the CTL Client plugin).
2. (Pre 10.0) Download and install the Cisco CTL Client plugin from Unified CM.
3. Run the CTL Client plugin to enable Mixed Mode. On 10.0 or later, you can use **`utils ctl set-cluster mixed-mode`** at the CLI.



4. Update the CTL file (via the plugin or **`utils ctl update CTLFile`**).
5. Restart the Cisco CallManager and Cisco Tftp services (via Cisco Unified Serviceability).

Configuring a SIP Trunk Security Profile on Unified CM

On Unified CM:

1. Select **Cisco Unified CM Administration**, click **Go** and log in.
2. Go to **System > Security > SIP Trunk Security Profile**.
3. Click **Add New**.
4. Configure the fields as follows:

Name	A name indicating that this is an encrypted profile.
Description	Enter a textual description as required.

Device Security Mode	*Encrypted*.
Incoming Transport Type	*TLS*.
Outgoing Transport Type	*TLS*.
Enable Digest Authentication	Leave unselected.
X.509 Subject Name	The subject name or an subject alternate name provided by the VCS in its certificate. For VCS clusters, ensure that this list includes all of the names contained within all of the peers' certificates. To specify multiple X.509 names, separate each name by a space, comma, semicolon or colon.
Incoming Port	5061
Accept Unsolicited Notification	Select this check box
Accept Replaces Header	Select this check box
Other parameters	Leave all other parameters unselected.

5. Click **Save**.

Updating the Unified CM Trunk to VCS to Use TLS

On Unified CM:

1. Go to **Device > Trunk**.
2. Using Find, select the **Device Name** previously set up for the trunk to the VCS.
3. Configure the following fields:

SIP Information section	
Destination Port	5061 (unless using DNS SRV, in which case ensure the SRV records are set up correctly). Note that some versions of Unified CM cannot perform TLS SRV lookups. See TLS calls fail when Unified CM uses SRV trunk destinations [p.33] for more information.
SIP Trunk Security Profile	Select the trunk profile set up above.

Leave other parameters as previously configured.

4. Click **Save**.
5. Click **Reset**.

Updating the VCS Neighbor Zone to Unified CM to Use TLS

Note that VCS will report that the Unified CM zone is active even while it is communicating with Unified CM over TCP. The changes below are necessary to enable communications over TLS.

On VCS:

1. Go to **Configuration > Zones > Zones**, then select the zone to Unified CM.
2. Configure the following fields:

SIP section	
Port	5061
Transport	*TLS*
TLS verify mode	*On*
Authentication trust mode	*Off*

Leave other parameters as previously configured.

3. Click **Save**.

Verifying That the TLS Connection is Operational

To verify correct TLS operation, check that the VCS zone reports its status as active and then make some test calls.

1. Check the VCS zone is active:
   a. Go to **Configuration > Zones > Zones**.
   b. Check the **SIP status** of the zone.
   If the zone is not active, try resetting or restarting the trunk again on Unified CM.
2. Make a test call from a VCS registered endpoint to a Unified CM phone.
3. Make a test call from a Unified CM phone to a VCS registered endpoint.

Network of VCSs

If there is a network of VCSs behind this VCS neighbored to Unified CM, then, either:

- Unified CM must trust the certificates of all the VCSs in the network ('optimal' routing mode), or
- The VCS neighbor zone to Unified CM must 'always' route the signaling. In effect this sets up this VCS as a gateway to Unified CM, and is the preferred option. The *Cisco Unified Communications Manager* and *Cisco Unified Communications Manager (8.6.1 or later)* zone profiles are pre-configured to 'always' route the signaling, thus no additional configuration is required providing one of these profiles is used.

Encrypted Calls to Endpoints Registered to Unified CM

Endpoints registered to Unified CM need to be configured with a "SIP Secure profile" to provide encrypted media and call negotiation. If such a profile is not available by default, it will need to be created via **System > Security > Phone Security**.

See *Securing Cisco TelePresence Products* for further information on using the Cisco CTL Client and configuring Unified CM for secure communications.

Checking Unified CM message size limit

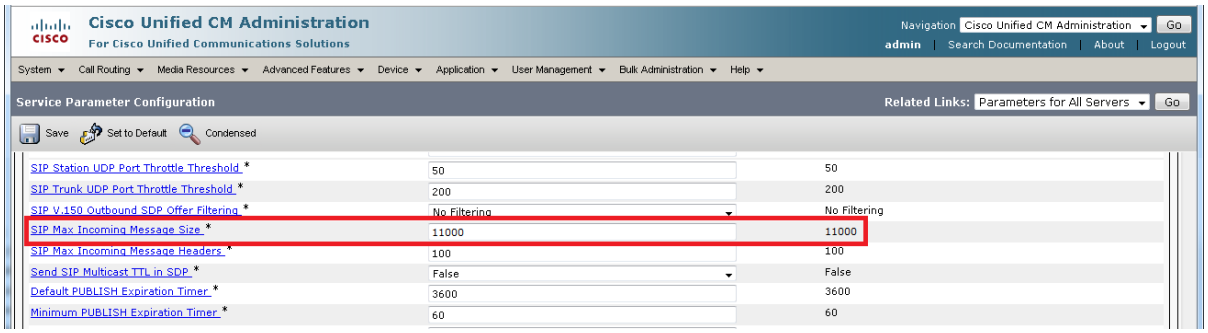
SIP messages for video are considerably larger than SIP messages for audio calls, in particular, when a Cisco TelePresence Server is used in the video network.

Ensure that the **SIP Max Incoming Message Size** on Unified CM is set to 11000:

1. Go to **System > Service Parameters**.
2. Select the appropriate server.
3. Select *Cisco CallManager (Active)* as the service.
4. Select **Advanced**.
5. In the **Clusterwide Parameters (Device – SIP)** configure the field as follows:

SIP Max Incoming Message Size	11000

6. Click **Save**.



Appendix 1: Troubleshooting

Problems connecting VCS Control local calls

Look at “Search history” to check the applied transforms

Search history entries report on any searches initiated from a SETUP/ARQ /LRQ in H323 and from an INVITE/OPTIONS in SIP.

1. Go to **Status > Search history**.
   The summary shows the source and destination call aliases, and whether the destination alias was found.
2. Select the relevant search attempt.

The search history for that search attempt shows:

- the incoming call’s details
- any transforms applied by admin or user policy or CPL
- and in priority order, zones which matched the required (transformed) destination, reporting on:
  - any transforms the zone may apply
  - found or not found status
  - if not found, the error code as seen in the zone’s search response
  - repeated until a zone is found that can accept the call, or all zone matches have been attempted (the search may be “not found” due to lack of bandwidth or because the search from the zone resulted in an H.323 rejection reason or a non 2xx response to a SIP request)

If the search indicates:

- Found: False
- Reason: 480 Temporarily Not Available

this could be because the VCS zone links are not correctly set up. From the command line execute **`xcommand DefaultLinksAdd`** to set up the required links for the VCS's default zones; also check the links for other zones that have been created.

Note that each H.323 call will have two entries in the search history:

- The first for an ARQ to see if the endpoint can be found.
- The second for the Setup to actually route the call.

The ARQ search does not depend on links or link bandwidth, and so if links do not exist or link bandwidth is insufficient it may still pass, even though the Setup search will subsequently fail.

Each SIP call will usually have only a single search history entry for the SIP INVITE.

Look at call history to check how the call progressed

1. Go to **Status > Calls > History**.
   The summary shows the source and destination call aliases, call duration and protocol (including any interworking).
2. Select the relevant call attempt and then the relevant call components.
   This shows the incoming and outgoing call leg details and the zone and subzone routing.

Check for errors

Check the Event Log which is accessible from the web browser: **Status > Logs > Event Log**.

Tracing calls

Tracing calls at SIP / H.323 level in VCS

1. Go to **Maintenance > Diagnostics > Diagnostic logging**.
2. Optionally, select **Take tcpdump while logging**.
3. Click **Start new log**.
4. (Optional) Enter some **Marker** text and click **Add marker**.
   - The marker facility can be used to add comment text to the log file before certain activities are performed. This helps to subsequently identify the relevant sections in the downloaded diagnostic log file.
   - You can add as many markers as required, at any time while the diagnostic logging is in progress.
   - Marker text is added to the log with a "`DEBUG_MARKER`" tag.
5. Reproduce the system issue you want to trace in the diagnostic log.
6. Click **Stop logging**.
7. Click **Download log** to save the diagnostic log archive to your local file system. You are prompted to save the archive (the exact wording depends on your browser).

Call failures with Cisco TelePresence Server

SIP messages from Cisco TelePresence Server can be > 5,000 bytes (which is the default **SIP Max Incoming Message Size** configured in Unified CM).

Increase the **SIP Max Incoming Message Size** – see Checking Unified CM message size limit [p.29].

In-call problems

Calls clear down when a call transfer from a video phone on Unified CM transfers a call to VCS

Even if use of a media termination point (MTP) is not requested on the SIP trunk between Unified CM and VCS, if DTMF signaling method is configured as “No preference” on the SIP trunk on Unified CM, Unified CM will try and use a Media Transfer Point and the call will fail.

To resolve this, ensure that DTMF signaling method is configured as *RFC 2833* on Unified CM on the SIP trunk from Unified CM to VCS.

Failure to join a Unified CM endpoint to a conference using Multiway

Ensure that your network is set up as described in Appendix 5: Multiway and Unified CM [p.40].

Poor video quality from Unified CM

Ensure that your Unified CM region has an appropriate session bit rate for video calls as described in Configuring the region with an appropriate session bit rate for video calls [p.11].

Taking a trace on Unified CM using RTMT

RTMT is a tool that lets you monitor system health, view graphs and collect logs from Unified CM. There are versions for both Linux and Windows. Unified CM must also be configured to specify what can be traced.

Configure Unified CM to enable tracing

1. Log in to Unified CM.
2. In the **Navigation** drop-down select **Cisco Unified Serviceability** and click **Go**.
3. Go to the **Troubleshooting Trace Settings** page (**Trace > Troubleshooting Trace Settings**).
4. Select the **Check All Services** check box.
5. Click **Save**.

Installing RTMT – Real Time Monitoring Tool

1. Log in to Unified CM using a Linux or Windows PC.
2. Go to **Application > Plugins**.
3. Select **Find** with 'Name begins with <blank>' and 'Plugin Type equals Installation'.
4. Scroll down to the entry for 'Cisco Unified CM Real-Time Monitoring Tool – Linux' or 'Cisco Unified CM Real-Time Monitoring Tool – Windows', as required.
5. Click on the **Download** link.
6. When downloaded, run the downloaded install file.
7. Follow the instructions in the install wizard.
8. When complete, click **Done** to exit the installer.

Running RTMT

1. Run RTMT. (For example, under windows this is in **Start > All Programs > Cisco > CallManager Serviceability > Real-Time Monitoring Tool**.)
2. In the Login window enter the **Host IP Address**, **User Name** and **Password**.
3. Click **OK**.

Taking a trace using RTMT

1. Select **Trace & Log Central**.
2. Double-click on **Real Time Trace**.
3. Double-click **View Real Time Data**.
4. Select a Node – the Unified CM instance that is to have the trace run on it.
5. Click **Next >**.

6. Select the following:
   - **Products** = *UCM*
   - **Services** = *Cisco CallManager*
   - **Trace File Type** = *sdi*
7. Click **Finish**.

Note:

- Logs can take a while to download.
- The sdi (System Diagnostic Interface) trace contains alarms, error information and SIP stack trace information.

Call failures

TLS calls fail when Unified CM uses SRV trunk destinations

Calls from Unified CM may fail if they use a TLS trunk security profile and SRV trunk destinations (requiring "_sips._tcp" SRV record lookups in DNS).

See bug CSCue37440 in the Cisco Bug Search Tool for up-to-date information regarding the versions of Unified CM in which this issue has been fixed.

If you need to address one or more VCS peers you can work around this problem by not using SRV records. Instead, in the SIP trunk, specify each VCS **Destination Address** individually using DNS A-records or static IP addresses. However, note that these addresses affect the domain portion of the URI received by VCS from Unified CM. You may need to set up appropriate transforms on the VCS to cater for this (see Creating a transform to convert Unified CM supplied domain information to the VCS SIP domain [p.21]).

Encrypted call failures

Calls between endpoints registered to Unified CM and endpoints registered to VCS (or proxied via VCS) will fail if the Unified CM endpoint requests best effort encryption and the other endpoint does not support encryption. (Unified CM has a proprietary method for indicating fall back to no encryption - X-cisco-srtp-fallback – which VCS currently does not support.)

Appendix 2: Known interworking capabilities and limitations

Capabilities

SIP and H.323 endpoints making basic calls

- SIP and H.323 endpoints can make calls via the VCS to endpoints registered to Unified CM.
- Endpoints registered to Unified CM can make calls to SIP and H.323 endpoints on the VCS.

Limitations

Cisco TelePresence Conductor

When VCS is configured to work with TelePresence Conductor, calls made from Unified CM over the SIP trunk may initiate or join conferences controlled by TelePresence Conductor. When there is a Cisco TelePresence System (CTS) Series endpoint registered to the Unified CM, the VCS is connected to the TelePresence Conductor's back-to-back user agent and the conference is hosted on a TelePresence Server, encrypted calls made from the CTS will drop when the SIP session is refreshed. In this scenario we recommend that you create a SIP trunk directly from the Unified CM to the TelePresence Conductor as detailed in *Optimized Conferencing for Cisco Unified Communications Manager Solution Guide*.

E20 encryption

If E20 has Encryption Mode = Best Effort then calls from Unified CM clear when E20 answers them. Set Encryption Mode = Off.

T150 running L6.0 code

If a SIP call is made from a T150 running L6.0 code to Unified CM 8.0 (and earlier), Unified CM does not handle the UPDATE message that the T150 sends immediately after the call is answered, and so on call answer the call is cleared down immediately.

If **`xConfiguration Conference H239`** is set to Off then no BFCP is offered and Unified CM handles the UPDATE from the T150 and the call completes as desired.

H.323 MXP and 9971

When an MXP registered to VCS is in a call with a 9971 registered to Unified CM and the MXP call is H.323, the video on the MXP will be CIF (small picture) rather than VGA (full size picture). (Seen on MXP F9.0 and 9971 version 9.0.2)

If the MXP call is SIP, a full size picture will be seen.

Appendix 3: Connecting Unified CM to a VCS cluster

From Unified CM version 8.5, to connect Unified CM with a cluster of VCS peers there are 2 methods of providing Unified CM with the addresses of the VCS cluster peers:

- the trunk to VCS specifies the DNS SRV address for the VCS cluster
- the trunk to VCS specifies a list of VCS peers

Prior to Unified CM 8.5, the trunk to VCS had to specify the DNS SRV address for the VCS cluster.

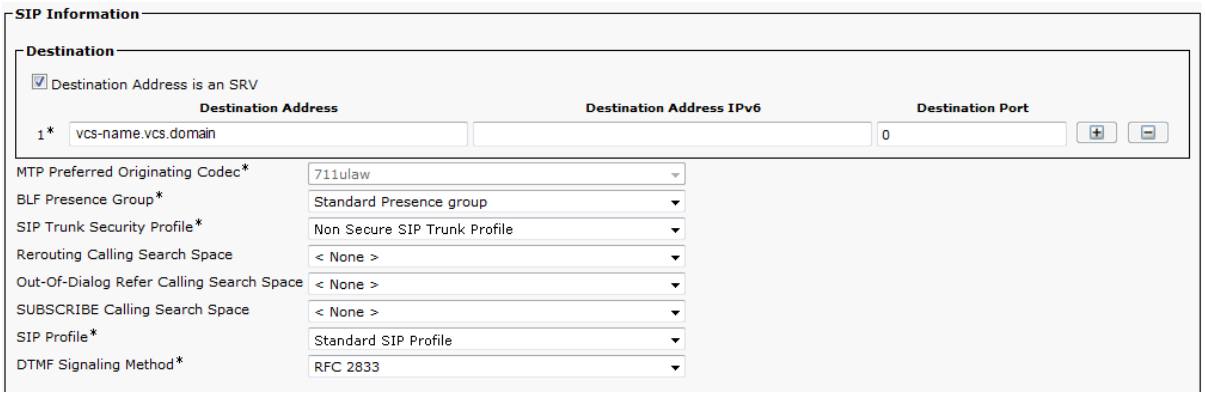
Configuring the trunk to VCS to specify the DNS SRV address for the VCS cluster

Ensure that in the DNS server used by Unified CM a DNS SRV record exists for the cluster of VCS peers; in the DNS SRV record each peer should be set with equal priority and equal weight.

1. On Unified CM, go to **Device > Trunk**.
2. Select the previously configured Trunk.
3. Scroll down and configure the **SIP Information** section fields as follows:

Destination address	<DNS SRV name of VCS cluster>
Destination address is an SRV	Select this check box.

4. Click **Save**.
5. Click **Reset**.
6. Click **Reset**.
7. On VCS, ensure that the cluster name is configured as a SIP domain (**Configuration > Domains**).



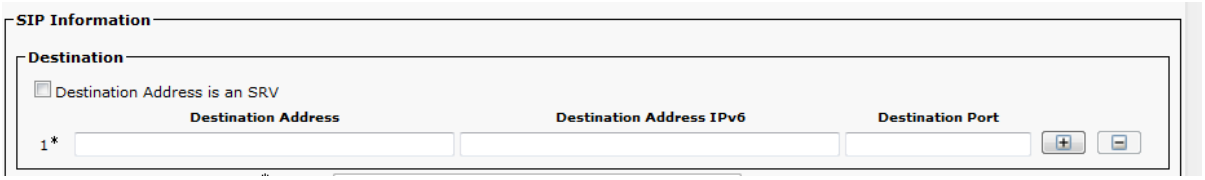
Configuring the trunk to VCS to specify a list of VCS peers

1. On Unified CM, go to **Device > Trunk**.
2. Select the previously configured Trunk.

3. Scroll down and configure the **SIP Information** fields as follows:
   (Click **+** to obtain additional destination address entries.)

Destination address is an SRV	Ensure that this check box is not selected
Destination address 1 and **Destination port 1**	IP address or DNS name of VCS peer 1 5060 or 5061 depending on connectivity (TCP/TLS)
Destination address 2 and **Destination port 2**	IP address or DNS name of VCS peer 2 5060 or 5061 depending on connectivity (TCP/TLS)
Destination address 3 and **Destination port 3**	IP address or DNS name of VCS peer 3 – if it exists 5060 or 5061 depending on connectivity (TCP/TLS)
... up to **Destination address 6** and **Destination port 6**	... repeat up to IP address or DNS name of VCS peer 6 – where they exist

4. Click **Save**.
5. Click **Reset**.
6. Click **Reset**.



Appendix 4: Connecting VCS to a cluster of Unified CM nodes

When connecting VCS to a cluster of Unified CM nodes, VCS needs to be able to route calls to each Unified CM node.

This can be done in 2 ways, in order of preference:

1. With a single neighbor zone in VCS with the Unified CM nodes listed as location peer addresses. This option is only available from VCS X7.0 or later.
2. By using DNS SRV records and a VCS DNS zone.

Note that both options ensure that the VCS to Unified CM call load is shared across Unified CM nodes.

Option 1: Using a single neighbor zone

Unified CM configuration

When in a cluster, Unified CM needs to accept calls routed to number@domain (instead of number@<ip address of Unified CM>) so that VCS can send the call to any Unified CM node without having to make sure that the domain portion matches the IP address of the node that the call is being sent to.

Ensure that the **Cluster Fully Qualified Domain Name** (**System > Enterprise parameters**, in the **Clusterwide Domain Configuration** section) is set to the same domain as the video network, for example vcs.domain.

VCS Control configuration

The VCS configuration requires an update to the neighbor zone:

1. Go to **Configuration > Zones**.
2. Select the Unified CM neighbor zone.
3. Configure the fields as follows:

Peer 1 address	IP address of Unified CM node 1, or the domain of Unified CM node 1.
Peer 2 address	IP address or the domain of Unified CM node 2.
Peer 3 address	IP address or the domain of Unified CM node 3, or blank if no Unified CM node 3.
.... up to **Peer 6 address**	... repeat up to the IP address or the domain of Unified CM node 6, or leaving it blank if there is no Unified CM node.

Option 2: Using a DNS zone

Unified CM configuration

Ensure that the **Cluster Fully Qualified Domain Name** (**System > Enterprise parameters**, in the **Clusterwide Domain Configuration** section) is set to the same domain as the video network, for example vcs.domain.

DNS server configuration

Configure the DNS server (that is used by the VCS) with DNS SRV records for the Unified CM cluster.

- _sips._tcp.fqdn_of_cucm_clusterrecords for TLS connectivity (one record for each Unified CM node); or
- _sip._tcp.fqdn_of_cucm_clusterrecords for TCP connectivity (one record for each Unified CM node)

VCS Control configuration

VCS configuration requires 3 steps:

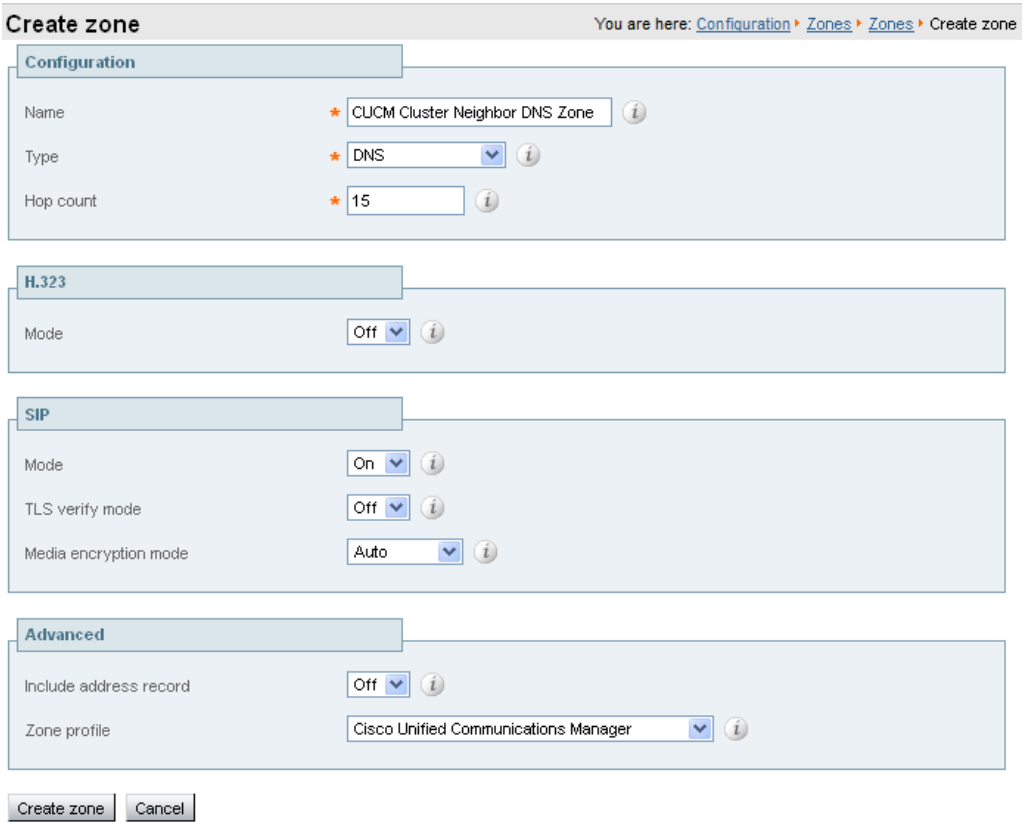
- Create a Unified CM DNS zone
- Adjust search rule to use the DNS zone
- Delete the old Unified CM neighbor zone

Creating a Unified CM DNS zone

1. Go to **Configuration > Zones > Zones**.
2. Click **New**.
3. Configure the fields as follows:

Name	CUCM Cluster Neighbor DNS Zone
Type	*DNS*
Hop count	15
H.323 mode	*Off* H.323 access is not required for communication with Unified CM
SIP mode	*On*
TLS verify mode	*Off*
Media encryption mode	*Auto*
Include address record	*Off*
Zone profile	Select *Cisco Unified Communications Manager* or *Cisco Unified Communications Manager (8.6.1 or later)* as appropriate.

4. Click **Create zone**.



Adjusting the search rule

Change the search rule to point to this Unified CM DNS zone.

1. Go to **Configuration > Dial plan > Search rules**.
2. Select the existing "Route to Unified CM" search rule.
3. Update the **Target zone** to use the *CUCM Cluster Neighbor DNS Zone* created above.
4. Click **Save**.

Deleting the old Unified CM neighbor zone

Delete the now unused neighbor zone "Unified CM Neighbor".

1. Go to **Configuration > Zones > Zones**.
2. Select the check box next to the "CUCM Neighbor" zone.
3. Click **Delete**.

Appendix 5: Multiway and Unified CM

To enable Unified CM registered endpoints to be joined into a Multiway conference, ensure that:

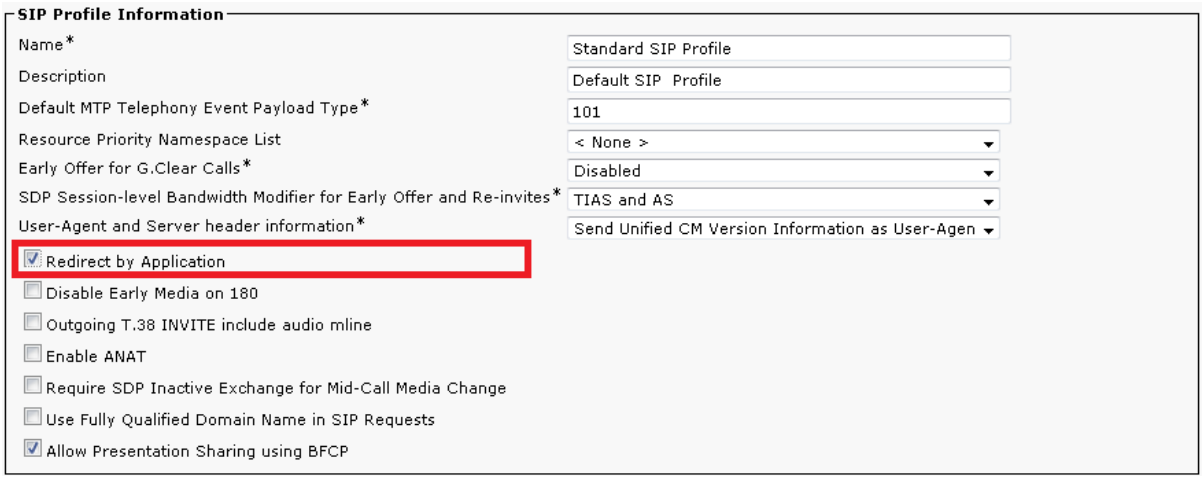
1. The VCS zone towards Unified CM uses a zone profile with **Call Signaling Routed Mode** set to *Always*.
2. Unified CM has Route Patterns that route calls for the Multiway alias domain to VCS.
3. Unified CM has **Redirect by Application** selected in the SIP profile used by the SIP trunk to VCS.

VCS configuration

The VCS zone towards Unified CM must use a zone profile with **Call Signaling Routed Mode** set to *Always*. To do this, ensure that the zone profile is set to either *Cisco Unified Communications Manager* or *Cisco Unified Communications Manager (8.6.1 or later)* as appropriate.

Unified CM configuration

1. Ensure that Unified CM has Route Patterns that route calls for a video domain to VCS. Follow the instructions in Enabling endpoints registered on Unified CM to call endpoints registered on VCS [p.21].
2. In the SIP profile used by the SIP trunk to VCS, ensure that **Redirect by Application** is selected:
   a. Go to **Device > Device Settings > SIP Profile**.
   b. Select the check box by **Redirect by Application**.
   c. Click **Apply Config** and click **OK**.



You can now test Multiway.

Appendix 6: Additional information

IP address dialing

Unified CM cannot dial out to IP addresses, but the VCS can. To support IP address dialing from endpoints registered to Unified CM, we recommend following the procedure in the knowledge base article *Dial IP Addresses from Endpoints Registered to CUCM with VCS/Expressway*.

Characters allowed in SIP URIs

The following character set is allowed in SIP URIs (further details may be found in RFC 3261):

a-z / A-Z / 0-9 / "-" / "_" / "." / "!" / "~" / "*" / "'"/ "(" / ")" "&" / "=" / "+" / "$" / "," / ";" / "?" / "/"

If other characters are needed they must be “escaped” using "%" followed by a pair of hexadecimal digits that represents the ASCII value for the required character.

For example, "alice smith@example.com" must be encoded as alice%20smith@example.com (where %20 represents the space character).

Document revision history

The following table summarizes the changes that have been applied to this document.

Date	Description
July 2015	Updated for X8.6.
April 2015	Updated for X8.5.2. Link to new IP address dialing article.
December 2014	Updated for X8.5. IP address dialing information modified.
June 2014	Republished for X8.2.
December 2013	Updated for X8.1. Included support for new *Cisco Unified Communications Manager (8.6.1 or later)* zone profile and hence removed the Unified CM zone profile parameters and Enabling BFCP appendices as Custom profiles are no longer required.
March 2013	Simplified the example dial plan. Moved the requirement to configure the Unified CM Cluster Fully Qualified Domain Name to be part of the main configuration, and not just required for clusters. Moved TLS connectivity instructions into the document body and added troubleshooting note about Unified CM DNS SRV lookups.
September 2012	Revised Appendix 7 and Appendix 8 for use of encrypted SRTCP. Document now only refers to Unified CM 8.x and 9.x.
August 2012	Updated for VCS X7.2.
March 2012	Updated for VCS X7.1.
October 2011	Updated guidance on configuration for Multiway (Call Signaling Routed Mode).
August 2011	Updated for VCS X7.0 and BFCP. Updated guidance on connecting VCS to a cluster of Unified CM nodes.
June 2011	Updated for VCS X6.
March 2011	Updates to Appendix 5 for Unified CM version 8.5 regarding connecting Unified CM to a cluster of VCS peers. Added Appendix 12 – Characters allowed in SIP URIs.
October 2010	Additions for Clustered Unified CMs. Additions to handle returning call to Unified CM callback URI. Updates to handle call transfer.
July 2010	Document title updated to refer to Cisco Unified Communications Manager. Added this document revision history table. General updates applied to reflect user interface differences in Unified CM v8.
June 2010	Added Appendix 10 - Connecting VCS to Unified CM using TLS.
April 2010	Updated for VCS X5.1. Added additional troubleshooting information.
January 2010	Initial release.