

This document details the process to configure Cisco Unified Communications Manager (UC Manager to be short - aka CallManager) with Cisco Unified Border Element (CUBE to be short - aka IP-to-IP Gateway) to support SIP trunking to Verizon VoIP Services.

CUBE is configured to support H.323 to SIP interworking for the following reasons:

1. Support of G.729 through the SIP trunk to Verizon WITH SIP Early Offer support. Please note that UC Manager 5.0 and later does support SIP trunking with G.729; however, when Early Offer is required, UC Manager can only support G.711 (μ - or a-law), and Early Offer is the VzB defacto standard.
2. Eliminates the use of DSPs on routers for transcoding to reduce cost.

Tested elements:

- + UC Manager: 5.1(1a)
- + CUBE: Cisco 3825 12.4(11)T Integrated Voice/Video, IPIPGW, TDMIP GW AES Feature Set. Prior to October 2007, CUBE was only supported on Integrated Voice/Video, IPIPGW, TDMIP, GW images. These images have "ivs" in the image names (e.g., c2800nm-ipvoice_ivs-mz.124-15.T1.bin). As of October 2007, CUBE is supported on other IOS feature sets, starting with IP Voice.

Step 1. Required media resources. UC Manager uses media resources such as Music on Hold servers, conference bridges, transcoders, and media termination points. The use of most of these resources are self evident; some are not. Please refer to the following link for more detailed information on the use of these resources.

http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_implementation_design_guide_chapter09186a0080637427.html

In order to provide SIP Early Offer to Verizon's network with the use of the H.323 protocol, H.323 FastStart must be configured on UC Manager, and this requires a Media Termination Point (MTP). An MTP is an entity that accepts two full duplex RTP streams. It bridges the media streams together and allows them to be set up and torn down independently. The MTP resource is invoked by the H.323 Gateway whenever a call is routed to it (via the route pattern). UC Manager can act as the MTP if only G.711 codec is required.

A hardware (DSP-based) conferencing resource may also be needed. UC Manager can act as a conferencing resource (co-resident) but only G.711 is supported. Since we are using G.729, a hardware resource (i.e., IOS DSPs) must be available whenever a UC Manager phone invokes conferencing.

If music-on-hold is desired, a music resource must be available. More than likely, if the customer wanted MOH, this resource is already configured. This resource is invoked by the phone when the user places the other party on HOLD. A tone is used to indicate a call is on hold if MOH is not configured.

Step 1a. Media Termination Point.

As stated above, MTPs are required for H.323 FastStart. This resource does not have to be DSP-based and can be just Cisco IOS. To make a Cisco IOS MTP resource be available for UC Manager, the resource must be defined in IOS and register to UC Manager.

Define the MTP in IOS.


```
voice-card 0
 dspfarm
 dsp services dspfarm
```


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```
!enables onboard DSPs to act as DSP services. These resources can be used for
conference bridges, transcoding, and voice termination.
!
sccp local GigabitEthernet0/0 !gi0/0 must have IP connectivity to UC Manager
sccp ccm 192.168.0.6 identifier 2 priority 2 version 5.0.1
sccp ccm 192.168.0.4 identifier 5 priority 1 version 5.0.1
!match UC Manager vers as close as possible to what is deployed
!each UC Manager server in the CallManager Group should be configured here
!identifier is locally significant
!priority should match the CallManager Group priority
sccp
! this single "sccp" command enables SCCP protocol on this router
!
sccp ccm group 10
  associate ccm 5 priority 1
  associate ccm 2 priority 2
  associate profile 10 register RTP001193484810 !MTP resource
  associate profile 12 register CON001193484810 !Conference resource
!The value immediately following "register" is what the "name" that IOS uses to
register this resource to UC Manager. It MUST match (case-sensitive) to what is
defined in UC Manager.
!
dspfarm profile 12 conference
  description conference bridge
  codec g711ulaw
  codec g711alaw
  codec g729ar8
  codec g729abr8
  codec g729r8
  codec g729br8
  maximum sessions 10
  associate application SCCP
!
dspfarm profile 10 mtp
  codec g729r8 !codec must match codec desired/negotiated for voice call
  maximum sessions software 12
!Notice "software" in this command. This means that this MTP resource is IOS software-
based, not DSPs. IOS will allow you to configure up to 500 sessions; however, you need
to consider the practical limits of the IOS hardware platform (i.e., CPU and memory).
  associate application SCCP
```

Now, define the IOS MTP resource in UC Manager.

Media Termination Point Configuration



Status
 Status: Ready

Media Termination Point Information

Registration	Registered with Cisco Unified CallManager 192.168.0.4
IP Address	192.168.0.10
Media Termination Point Type*	Cisco IOS Enhanced Software Media Termination Point
Media Termination Point Name*	<input type="text" value="RTP001193484810"/>
Description	<input type="text" value="IOS SW-based MTP H.323 FastStart"/>
Device Pool*	<input type="text" value="Default"/>

Step 1b. Hardware-based (DSP) Conference Bridges.

A DSP-based conference resource will be needed if you need to support conferencing with non-G.711 codec parties.

Configuring the conference bridge on a Cisco router with PVDM2 DSPs is detailed above in Step 1a. However, if using older DSPs (i.e., NM-HDV which uses PVDMs), then refer to the following link.

http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_configuration_example09186a0080334294.shtml#conf

Define a conference bridge in UC Manager.

Conference Bridge Configuration Relate

Status
Status: Ready

Conference Bridge Information
Conference Bridge : CON001193484810 (Cisco 3845 DSP CONF Bridge)
Registration Registered with Cisco Unified CallManager 192.168.0.4
IP Address 192.168.0.10

IOS Conference Bridge Info
Conference Bridge Type* Cisco IOS Enhanced Conference Bridge
Conference Bridge Name* CON001193484810
Description Cisco 3845 DSP CONF Bridge
Device Pool* Default
Location* Hub_None

Save Delete Copy Reset Add New

Step1c. Verify that the MTP and conference resources registered to UC Manager.

From the router's perspective, use `show sccp`. Below is a sample. Note that the operation state is **ACTIVE** and the UC Managers that it is using. The protocol between UC Manager and the router is **SCCP**. If **SCCP** is disrupted between the router and UC Manager, it is recommended that **SCCP** be reset on the router (i.e., use `no sccp` then `sccp` in the configuration mode).

```
cube#sho sccp
SCCP Admin State: UP
Gateway IP Address: 192.168.0.10, Port Number: 2000
IP Precedence: 5
User Masked Codec list: None
Call Manager: 192.168.0.4, Port Number: 2000
                Priority: 1, Version: 5.0.1, Identifier: 5
Call Manager: 192.168.0.6, Port Number: 2000
                Priority: 2, Version: 5.0.1, Identifier: 2

Software MTP Oper State: ACTIVE - Cause Code: NONE
Active Call Manager: 192.168.0.4, Port Number: 2000
TCP Link Status: CONNECTED, Profile Identifier: 10
Reported Max Streams: 24, Reported Max OOS Streams: 0
```

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!24 streams are double the configured maximum sessions. Two MTP resources will be allocated per call. However, only configure maximum sessions to correlate with the number of calls needed.

Supported Codec: g729r8, Maximum Packetization Period: 60
Supported Codec: rfc2833 dtmf, Maximum Packetization Period: 30
Supported Codec: rfc2833 pass-thru, Maximum Packetization Period: 30
Supported Codec: inband-dtmf to rfc2833 conversion, Maximum Packetization Period: 30

Conferencing Oper State: **ACTIVE** - Cause Code: NONE
Active Call Manager: 192.168.0.4, Port Number: 2000
TCP Link Status: CONNECTED, Profile Identifier: 12
Reported Max Streams: 80, Reported Max OOS Streams: 0
Supported Codec: g711ulaw, Maximum Packetization Period: 30
Supported Codec: g711alaw, Maximum Packetization Period: 30
Supported Codec: g729ar8, Maximum Packetization Period: 60
Supported Codec: g729abr8, Maximum Packetization Period: 60
Supported Codec: g729r8, Maximum Packetization Period: 60
Supported Codec: g729br8, Maximum Packetization Period: 60
Supported Codec: rfc2833 dtmf, Maximum Packetization Period: 30
Supported Codec: rfc2833 pass-thru, Maximum Packetization Period: 30
Supported Codec: inband-dtmf to rfc2833 conversion, Maximum Packetization Period: 30

To verify that UC Manager has allocated a resource, use `show sccp connection`.

```
cube#sho sccp conn !note the resource type allocated and the codec used
sess_id  conn_id  stype mode   codec  dtmf_method  ripaddr      rport sport
33556460 33554517  mtp  sendrecv g729    rfc2833_pt thru  192.168.0.105 16384 17750
33556460 33554516  mtp  sendrecv g729    rfc2833_pt thru  192.168.0.10  17744 19064
```

```
cube#sho sccp conn !note the resource type allocated and the codec used
sess_id  conn_id  stype mode   codec  dtmf_method  ripaddr      rport sport
33556462 33554528  conf sendrecv g711u    none         192.168.0.105 16384 16794
33556462 33554527  conf sendrecv g729    none         192.168.0.10  18730 19344
33556462 33554525  conf sendrecv g729    none         192.168.0.10  17806 17156
```

Step 2. Media Resource Group (MRG) and Media Resource Group List (MRGL). In order to use media resources, they must be placed into an MRG. MRGs are placed into MRGLs. An MRGL must be associated with a device, such as a phone, gateway or trunk in UC Manager, so that that entity can use the resources configured and that are available. Please refer to the following link for more detailed information on MRG and MRGLs.

http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_implementation_design_guide_chapter09186a0080637427.html#wp1094998

As stated above in Step 1, the MTP resource is invoked by the H.323 gateway whereas the MOH and conference resources are invoked by the phone. Thus, we need to define two MRGs and place them into two MRGLs. Specifically, the MTP and its associated MRG and MRGL should be accessible by the H.323 gateway; and, the conference bridge and MOH resource and their associated MRGs should be placed in the MRGL for the UC Manager phones.

If there are multiple MTP resources, there are two ways to group the resources to control how they are used.

- Place all the resource (e.g., IOS software MTP, UC Manager MTP) into the same MRG. UC Manager will allocate the capacity of these resources in a round-robin fashion.
- Each resource is placed into its own MRG. In the case, UC Manager will use a resource until its capacity is reached before moving to the next resource.

The grouping of resources into MRGs/MRGLs are also used to control how the resources should be allocated depending on location. Please refer to the link above for more information.

Configure the MRG and MRGL for the H.323 gateway.

Media Resource Group Configuration

Status
Status: Ready

Media Resource Group Status
Media Resource Group: MTP_MRG (used by 2 devices)

Media Resource Group Information

Name* MTP_MRG
Description IOS SW-based MTP

Devices for this Group

Available Media Resources**
ANN_2
ANN_3
CFB_2
CFB_3
CON001193484810






Selected Media Resources*
RTP001193484810 (MTP)


Use Multicast for MOH Audio (If at least one multicast MOH resource is available)

Save Delete Copy Reset Add New

The MRG must be placed into an MRGL.

Media Resource Group List Configuration

Status
 Status: Ready

Media Resource Group List Status
Media Resource Group List: H323_GW_MRGL (used by 2 devices)

Media Resource Group List Information
Name*

Media Resource Groups for this List

Available Media Resource Groups

▼ ▲

Selected Media Resource Groups

▼ ▲

Then, the final step is to associate the MRGL to the devices so that the devices can invoke the resource when needed.

Gateway Configuration

Status
 Status: Ready

Device Information

Product	H.323 Gateway
Device Protocol	H.225
Registration	Unknown
IP Address	192.168.0.10
Device Name*	<input type="text" value="192.168.0.10"/>
Description	<input type="text" value="Cisco 3825"/>
Device Pool*	<input type="text" value="Default"/> ▼
Call Classification*	<input type="text" value="OffNet"/> ▼
Media Resource Group List	<input type="text" value="H323_GW_MRGL"/> ▼
Packet Capture Mode*	<input type="text" value="None"/> ▼
Packet Capture Duration	<input type="text" value="0"/>
Location*	<input type="text" value="Hub_None"/> ▼
AAR Group	<input type="text" value="< None >"/> ▼
Tunneled Protocol*	<input type="text" value="None"/> ▼
Signaling Port*	<input type="text" value="1720"/>

Media Termination Point Required
 Retry Video Call As Audio

Similarly, repeat these steps for the conference and MOH resources and make these resources available to the phone.

Conference Bridge Configuration Relate

Status
 Status: Ready






Conference Bridge Information


Conference Bridge : CON001193484810 (Cisco 3845 DSP CONF Bridge)
 Registration Registered with Cisco Unified CallManager 192.168.0.4
 IP Address 192.168.0.10

IOS Conference Bridge Info

Conference Bridge Type* Cisco IOS Enhanced Conference Bridge
 Conference Bridge Name*
 Description
 Device Pool* ▼
 Location* ▼

Media Resource Group Configuration

Status
 Status: Ready

Media Resource Group Status
Media Resource Group: CON_MRG (used by 1 devices)






Media Resource Group Information
Name*
Description


Devices for this Group
Available Media Resources**

Selected Media Resources*

Use Multicast for MOH Audio (If at least one multicast MOH resource is available)

Media Resource Group List Configuration

Status
 Status: Ready

Media Resource Group List Status
Media Resource Group List: PHONE_MRGL (used by 1 devices)

Media Resource Group List Information
Name*

Media Resource Groups for this List
Available Media Resource Groups

Selected Media Resource Groups

The screenshot displays the 'Phone Configuration' interface. At the top, there are navigation icons and a 'Related Links: Bac' button. Below this is a 'Status' section showing 'Status: Ready'. The main area is divided into two columns. The left column, 'Association Information', includes a 'Modify Button Items' button and a list of lines: 1 Line [1] - 1016 (no partition), 2 Add a new SD, 3 Add a new SD, 4 Add a new SD, 5 Add a new SD, 6 Line [2] - Add a new DN, 7 Add a new SD, 8 Privacy, and 9 None. The right column, 'Device Information', shows 'Product Type: Cisco 7905' and 'Device Protocol: SCCP'. Below this, a table lists various settings: Registration (Registered with Cisco Unified CallManager 192.168), IP Address (192.168.0.105), MAC Address* (000F8F4A27BD), Description (SEP000F8F4A27BD), Device Pool* (Default), Phone Button Template* (Standard 7905 SCCP), Softkey Template (Standard User), Common Phone Profile* (Standard Common Phone Profile), Calling Search Space (< None >), AAR Calling Search Space (< None >), Media Resource Group List (PHONE_MRGL), User Hold MOH Audio Source (1-SampleAudioSource), and Network Hold MOH Audio Source (1-SampleAudioSource).

Step 3. Define an H.323 Gateway and Select Codec. As stated in the beginning, to achieve G.729 H.323 must be used by UC Manager. Thus, define an H.323 Gateway. The following must be configured on the Gateway

- The Device Name is the IP address of the router interface that has been “binded” to source H.323. Please refer to the router configuration for more details.
- MRGL must use the previously-configured MRGL with the MTP.
- “MTP Termination Point Required” must be checked, which is required for H.323 FastStart.
- “Wait for Far End H.245 Terminal Capability Set” must be unchecked. This allows media location information to be sent in the initial call set up messages.
- “Enable Inbound FastStart” and “Enable Outbound FastStart” must be checked.
- “Codec For Outbound FastStart” should be set to “G729”.

We will not discuss in detail how UC Manager implement Call Admission Control (CAC). However, it is important to note that UC Manager uses two methods for CAC.

- Topogy un-aware method: Use of locations and regions
- Topology aware method: Use of Resource Reservation Protocol (RSVP)

Please refer to the following link in the Solution Reference Design Guide (SRND) for more details. http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_implementation_design_guide_chapter09186a00806375f0.html

It is relevant to discuss locations and regions briefly here. Devices, gateways, and trunks are placed into “locations”, and locations are defined with audio and video bandwidth limits. Regions are used to define the codec that should be used between locations. The combination of the two controls how many calls can traverse between locations. CAC is critical when remote sites exist and IP trunks are used.

When defining an H.323 gateway, it is recommended that it be placed in a different location and device pool from the phones that will use it to make offnet calls. [Regions are placed into Device Pools and Device Pools are associated with each device.]

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Gateway Configuration

Status
i Status: Ready

Device Information

Product	H.323 Gateway
Device Protocol	H.225
Registration	Unknown
IP Address	192.168.0.10
Device Name*	<input type="text" value="192.168.0.10"/>
Description	<input type="text" value="Cisco 3825"/>
Device Pool*	<input type="text" value="Default"/>
Call Classification*	<input type="text" value="OffNet"/>
Media Resource Group List	<input type="text" value="H323_GW_MRGL"/>
Packet Capture Mode*	<input type="text" value="None"/>
Packet Capture Duration	<input type="text" value="0"/>
Location*	<input type="text" value="Hub_None"/>
AAR Group	<input type="text" value="< None >"/>
Tunneled Protocol*	<input type="text" value="None"/>
Signaling Port*	<input type="text" value="1720"/>

Media Termination Point Required
 Retry Video Call As Audio
 Wait for Far End H.245 Terminal Capability Set
 Path Replacement Support
 Transmit UTF-8 for Calling Party Name
 SRTP Allowed - When this flag is checked, IPSec needs to be configured in the network to provide end to end information.

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain
 MLPP Indication Not available on this device
 MLPP Preemption Not available on this device

Call Routing Information - Inbound Calls

Significant Digits*
 Calling Search Space
 AAR Calling Search Space
 Prefix DN
 Redirecting Number IE Delivery - Inbound
 Enable Inbound FastStart

Call Routing Information - Outbound Calls

Calling Party Selection*
 Calling Party Presentation*
 Called party IE number type unknown*
 Calling party IE number type unknown*
 Called Numbering Plan*
 Calling Numbering Plan*
 Caller ID DN
 Display IE Delivery
 Redirecting Number IE Delivery - Outbound
 Enable Outbound FastStart
 Codec For Outbound FastStart




Save
Delete
Copy
Reset
Add New


Step 4. Routing Calls. Configure a Route Pattern that would point to the H.323 Gateway for outbound calls.

A router pattern can point directly to the gateway or a route list. A route list is used if you have multiple gateways.

Step 4a. Route Pattern points to a specific Gateway.

Route Pattern Configuration

Status
 Update successful

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan*

Route Filter

MLPP Precedence*

Gateway/Route List*

Route Option
 Route this pattern
 Block this pattern

Call Classification*

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority
 Require Forced Authorization Code

Authorization Level*

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Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service
<input type="text" value="-- Not Selected --"/>	<input type="text" value="< Not Exist >"/>	<input type="text"/>

Call treatments such as discard digits and transform masks may change depending how VzB present dialed digits to CUBE/UC Manager and what the customer internal dial plan is. Leave this to the customer after discussing the dial plan with the customer.

Step 4b. Route Pattern pointed to a Route List.




When multiple gateways/trunks are available to route calls offnet, route groups and route lists are used to prioritize and control the usage of these resources. Individual resources are placed into route groups and route groups are placed into route lists.

Please refer to the following link for more detailed information on Route Groups and Route Lists.
http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_administration_guide_chapter09186a00808a8797.html#wp1045311

Configure a Route Group and place the H.323 Gateways into the Route Group.

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Route Group Configuration

Route Group Information

Route Group Name*

Distribution Algorithm*

Route Group Member Information

Find Devices to Add to Route Group

Device Name contains

**

192.168.0.10
192.168.0.11

Port(s)

Current Route Group Members



Selected Devices***

192.168.0.10 (All Ports)
192.168.0.11 (All Ports)

Removed Devices****

--

Route Group Members

 192.168.0.10 <small>H.323</small>
 192.168.0.11 <small>H.323</small>

Then, define a Route List and place the Route Group into the Route List.

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Route List Configuration

Status

Status: Ready

Route List Information

Name*

Description

Cisco Unified CallManager Group* ▼

Enable this Route List (change effective on Save; no reset required)

Route List Member Information

Selected Groups** ▼

▲

Removed Groups***

Route List Details

[H323_gateways](#)

Once the route list is defined, make sure it has registered to UC Manager; otherwise, it cannot be used.

Find and List Route Lists

Status

1 records found

Search Options

FindRoute List where ▼ ▼ Search Within Results

(device.name begins with any)

Search Results

Name	Description	Enabled	Status
<input type="checkbox"/> H323_gateways		true	Registered with 192.168.0.4

Rows per Page ▼

Point the route pattern to the Route List.

Route Pattern Configuration

Status
i Status: Ready

Pattern Definition
 Route Pattern*
 Route Partition
 Description
 Numbering Plan*
 Route Filter
 MLPP Precedence*
 Gateway/Route List* (Edit) Find
 Route Option
 Route this pattern
 Block this pattern
 Call Classification*
 Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority
 Require Forced Authorization Code
 Authorization Level*

Step 5. Configure CUBE.

Configure CUBE to support H.323 to SIP internetworking.

- **voice service voip (Required):** Allows the router to route voip to voip calls.

```
voice service voip
  allow-connections h323 to sip
  allow-connections sip to h323
```

- **dial-peers (Required):** Define the targets (i.e., UC Manager and the SIP Service Provider)

```
dial-peer voice 100 voip
  description voip dial peer to Verizon
  destination-pattern .T
  voice-class codec 1
  session protocol sipv2
  session target sip-server
!session target can be an IPv4 address or a DNS Fully Qualified Domain Name
(FQDN). In this example, we are using an alias that is defined in sip-ua (seen
later).
  incoming called-number 1...
  dtmf-relay rtp-nte digit-drop
!"digit-drop" is added to avoid sending both in-band and out-of-band tones to
the outgoing call leg when sending CUBE calls in-band via rtp-nte to out-of-
band via h245-alphanumeric.
  no vad
!
dial-peer voice 200 voip      !need one dial-peer per UC Manager server
!Dial-peers to UC Manager are H.323, not sipv2. H.323 is the default protocol.
  description dial-peer to UC Manager
  preference 2
```

```
!"preference" provides preferred matching order for similarly configured dial-
peers. The preferred order should match the CallManager Group configuration.
destination-pattern 1...
voice-class codec 1
session target ipv4:192.168.0.4
incoming called-number .
dtmf-relay h245-alphanumeric
!
dial-peer voice 205 voip      !second dial-peer to UC Manager cluster
description dial-peer to UC Manager
preference 5
destination-pattern 1...
voice-class codec 1
session target ipv4:192.168.0.6
incoming called-number .
dtmf-relay h245-alphanumeric
```

- **sip-ua (optional):** Allows customization of the SIP methods (e.g., retransmissions, target, etc.).

```
sip-ua
retry invite 2
retry bye 2
retry cancel 2
retry options 0
sip-server dns:abc.com
g729-annexb override
!In 12.4T, Cisco implemented a strict compliance to G.729 IETF specifications.
However, with this implementation, there is no backwards compatibility to
devices that are not in strict compliance. Therefore, add this command to
implement IOS behavior in the 12.3T IOS train.
```

- **call admission control (optional):** CAC can be configured to limit the concurrent number of calls supported on a single platform using total calls, CPU utilization, memory utilization, IP call capacity with an H.323 gatekeeper, and maximum connections. More than one element may be used to provide a more comprehensive CAC result. For example, total calls with CPU and memory utilization can be used together so that the router is not overwhelmed. Thus, either the total concurrent calls or high CPU or high memory utilization would trigger the router to reject the call request.

Please refer to the following link for more details on each.

http://www.cisco.com/en/US/products/sw/iosswrel/ps1839/products_feature_guide09186a00800e0d4b.html#wp1127100

The goal is to push the router to its limits and not cause a flapping situation.

```
call threshold global total-calls low 10 high 12
!This is an optional component to control the number of calls to be supported
on the router/CUBE. This particular command tracks the number of calls,
rejecting the 12th call and not accepting calls again until the total number of
calls falls below 10.
call threshold global cpu-avg low 68 high 75
call threshold global total-mem low 75 high 80
call treatment cause-code no-resource
!This correlates (by default) to a 503 Service Unavailable message being sent
when calls are rejected.
call treatment on
!Enables call admission control
!
```


- **H.225 timers (optional):** H.225 is the signaling control protocol in the H.323 protocol suite. Cisco IOS allows custom configuration of some timers to allow faster communications to another H.323 endpoint (UC Manager, in our case). This is highly recommended when there are more than one UC Manager server.

As previously discussed, a dial-peer is required for each UC Manager server when the targets are defined with IPv4 addresses. With the exception of the IPv4 address, all other aspects of the dial-peer configuration is exactly the same; therefore, to set a preference on which UC Manager server to send a call to, IOS uses the `preference` command.

There are two different types of timers that are used to redirect a call if the preferred UC Manager server is unavailable. In order to get the desired behavior of going from one UC Manager to another (i.e., one dial-peer to another), the H.225 timer should be adjusted (from the default).

More details are here:

http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_configuration_example_09186a0080094852.shtml

```
voice class h323 1
!create a voice class to be able to apply to multiple dial-peers
  h225 timeout tcp establish 3
!when this timer is set to three seconds, the router attempts a connection to
the primary UC Manager server. If it does not receive a response in three
seconds, it falls back to the secondary UC Manager server.
  h225 timeout setup 3
!this is the timeout value for a response for an outgoing SETUP message.
!
dial-peer voice 205 voip
  description dial-peer to UC Manager
  preference 5
  destination-pattern 1...
  voice-class codec 1
  session target ipv4:192.168.0.6
  incoming called-number .
  dtmf-relay h245-alphanumeric
  voice-class h323 1      !apply the voice class to the dial-peer
!
dial-peer voice 200 voip
  description dial-peer to UC Manager
  preference 2
  destination-pattern 1...
  voice-class codec 1
  session target ipv4:192.168.0.4
  incoming called-number .
  dtmf-relay h245-alphanumeric
  voice-class h323 1      !apply the voice class to the dial-peer
```

A full sample configuration is below. This configuration has optional components:

- **The router providing DHCP server services**
- **The router providing MTP and Conference bridge services**

Building configuration...

```
Current configuration : 3743 bytes
!
version 12.4
```

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```
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
service sequence-numbers
!
hostname cube
!
boot-start-marker
boot-end-marker
!
logging buffered 200000
no logging console
enable password xxxx
!
no aaa new-model
no network-clock-participate slot 2
no network-clock-participate wic 0
no ip dhcp use vrf connected
ip dhcp excluded-address 192.168.0.0 192.168.0.100
!
ip dhcp pool IPPHONES
    network 192.168.0.0 255.255.255.0
    default-router 192.168.0.10
    option 150 ip 192.168.0.6
!
ip cef
!
no ip domain lookup
ip name-server 166.34.87.108
ip name-server 166.34.87.49
!
multilink bundle-name authenticated
!
voice-card 0
    dspfarm
    dsp services dspfarm
!
voice-card 2
    dspfarm
!
voice service voip
    allow-connections h323 to sip
    allow-connections sip to h323
    h323
    sip
        bind control source-interface GigabitEthernet0/1
        bind media source-interface GigabitEthernet0/1
!
voice class codec 1
    codec preference 1 g729r8
    codec preference 2 g711ulaw
    codec preference 3 g711alaw
!
voice class h323 1
    h225 timeout tcp establish 3
    h225 timeout setup 3
!
!
interface GigabitEthernet0/0
    description CUBE inside interface
    ip address 192.168.0.10 255.255.255.0
    ip virtual-reassembly
    load-interval 30
```

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```
duplex half
speed auto
media-type rj45
no keepalive
h323-gateway voip bind srcaddr 192.168.0.10
!
interface GigabitEthernet0/1
description CUBE outside interface
ip address 166.38.98.31 255.255.255.0
ip virtual-reassembly
load-interval 600
duplex auto
speed auto
media-type rj45
no keepalive
!
ip route 0.0.0.0 0.0.0.0 166.38.98.1
!
ip http server
no ip http secure-server
!
control-plane
!
call threshold global total-calls low 9 high 12
call threshold global cpu-avg low 68 high 75
call threshold global total-mem low 75 high 80
call treatment cause-code no-resource
call treatment on
!
sccp local GigabitEthernet0/0
sccp ccm 192.168.0.4 identifier 5 priority 1 version 5.0.1
sccp ccm 192.168.0.6 identifier 2 priority 2 version 5.0.1
sccp
!
sccp ccm group 10
associate ccm 5 priority 1
associate ccm 2 priority 2
associate profile 12 register CON001193484810
associate profile 10 register RTP001193484810
!
dspfarm profile 12 conference
description conference bridge
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729abr8
codec g729r8
codec g729br8
maximum sessions 10
associate application SCCP
!
dspfarm profile 10 mtp
codec g729r8
maximum sessions software 12
associate application SCCP
!
!
dial-peer voice 100 voip
description voip dial peer to Verizon
destination-pattern .T
voice-class codec 1
session protocol sipv2
session target sip-server
```

```
incoming called-number 1...
dtmf-relay rtp-nte digit-drop
no vad
!
dial-peer voice 205 voip
description dial-peer to UC Manager
preference 5
destination-pattern 1...
voice-class codec 1
session target ipv4:192.168.0.6
incoming called-number .
voice-class h323 1
dtmf-relay h245-alphanumeric
!
dial-peer voice 200 voip
description dial-peer to UC Manager
preference 2
destination-pattern 1...
voice-class codec 1
session target ipv4:192.168.0.4
incoming called-number .
voice-class h323 1
dtmf-relay h245-alphanumeric
!
sip-ua
retry invite 2
retry bye 2
retry cancel 2
retry options 0
sip-server dns:abc.com
g729-annexb override
!
!
!
line con 0
exec-timeout 0 0
stopbits 1
line aux 0
stopbits 1
line vty 0 4
exec-timeout 0 0
password xxxx
login
!
scheduler allocate 20000 1000
!
end
```

Troubleshooting

This document's goal is to help identify troublesome/problematic areas with some basic troubleshooting techniques for this topology. This is not meant to be an exhaustive resource for all types of problems.

Basic call troubleshooting

There are four main components in troubleshooting UC Manager SIP trunking with CUBE (H.323 to SIP):

- **UC Manager**
The only way to see how UC Manager is behaving is to enable traces on UC Manager. Analyzing these traces are best left to Cisco TAC and software developers. The recommendation for this process is to make sure the configuration is correct.
- **MTP registration and usage**
There are two checks here. UC Manager must show that the MTP is registered and the router MTP must show that it has registered.

Find and List Media Termination Points

Status
3 records found

Search Options
Find Media Termination Point where Name begins with Find Search Within Results
(device.name begins with any)

Search Results

Name	Description	Device Pool	Status
MTP_2	CCM SW-based MTP	Default	Registered with 192.168.0.4
MTP_3	CCM SW-based MTP	Default	Registered with 192.168.0.4
RTP001193484810	IOS SW-based MTP H.323 FastStart	Default	Registered with 192.168.0.4

Add New Select All Clear All Delete Selected Reset Selected Rows per Page 50

On the router, use show sccp.

```
cube#sho sccp
SCCP Admin State: UP
Gateway IP Address: 192.168.0.10, Port Number: 2000
IP Precedence: 5
User Masked Codec list: None
Call Manager: 192.168.0.4, Port Number: 2000
Priority: 1, Version: 5.0.1, Identifier: 5
Call Manager: 192.168.0.6, Port Number: 2000
Priority: 2, Version: 5.0.1, Identifier: 2
```

```
Software MTP Oper State: ACTIVE - Cause Code: NONE
Active Call Manager: 192.168.0.4, Port Number: 2000
TCP Link Status: CONNECTED, Profile Identifier: 10
Reported Max Streams: 24, Reported Max OOS Streams: 0
!24 streams are double the configured maximum sessions. Two MTP resources will
allocated per call. However, only configure maximum sessions to correlate with
the number of calls needed.
```

```
Supported Codec: g729r8, Maximum Packetization Period: 60
Supported Codec: rfc2833 dtmf, Maximum Packetization Period: 30
Supported Codec: rfc2833 pass-thru, Maximum Packetization Period: 30
Supported Codec: inband-dtmf to rfc2833 conversion, Maximum Packetization
Period: 30
```

```
Conferencing Oper State: ACTIVE - Cause Code: NONE
Active Call Manager: 192.168.0.4, Port Number: 2000
TCP Link Status: CONNECTED, Profile Identifier: 12
Reported Max Streams: 80, Reported Max OOS Streams: 0
Supported Codec: g711ulaw, Maximum Packetization Period: 30
Supported Codec: g711alaw, Maximum Packetization Period: 30
Supported Codec: g729ar8, Maximum Packetization Period: 60
Supported Codec: g729abr8, Maximum Packetization Period: 60
Supported Codec: g729r8, Maximum Packetization Period: 60
Supported Codec: g729br8, Maximum Packetization Period: 60
Supported Codec: rfc2833 dtmf, Maximum Packetization Period: 30
Supported Codec: rfc2833 pass-thru, Maximum Packetization Period: 30
```

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Supported Codec: inband-dtmf to rfc2833 conversion, Maximum Packetization Period: 30

How UC Manager allocates the MTPs can only be seen on traces. Again, engage Cisco TAC to help troubleshoot those issues.

If an MTP is not allocated, the result is a SIP delayed offer (i.e., INVITE without SDP).

- **Check to see if the MTP resource is registered on UC Manager.**
- **Check to see if the router also shows registration to UC Manager**
- **Make sure the codec configured for the MTP (on the router) matches the codec that you see in the SIP SDP.**
- **Make sure the H.323 gateway has been configured properly (i.e., checks and unchecks, FastStart codec selection.**
- **Make sure the MRGL associated to the H.323 gateway contains the MTP resource configured.**

• CUBE H.323

Step 1. Make sure you are matching the dial peers as designed. There should be two dial peers for each call. Dial peer matching should be deterministic and predictable. Use `debug voip dialpeer inout`. The matched dial peer determines how you call will be treated.

```
010033: *Oct 19 19:13:52.003: //-  
1/00914EBF0600/DPM/dpAssociateIncomingPeerCore:  
  Calling Number=1016, Called Number=5303520012, Voice-Interface=0x0,  
  Timeout=TRUE, Peer Encap Type=ENCAP_VOIP, Peer Search Type=PEER_TYPE_VOICE,  
  Peer Info Type=DIALPEER_INFO_SPEECH  
010034: *Oct 19 19:13:52.003: //-  
1/00914EBF0600/DPM/dpAssociateIncomingPeerCore:  
  Result=Success(0) after DP_MATCH_INCOMING_DNIS; Incoming Dial-peer=205  
010035: *Oct 19 19:13:52.003: //-  
1/00914EBF0600/DPM/dpAssociateIncomingPeerCore:  
  Calling Number=1016, Called Number=5303520012, Voice-Interface=0x0,  
  Timeout=TRUE, Peer Encap Type=ENCAP_VOIP, Peer Search Type=PEER_TYPE_VOICE,  
  Peer Info Type=DIALPEER_INFO_SPEECH  
010036: *Oct 19 19:13:52.003: //-  
1/00914EBF0600/DPM/dpAssociateIncomingPeerCore:  
  Result=Success(0) after DP_MATCH_INCOMING_DNIS; Incoming Dial-peer=205  
010037: *Oct 19 19:13:52.003: //-1/00914EBF0600/DPM/dpMatchPeersCore:  
  Calling Number=, Called Number=5303520012, Peer Info  
  Type=DIALPEER_INFO_SPEECH H  
010038: *Oct 19 19:13:52.003: //-1/00914EBF0600/DPM/dpMatchPeersCore:  
  Match Rule=DP_MATCH_DEST; Called Number=5303520012  
010039: *Oct 19 19:13:52.003: //-1/00914EBF0600/DPM/dpMatchPeersCore:  
  Result=Success(0) after DP_MATCH_DEST  
010040: *Oct 19 19:13:52.003: //-1/00914EBF0600/DPM/dpMatchPeersMoreArg:  
  Result=SUCCESS(0)  
  List of Matched Outgoing Dial-peer(s):  
  1: Dial-peer Tag=100
```

Step 2. Use `debug h225 events` to see that you are receiving something:

```
007030: *Oct 4 22:36:06.132: h323chan_chn_process_read_socket: fd=0 of type  
LISTENING has data
```

```
007031: *Oct 4 22:36:06.136: Changing to new event: ACCEPT  
h323chan_chn_accept: fd=0
```

```
007032: *Oct 4 22:36:06.136: h323chan_gw_accept: TCP connection accepted from  
192.168.0.6:57492 on fd=2
```

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```
007033: *Oct  4 22:36:06.136: h323chan_chn_accept: Local(0x0) accepts TCP conn
from 192.168.0.6(0xC0A80006) port (57492); fd=2changing from LISTENING state to
ACCEPTED state
```

```
007034: *Oct  4 22:36:06.144: h323chan_chn_process_read_socket: fd=2 of type
ACCEPTED has data
```

Step 3. Use `debug h225 asn1` look for

- `fastStart`
- `mediaWaitForConnect FALSE`

```
007080: *Oct  4 22:38:56.588: H225.0 INCOMING PDU ::=
```

```
value H323_UserInformation ::=
{
  h323-uu-pdu
  {
    h323-message-body setup :
    {
      protocolIdentifier { 0 0 8 2250 0 5 }
      sourceAddress
      {
        dialedDigits : "1016",
        h323-ID : {"1016..."}
      }
      sourceInfo
      {
        vendor
        {
          vendor
          {
            t35CountryCode 181
            t35Extension 0
            manufacturerCode 18
          }
          productId '436973636F43616C6C4D616E61676572'H
          versionId '31'H
        }
        terminal
        {
        }
        mc FALSE
        undefinedNode FALSE
      }
      destinationAddress
      {
        dialedDigits : "5303521520"
      }
      activeMC FALSE
      conferenceID '80D4D2945686517003001101C0A8006C'H
      conferenceGoal create : NULL
      callType pointToPoint : NULL
      sourceCallSignalAddress ipAddress :
      {
        ip 'C0A80006'H
        port 1720
      }
      callIdentifier
      {
        guid '80D4D2945686517003001101C0A8006C'H
      }
      fastStart
    }
  }
}
```

```
    {
      '0000000D4001800A04000100C0A8000A41D1'H,
      '40FFFE060401004D4001801114000100C0A8000A...'H
    }
    mediaWaitForConnect FALSE
    canOverlapSend FALSE
    multipleCalls FALSE
    maintainConnection FALSE
  }
  h245Tunneling FALSE
  nonStandardControl
  {
    {
      nonStandardIdentifier h221NonStandard :
      {
        t35CountryCode 181
        t35Extension 0
        manufacturerCode 18
      }
      data '80A4000400010200'H
    }
  }
}
```

Step 4. Use `debug h245 asnl` to see the codec offered. Unfortunately, this does not provide the specific flavor of G.729.

```
007086: *Oct  4 22:38:56.592: H245 FS OLC INCOMING PDU ::=
value OpenLogicalChannel ::=
{
  forwardLogicalChannelNumber 1
  forwardLogicalChannelParameters
  {
    dataType audioData : g729 : 2
    multiplexParameters h2250LogicalChannelParameters :
    {
      sessionID 1
      mediaControlChannel unicastAddress : ipAddress :
      {
        network 'C0A8000A'H
        tsapIdentifier 16849
      }
    }
  }
}
```

- **CUBE SIP. Use `debug ccsip` messages to see the SIP messages being sent out to Verizon.**

```
001553: *Oct 15 19:03:02.350: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
INVITE sip:5303520012@schidan0017.iptrunksit1.gshiv.com:5072 SIP/2.0
Via: SIP/2.0/UDP 166.38.98.31:5060;branch=z9hG4bK5247D
Remote-Party-ID: <sip:1016@166.38.98.31>;party=calling;screen=yes;privacy=off
From: <sip:1016@166.38.98.31>;tag=23852C38-A63
To: <sip:5303520012@schidan0017.iptrunksit1.gshiv.com>
Date: Mon, 15 Oct 2007 19:03:02 GMT
```


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```
Call-ID: 165AE602-7A8811DC-87E28522-144506E5@166.38.98.31
Supported: 100rel,timer,resource-priority,replaces
Min-SE: 1800
Cisco-Guid: 16239765-3486790001-50332418-3232235625
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE,
NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Max-Forwards: 70
Timestamp: 1192474982
Contact: <sip:1016@166.38.98.31:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 270

v=0
o=CiscoSystemsSIP-GW-UserAgent 5025 1992 IN IP4 166.38.98.31
s=SIP Call
c=IN IP4 166.38.98.31
t=0 0
m=audio 18430 RTP/AVP 18 101
c=IN IP4 166.38.98.31
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
```

One-way audio issues

1. **Make sure IP routing is accurate.**
2. **When the router has multiple active IP addresses, some H.323 or SIP signaling may be sources from one IP address and other parts of it may reference a different source address. This may result in one-way audio. In order to get around this problem, you can bind the H.323 and SIP signaling to their own specific source addresses.**

```
interface fa0/0
  h323-gateway voip bind srcaddr a.b.c.d
voice service voip
  sip
    bind media source interface y
```

3. **Use debug voip rtp to determine where media is flowing (i.e., any tx/rx packets) and if you are actually getting two way communications (i.e., look for tx AND rx packets from the same set of IP addresses).**

```
002680: *Oct 17 14:09:40.220: RTP(64322): fs tx s=192.168.0.10(17300),
d=192.168.0.105(16384), pt=18, ts=A07D80, ssrc=F25EBC69
002681: *Oct 17 14:09:40.244: RTP(64323): fs rx s=166.34.84.44(16056),
d=166.38.98.31(17782), pt=18, ts=A07E20, ssrc=F25EBC69
002682: *Oct 17 14:09:40.244: RTP(64323): fs tx s=192.168.0.10(19480),
d=192.168.0.10(18420), pt=18, ts=A07E20, ssrc=F25EBC69
002683: *Oct 17 14:09:40.244: RTP(64323): fs rx s=192.168.0.10(19480),
d=192.168.0.10(18420), pt=18, ts=A07E20, ssrc=F25EBC69
002684: *Oct 17 14:09:40.244: voip_rtp_xmit:no context
002685: *Oct 17 14:09:40.244: RTP(64323): fs tx s=192.168.0.10(17300),
d=192.168.0.105(16384), pt=18, ts=A07E20, ssrc=F25EBC69
002686: *Oct 17 14:09:40.260: RTP(64324): fs rx s=166.34.84.44(16056),
d=166.38.98.31(17782), pt=18, ts=A07E20, ssrc=F25EBC69
002687: *Oct 17 14:09:40.260: RTP(64324): fs tx s=192.168.0.10(19480),
d=192.168.0.10(18420), pt=18, ts=A07E20, ssrc=F25EBC69
```

```
002688: *Oct 17 14:09:40.260: RTP(64324): fs rx s=192.168.0.10(19480),
d=192.168.0.10(18420), pt=18, ts=A07EC0, ssrc=F25EBC69
002689: *Oct 17 14:09:40.260: voip_rtp_xmit:no context
002690: *Oct 17 14:09:40.260: RTP(64324): fs tx s=192.168.0.10(17300),
d=192.168.0.105(16384), pt=18, ts=A07EC0, ssrc=F25EBC69
002691: *Oct 17 14:09:40.268: RTP(0): ps rx s=192.168.0.105(16384),
d=192.168.0.10(17300), pt=18, ts=370, ssrc=BA1A9E0B
002692: *Oct 17 14:09:40.268: voip_rtp_xmit:no context
002693: *Oct 17 14:09:40.268: RTP(0): fs tx s=192.168.0.10(18420),
d=192.168.0.10(19480), pt=18, ts=370, ssrc=BA1A9E0B
002694: *Oct 17 14:09:40.268: RTP(0): ps rx s=192.168.0.10(18420),
d=192.168.0.10(19480), pt=18, ts=370, ssrc=BA1A9E0B
002695: *Oct 17 14:09:40.268: RTP(0): fs tx s=166.38.98.31(17782),
d=166.34.84.44(16056), pt=18, ts=370, ssrc=BA1A9E0B
002696: *Oct 17 14:09:40.280: RTP(64325): fs rx s=166.34.84.44(16056),
d=166.38.98.31(17782), pt=18, ts=A07F60, ssrc=F25EBC69
002697: *Oct 17 14:09:40.280: RTP(64325): fs tx s=192.168.0.10(19480),
d=192.168.0.10(18420), pt=18, ts=A07F60, ssrc=F25EBC69
002698: *Oct 17 14:09:40.280: RTP(64325): fs rx s=192.168.0.10(19480),
d=192.168.0.10(18420), pt=18, ts=A07F60, ssrc=F25EBC69
002699: *Oct 17 14:09:40.280: voip_rtp_xmit:no context
002700: *Oct 17 14:09:40.280: RTP(64325): fs tx s=192.168.0.10(17300),
d=192.168.0.105(16384), pt=18, ts=A07F60, ssrc=F25EBC69
002701: *Oct 17 14:09:40.284: RTP(1): fs rx s=192.168.0.105(16384),
d=192.168.0.10(17300), pt=18, ts=410, ssrc=BA1A9E0B
002702: *Oct 17 14:09:40.284: voip_rtp_xmit:no context
```

You can refer to the following for more details.

http://www.cisco.com/en/US/tech/tk652/tk698/technologies_tech_note09186a008009484b.shtml

Last-ditch effort

If you are able to get any send or receive of SIP message to/from Verizon, you can use the disconnect cause in the 200Ok message (from debug ccsip message) to point you in the right direction.

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 166.34.84.88:5072;branch=z9hG4bKvc0ap8308gfhkdcj77k1sds811pm3.1
From: <sip:5303520012@schidan0017.iptrunksit1.gsiv.com>;tag=1402136203-1192820941399
To: <sip:1016@166.38.98.31>;tag=3828860C-1DEF
Date: Fri, 19 Oct 2007 19:13:59 GMT
Call-ID: 433C2BF4-7DAE11DC-8F088522-144506E5@166.38.98.31
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 483758317 BYE
Reason: Q.850;cause=16
Content-Length: 0
```

The most common disconnect cause is 16, which indicates a normal call clearing condition. However, 65 equates to bearer capability not implemented, which is most likely due to a codec mismatch.

Here is the entire list:

http://www.cisco.com/en/US/docs/ios/11_3/debug/command/reference/disdn.html#wp94

Where to get help

As stated previously, this document is not all inclusive. Please refer to the Cisco Technical Assistance Center for further troubleshooting assistance.