IC 4.0 SIP INTEGRATION TO

CISCO UNIFIED COMMUNICATIONS MANAGER 10.x

Validated Integrations:
CUCM 10.x with xIC version 4.0 SU-6 (support included for all 4.0 SU’s)

INTEGRATION DOCUMENT

Version 4.08

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## Revision Control

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<th>Author</th>
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<tr>
<td>4.00</td>
<td>3/27/11</td>
<td>Initial draft.</td>
<td>Jason Probala</td>
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<td>Jason Probala</td>
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<td>Jason Probala</td>
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<td>Jason Probala</td>
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<td>Added CUCM configuration screenshot for SIP Profile</td>
<td>Wesley Cook</td>
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<tr>
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<td>9/05/2013</td>
<td>Added Interaction Administrator Server Parameter section</td>
<td>Wesley Cook</td>
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<td>1/02/2015</td>
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<td>Wesley Cook</td>
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Introduction

Integrating with Cisco Unified Communications Manager via SIP opens the door for leveraging existing flexibility and product support without relying on the Cisco TSP. It allows for customers to bring converged architectures onto a single platform without sacrificing important product functionality. The integration will continue to use CUCM station configuration, MGCP and H.323 gateway connections, as well as add support for a distributed environment with audio flowing directly between the IC Media Server and the two endpoints. Remote offices, with their local SIP gateways can take full use of Media Server capabilities, by keeping the audio local yet maintaining call recording.

Stations are configured as standard IC unmanaged workstations, allowing for MWI and consistency of dialing from either the CUCM phone or the IC client. Phone functionality, such as hold and conference, are also persevered when done either from the phone or the IC client. Inbound calls, both from another CUCM station or the PSTN via an MGCP/H.323 gateway, are sent over from CUCM and routed to the intended user. The user, if logged on to a CUCM workstation configured in IA, will be alerted on their CUCM controlled phone. Outbound calls (from the phone) can be configured to appear in the IC client as if it were dialed from the client itself.

Products that have been limited in a TAPI environment, such as Interaction Dialer, because of the lack of call analysis, can now be integrated into a CUCM environment. Using an Interaction Intelligence Gateway for outbound dialing is now an option with agents using CUCM stations.

Testing has been completed for the use of SIP stations connected directly to the IC server concurrently with CUCM stations. The testing required a significant number of additional conference, transfers, hold, coaching, and monitoring scenarios that were proved to integrate well when using a media server.

Interactive Intelligence has completed interoperability testing with Cisco Unified Communication Manager version 10.0 with xIC version 4.0 SU-6. This document is provided to show configuration, limitations, additions, and best practices for a successful integration.

Naming Definitions

<table>
<thead>
<tr>
<th>PartitionA</th>
<th>Partition with only the Route Pattern going to IC</th>
</tr>
</thead>
<tbody>
<tr>
<td>PartitionB</td>
<td>Partition with all the Agents phone directory numbers</td>
</tr>
<tr>
<td>PartitionC</td>
<td>Partition with non-Agent DNS</td>
</tr>
<tr>
<td>PartitionD</td>
<td>Partition with non-Agent DNS (if there are multiple non-agent partitions)</td>
</tr>
<tr>
<td>SearchSpaceA</td>
<td>Search Space with PartitionA Only</td>
</tr>
<tr>
<td>SearchSpaceBCD</td>
<td>Search space with PartitionB, PartitionC, PartitionD (order is important)</td>
</tr>
<tr>
<td>ICRTrunk1</td>
<td>SIP Trunk configured to route calls to IC redundant servers</td>
</tr>
<tr>
<td>ICRTrunk2</td>
<td>SIP Trunk configured to route calls to IC redundant servers</td>
</tr>
<tr>
<td>ICSecurityProfile</td>
<td>Security Profile used to set protocol, port, and services of the SIP Trunk</td>
</tr>
<tr>
<td>ICSIPProfile</td>
<td>Profile used to define the DTMF Payload type, SIP timers, and Hold type</td>
</tr>
<tr>
<td>ICRRouteGroup</td>
<td>Route Group with both ICRTrunk1 and ICRTrunk2 for redundancy</td>
</tr>
<tr>
<td>ICRRouteList</td>
<td>Used to set the Wildcard route pattern to both IC Servers</td>
</tr>
<tr>
<td>Route Pattern</td>
<td>Wildcard Route Pattern to send all calls in SearchSpaceAB to CaaS</td>
</tr>
</tbody>
</table>
Configuration Summary

This section highlights the major configuration settings, requirements, and limitations that are important to a successful integration. These items are described in more detail in the appropriate sections later in the document.

IC Configuration

- IC Station and Line audio settings need to be configured to use TCP and Always-In and with Normal Media.
- The connection address is manually configured because the CUCM SIP trunk does not dynamically register.
- The connection address is set to the Cisco Phone extension. The full connection address will be set when the proxy configured on the SIP line is added to complete the address.
- The Station configuration will have Use Proxy for Station Connection selected.
- The Line configuration will use TCP and the proxy on the line will have a list of the CUCM's for the Proxy address.
- Interaction Media Server should be used in this configuration and is used on all calls because the line setting is configured as Always-In.

CUCM Configuration

- In the CUCM SIP Trunk configuration, it is preferred that the Require MTP resource be unselected. An MTP resource can be used to resolve DTMF RFC2833 issues.
- SIP Trunk Security Profile needs to have TCP+UDP for Incoming Transport Type and TCP for Outgoing Transport Type. This profile also needs Accept Presence Subscription, Accept Out-of-Diall REFER, Accept Unsolicited Notifications, Accept Replaces Header selected.
- Create a Route Pattern that will send all calls dialed from the unmanaged Cisco phones to the IC server(s)
- The Cisco phones are created as SIP workstations in IA
- In the SIP Profile, select “Conference Join Enabled, RFC 2543 Hold, and Semi Attended Transfer” and “Early Offer support for voice and video calls (insert MTP if needed) and Send send-receive SDP in mid-call INVITE, and Allow multiple codecs in answer SDP.”
- Setting the Cisco Phone to Auto Answer using the Headset is recommended in an agent environment. The SIP Trunk does not support the Talk Event.

Additional Information

- Faxing is supported over the SIP trunk with the required gateway IOS and IC version.
- Off hook and dialing call states from the Cisco phone will not be indicated on the IC client. This is standard for any SIP implementation, but differs from a TAPI integration.
- Pressing Hold from the Cisco Phone will provide Cisco Hold Music. Selecting Hold from the IC client will provide IC hold Music.
- If the CUCM station is unplugged or loses network connectivity, IC will continue to send calls to the user and then roll to VM. If the user is an agent, the first ACD routed call will alert, then change the user’s status to ACD-Agent Not Available.
- Auto Answer must be set on the CUCM to auto off-hook a connection call
- CUCM does not do a Putback transfer, but can accept it
- Transfer from the phone keeps the call in the IC client until the call has ended
• MWI is not supported with Unified Communications Express
• MTP resources are needed for devices that do not support RFC 2833 DTMF

**Major Configuration Points**

**Both IC and CUCM**

• No IC ACD voicemail – timeout on ININ ACD must be less than FNA on CUCM station
• Direct voicemail is supported by configuring a Unified Messaging station pointed to Unity and applied to each user
• IC User Extension is set to the Cisco DN extension
• Redundant CUCM Trunks are setup on the CUCM server to point to each IC server
• CUCM Wildcard Route Pattern is set up to send all dialed numbers to the IC server pair
• IC Wildcard Dial Plan Pattern is set up to send all dialed numbers to CUCM
• Search Spaces are setup to facilitate all dialed numbers sent over to IC
• IC users’ have the Cisco station as their default station
• IC stations have the Cisco DN as their station name
• A new Location (non-default) is setup in IC to allow checking of local extensions first
• The new IC Location will have ‘Enable Regional Dialing’ selected and all Lines, Media Servers, Users, and Unmanaged Stations

**Extension Details**

All calls dialed from or sent to the CUCM phone crosses the SIP trunk connecting IC to the CUCM cluster. This allows every call to appear in the IC client. It also provides dialing consistency for the end user, but requires a customer to be reliant on the IC server being active for outbound calls. Dialing from the phone and the IC client are all consistent if configured correctly.

1. **CaaS Station Configuration**
   • IC user **extensions** will match the CUCM directory numbers
   • IC stations should be **named** the Cisco DN that appears on the phone (could be an E.164 number)
   • IC Station **extension** will be unnecessary, but for consistency, it should be given the last 4 digits of the CUCM E.164 DN or some other defined extension (Dialing this extension should be rare)

2. **IC User Configuration**
   • IC user **extension** should be given the CUCM DN extension (possibly an E.164 number)
   • Default stations should be set – allows for the station to alert without the agent logged on to the CaaS client. This is needed so that DID calls can roll to Unity Voicemail
   • Timeout on the User for incoming interactions is to be set greater than that of CUCM RNA to allow the call to roll to Unity Voicemail
   • Outbound ANI on the User configuration should be set to the **User Extension** – station to station dialing will show the Outbound ANI on the remote party’s phone
Configuring a SIP trunk on the CUCM

The configuration of a SIP trunk on CUCM consists of three major components:

- **SIP Trunk Security Profile**: Configures the Protocol of the SIP Trunk
- **SIP Profile**: Configures RFC 2543 Hold
- **Inbound Search Space**: Configures the partition(s) to match on the inbound invite
- **SIP Trunk**: Configures MTP and Proxy Destination address

These four components are needed for a successful SIP Trunk configuration. Cisco has split these configuration settings into three components to allow an administrator to reuse the SIP Trunk Security Profile and the SIP Profile for every SIP Trunk that is configured. If multiple SIP Trunks were needed, only the SIP Trunk portion of the guide below would need to be re-executed. A step by step guide is provided below.

**SIP Trunk Security Profile**

1.) Under the Systems tab, highlight Security Profile and then select the SIP Trunk Security Profile option.
2. Select the default profile or create a new one and check to see that Accept Presence Subscription, Accept Out-of-Dialog Refer, Accept Unsolicited Notifications, and Accept Replaces Header are all selected. Also, setup TCP on the SIP Trunk in CUCM for faster failover to additional CUCM servers.

**SIP trunk Security Profile Setup** (System>Security>SIP Trunk Security Profile)

- **a.** Profile Name: ICSecurityProfile
- **b.** Security Mode: Non-secure
- **c.** Incoming Transport Type: TCP+UDP
- **d.** Outgoing Transport Type: TCP
- **e.** Incoming Port: 5060
- **f.** Checked: Accept Presence Subscription, Accept Out of Dialing REFER, Accept Unsolicited Notification, Accept Replaces Header
3. Under the DEVICE tab, inside DEVICE SETTINGS, select the SIP Profile option. Select “Conference Join Enabled, RFC 2543 Hold, and Semi Attended Transfer” and also select “Early Offer support for voice and video calls (insert MTP if needed) and Send send-receive SDP in mid-call INVITE.”
(Note: For IC Server to display the intended results for Hold, Transfer, and Conferencing, it is recommended that the CUCM Clusterwide Service Parameter for “Duplex Streaming Enabled” be set to the system default setting of “False.”)
The SIP Normalization Script is optional, but the provided script will add the original gateway or phone’s location to a SIP header on all outgoing SIP invites in a form that CIC can recognize as its location. This is very useful in determining which media server is needed on the original invite. This can also be set on the SIP Triunk itself, instead of the profile.

```lua
M= {}
M.allowHeaders = {"x-inin-crn"}
function M.outbound_INVITE(msg)
    local location = msg:getHeaderValueParameter("Call-info", "x-cisco-loc-name")
    msg:addHeader("x-inin-crn", "2000")
    msg:addHeaderValueParameter("x-inin-crn", "loc", location)
end
return M
```
CUQM Inbound Search Space and Partition Setup

Creating a partition for all CUCM phones needing control via CIC should be placed in a single partition. If that is not possible, then the inbound search space configured on the SIP trunk will need to include all of the phones partitions. The Phones are in location B, with a Search Space that only includes Partition A. A Route Pattern is configured for Partition A that routes all calls across the IC SIP Trunk.

1. **Partition Configuration** (Call Routing>Class of Control>Partition)
   a. Two Partitions are configured on CUCM
   b. PartitionA is for Route Pattern to CaaS
   c. PartitionB is where the Agents phone Directory Number resides

![Status](image)

**Partition Information**

- **Name**: PartitionA
- **Description**: PartitionA
- **Time Zone**: Originating Device

2. **Search Space Configuration** (Call Routing>Class of Control>Calling Search Space)
   a. Two Search Spaces are created
   b. SearchSpaceA will contain PartitionA
   c. SearchSpaceBCD will contain PartitionB, PartitionC, PartitionD (the order is important) to allow calls from the IC server to reach any of the Cisco DN's first and then any MGCP/h.323 gateways. Do not include Partition A.

![Status](image)

**Calling Search Space Information**

- **Name**: SearchSpaceBCD
- **Route Partitions for this Calling Search Space**
  - **Available Partitions**
    - LocalPortion
    - LocalOutboundPortion
    - StationsPortion
  - **Selected Partitions**
    - PartitionB
    - PartitionC
    - PartitionD
**SIP Trunks**

1. **CUCM SIP Trunk Setup (two Trunks)**

2. **Trunk Type**
   Set the Trunk type to "SIP Trunk." Be sure the Device Protocol automatically sets itself to SIP and then click Next.

3. **Configuration Settings**
   a. SIP Trunk Named CaaSTrunk1, CaaSTrunk2
   b. MTP resource not required
   c. Inbound Calling Search Space = SearchSpaceBCD
   d. Select Unattended Port
   e. SIP Information Destination Address =<IC Server 1 IP Address> (Trunk #1).
   f. SIP Information Destination Address =< IC Server 2 IP Address > (Trunk #2).
   g. Set the SIP Trunk Security Profile to CaasSecurityProfile
   h. Set the SIP Profile to CaaSSIPProfile
   i. DTMF Signaling Method = RFC2833
4. **Trunk Name Settings**

Under the Trunk Configuration dialog, create a Device Name for the trunk and set the Device Pool to Default.

5. **MTP Settings**

Also in the SIP Trunk configuration, there is an option to require a Media Termination Point. In most cases, the MTP should remain unselected. However, if Cisco endpoints, such as phones and gateways, are not configured for or do not support RFC 2833 DTMF, the MTP resource is required by Cisco to convert the DTMF to RFC2833. Older phones such as the 7910’s would require the MTP resource. Using an MTP resource on the CUCM server itself will force the RTP traffic through the CUCM server in both directions. If not using an MTP resource the RTP can travel from endpoint-to-endpoint (station) instead of forcing the CUCM to be in the audio path if using a SIP enabled gateway.
6. **Inbound Calling Search Space**  
Set the appropriate Calling Search Space to the partitions containing your CUCM phones and/or gateways.

In this example using SearchSpaceBCD - Partition B contains the phones, Partition C contains non agent phones, Partitition D contains the gateways.

7. **Outbound Diversion Header**  
The Redirection Diversion Header is selected in case there is a scenario where the call is forwarded on to ININ from another phone.
8. Destination Address Settings

In the Destination Address field, type the IP address of the IC server or SIP Proxy.

9. SIP Security Settings

Set the SIP Trunk Security Profile to the one created in sections 1 and 2 above.
Select Redirecting Diversion Header Delivery for both Inbound and Outbound. This is selected to support voicemail support on calls forwarded from the Cisco endpoint.
**Redundant SIP Trunk Configuration (Fault Tolerant)**

Configuring fault tolerance to allow for redundant IC servers has two parts

   a. Creating a second SIP Trunk as was done in the section above step 3  
   b. Creating a Route Group and Route List that contains both these SIP Trunks

Keep in mind that this is not limited to (2) SIP Trunks, this can be scaled increased to any number of SIP Trunks. The testing was completed with 7 SIP Trunks.

1. **Creating additional SIP Trunks**
   As indicated above, the section above shows the creation of a single SIP Trunk. Repeating this process, selecting the same configuration settings (including protocol and port number) is followed. The difference is on Step 3 section e; entering the secondary IC server’s IP address instead of the primary server allows for two SIP Trunks, each pointing to each IC server in the pair.

2. **Creating a Route Group and Route List**
   The last two remaining configurations are to create a Route Group and Route List.
   A Route Group is where the (X) number of SIP trunks that have been created will be grouped together.

   a. Setup a Route Group named: CaaSRouteGroup  
   b. Add the two SIP Trunks created in the earlier section: CaaSTrunk1 and CaasTrunk2 to the Route group members  
      i. If the trunk is already associated with a Route pattern it won’t be in the list  
   c. The order or the trunks in the group is ordered by priority

![Route Group Configuration](image)
A Route List is where the Route Groups can be grouped together. In most cases it will be the one Route Group that was created for the (X) number of IC server SIP Trunks.

a. Setup a Route list named CaaSRouteList
b. Add the CaaSRouteGroup as a List member
c. Verify the Route list Enabled checkbox is set
d. The List will register to the CaaS server and should indicate Registered

**Route Pattern**

1. **Route Pattern Setup** (Call Routing>Route/Hunt>Route Pattern)  
A single Route Pattern is created to force all dialing through the IC server with minimal route pattern configuration when dialing from the phone.

a. Create a wildcard route pattern to send all dialed traffic to the SIP Route List and set its route partition to PartitionA.
b. The Route Pattern set up with X# as the pattern – this allows the # to be dialed to end the dial string but is not required
c. The Route Pattern is put into PartitionA
d. The Gateway/RouteList is set to CaaSRouteList
e. Provide Outside Dial tone is un-checked
**Soft Key Button Configuration**

a. Create a new Softkey Layout Configuration for the CaaS Agent – CaaSUser

b. Select Add, copy from the std. user, save, then select Go in the upper right corner

c. On Hook: Add Redial, New Call, and Forward All as available softkeys for the agent

d. Connected: Add Hold and End as available softkeys for the agent

e. Assign this new Softkey Layout to the agent’s CUCM phone
**Phone DN Setup and Configuration**

- On the Device Information, the Calling Search Space is set to SearchSpaceA
- A new DN is added to the device by selection Add a new DN on Line [1]
- The E.164 or extension number is added as the Directory number on the line
- Provide the list of Phone E.164 or Extension Directory Numbers to the CaaS Implementation Team or IC administrator
- The Route Partition is set as PartitionB
- The No Answer Ring Duration is set to 38 (WAN speeds and latency may require this to be modified)
Inbound calls to Agents from non-Agents/Gateways

a. Inbound search spaces for Gateways and calling Search Spaces for non-Agent phones need to include Partition A. **The order is very important.**

b. Partition A needs to be immediately following the partitions with similar DNs

c. Partition A will have a wildcard route pattern, as well as route pattern – 19529xxxxxx configured exactly the same as the wildcard route pattern in section 8

d. Non-agent phone configuration (Calling Search Space and partition):

e. H.323 gateway configuration (Inbound search space):
Alerting Timeouts

a. ACD timeout – 30 seconds Default time (8 rings)
b. CUCM timeout – 38
c. CaaS user alert timeout – 42

Voicemail Pilot

1. Configure a Voice Mail Pilot number in the CUCM. Putting the Voicemail Pilot number in same search space as the route to the IC server ensures that calls rolling to voicemail go to the IC server.

Add the Voice Mail Pilot number to the CUCM Routing Pattern to forward to the IC Server over the SIP Trunk. In IC, setup the Message button under Server>Telephony Parameters>SIP.

To Allow MWI usage, ensure “Accept Unsolicited Notifications” is selected in the CUCM SIP Profile as indicated in step 2.

If using MWI in a CUCM cluster environment, ensure the cluster is set up as indicated in the “Cisco CUCM Cluster Support” section of this document.
Using IC as a Unified Messaging Provider Only

If using IC as a provider for voicemail only, with no other station requirements, then only users need to be created on the IC side. The Cisco Directory number will need its FNA (Forward No answer) set to the IC Voicemail Direct number.

Cisco DN = User Extension with no station on IC

1. Set Forward No answer on the Directory Number of the phone:

2. Create a Voice Mail Pilot Number then a Voice Mail Profile:

3. Assign that Profile to the Phone’s Directory Number:
4. On the IC server, in the Server Container\Configuration\Telephony Parameters\SIP, set the Message button to the Voicemail Pilot Number and the Voicemail Direct to the FNA number:

![Server Configuration](image1)

5. Also on the IC server side, the User extension linked to the appropriate mailbox is created with the extension of the Cisco DN. No station is needed in this setup. Neither are any additional licenses needed on the IC user.

![User Configuration - Cisco UM User 1](image2)
SRST Support

For remote locations needing remote survivability while maintaining the integration with CIC, Cisco’s SRST feature is supported by Interactive Intelligence. The primary configuration settings of the gateway are bolded below:

```
service timestamps debug datetime msec  
service timestamps log datetime msec  
no service password-encryption  
hostname SRST  
!  
boot-start-marker  
boot-end-marker  
!  
no aaa new-model  
!  
ip source-route  
!  
ip cef  
!  
no ipv6 cef  
nntp server 64.16.211.38  
!  
multilink bundle-name authenticated  
!  
voice service voip  
allow-connections sip to h323  
allow-connections sip to sip  
fax protocol cisco  
sip  
!  
voice-card 0  
!  
archive  
log config  
hidekeys  
!  
interface FastEthernet0/1  
ip address 10.10.11.255.255.0.0  
duplex auto  
speed auto  
```
Cisco CUCM Cluster Support

Supporting a cluster of CUCM servers consisting of multiple subscribers and a publisher is supported with the following requirements:

**Interaction Message Indicator:**
- The gateway/route configured in IMI must point to a CUCM server that is in each of the endpoint Device Pool. For example, if the Cisco phone is using Subscriber 1 as its primary and Subscriber 2 as its backup, a 503 error will be returned if the gateway/route in IMI is configured to send the MWI event to the publisher. The Publisher will need to be added as a tertiary server in the group or the IMI gateway/route will need to be set to either of the subscriber servers. Also, the same CUCM group assigned to the phone (via Device Pool) will need to be set on the SIP Trunk as well.

**CIC and MIC:**
- The SIP line configured in IA must point to CUCM server(s) that are in each of the endpoint Device Pools. For example, if the Cisco Phone is using Subscriber 1 as its primary and Subscriber 2 as its backup, a 503 error will be returned if the Proxy on the SIP line is pointed to the Publisher. The server indicated on the IC SIP line will need to be added to the CUCM group in the Device Pool, in this example as a tertiary, that the endpoints are associated with. Also, the SIP Trunk will need to be part of this same Device Pool.

Setup a CUCM group for the endpoints to receive messaging:
Setup a Device Pool that includes the server group defined above:

Set the Device Pool to the DP created above for both the SIP Trunk and the endpoints receiving the MWI and/or SIP messaging: (Screen capture below only shows the SIP trunk)
**Additional CUCM Feature Configuration**

**Putback Transfer (All call control and RTP out of IC)**
- Verify Step 2 in the [SIP Trunk Security Profile](#) configuration is set
  - Accept Replaces Header is selected
- Verify Step 10 in the [SIP Trunk](#) configuration is set
  - Set the appropriate Rerouting Calling Search Space to the partitions containing your CUCM phones and/or gateways receiving the transfer

**Multiple CUCM Regions**

Some environments have multiple regions configured in the CUCM architecture. Using different codecs between regions is supported with IC 4.0 SU-1. The SIP trunk may be in a region that uses G.711 while a remote phone may be across a WAN link in a separate region using G729. With SU-1, the invite message sent to CUCM will include all available codecs configured in the Regionalization container for the Location.

**Media Termination Point Troubleshooting**

CUCM will utilize a Media Termination Point for some operations regardless of selecting “Media Termination Point Required” on the SIP line. If there are MTP resources in the same Device Pool as the SIP line MTP resources may be in use without the administrator knowing. To determine the state of the MTP resources, as well as failed attempts to use them, the Cisco Real Time Monitoring Tool can help understand what resources are being used. If an administrator is oversubscribing the number of available MTP resources, the RTMT (Real Time Monitoring Tool) can be used to see the rate of these failed attempts.
IC Server Configuration

**Media Server**

1. The Interactive Media Server is required in the CUCM integration. This is configured using the same process as a traditional SIP solution. For a distributed architecture, consisting of multiple locations and multiple media servers, create locations in the Regionalization container and place the media servers in the appropriate location. This will allow Audio plays to flow from a local SIP gateway to the CUCM phone via the local media server. IC 4.0 includes the support for play prompts by default and is supported in the CUCM integration. The Media Server is required for all audio operations in 4.0. Refer to the SIP Normalization Script in
Line Configuration

2. The Audio settings on the Stations and SIP Line should be configured with *Always In* as the audio path and DTMF Payload set as 101. RFC 2833 is the only DTMF type supported in 4.0 and is thus removed as a configurable option on the Audio page.

3. The *Session* configuration page allows for the setting of the Media Timing. Setting both the *Media Timing* and the *Media reInvite Timing to Normal* ensures that the SDP is sent on all invite messages from IC to CUCM/UCME.
4. The Transport Protocol should be set to TCP on port 5060 to match the CUCM/UCME configuration. This allows for more efficient failover by reducing the time it takes to contact the secondary CUCM/UCME.

5. Setting the Proxy to the Primary and backup CUCM / UCME servers will allow for redundancy. If there are more CUCMs or UCMEs in the cluster continue to add them in priority order. This order should be consistent with the CUCM order configured on the stations.
**MWI**

6. Enable the MWI light on each IC User and Station
Station Configuration

7. **Station Name**

   The station name should be set as the Cisco DN. This allows the agent\Cisco user to log on to their client by using the Cisco DN on their desk as the station name.

**The Station Extension**

is set to the ININ Station Extension [8000]. This extension would rarely if ever be dialed; the user extension will match the Cisco DN number. This station will be set as the user's default station to allow calls that are made to an ININ user that is not logged in to alert the phone. If a phone is an intercom/lobby phone, the Station extension will be set to the Cisco DN extension.

**The Identification Address**

[7000] is configured as the Cisco Phone Extension. The “Use User Portion only” is selected because in a CUCM cluster architecture the Host portion will change depending on the active CUCM server.
The station’s Connection SIP Address uses the Cisco Phone Extension [7000] in the “Use an Alternate format” textbox to statically assign the CUCM’s phone extension. **This is done manually because CUCM does not dynamically register its phones with IC.**

8. To add the host portion to the station connection address the “Use Proxy for Station Connections” must be selected in the Default Station container in IA. This is located under the Global SIP Station tab in the Transport container.
User Configuration

The User Extension will be set to a number that will be dialed to contact the user, normally the Cisco DN of the station endpoint. This extension number cannot be the same as any other configured extensions. When a user extension is dialed, the call will route to the user, indicate in the IC client, and then will alert the station that the user is logged into. In most cases, the Cisco DN number will be the user extension with the default station set to the station with the same connection address. The Outbound ANI is also set to the caller id number that is desired to appear on the Cisco phone on a call that is initiated by the client.

A user dialing from a CUCM station will always dial the user extension to contact an IC user. The route pattern configured in CUCM will allow for the dialing of 9xxx, or if using the X!# wildcard then all user extensions, to be routed across the SIP trunk and alert the station the intended user is logged into. The intent of this call flow is to get IC in the call signaling path and Media Server in the Audio Path.

1. Outline of the IC user configuration
   a. IC user extension should be given the E.164 or station DN extension
   b. Default stations should be set – allows for the station to alert without the agent logged on to the CaaS client. This is needed so that DID calls can roll to Unity Voicemail
   c. Timeout on the User for incoming interactions is to be set greater than that of CUCM RNA to allow the call to roll to Unity Voicemail
   d. Outbound ANI on the User configuration should be set to the User Extension – station to station dialing will show the Outbound ANI on the remote party’s phone
2. **Dial Plan Configuration**

CaaS Dial plan will contain one Pattern – ‘Z’ directing all calls to the CUCM Line group

![Dial Plan Configuration](image)

3. **Agent Client Logon**

a. Agent will enter their IC username and password or check the option to “Use Windows Login Authentication”.

b. Host address will be setup on the DNS on the customer site to resolve to the IP address

c. Station Type is Workstation and the Cisco DN is the workstation name

![Agent Client Logon](image)
Interaction Administrator

Server Parameter, “DelayedHoldAllowed” must be added to allow support for Hold, Transfer, and Conferencing. The value must be set to “Yes” to utilize the added parameter.

(Note: This parameter was added in IC 4.0 SU2, and is not available in previous releases.)
Dialing Scenarios

1. Agent to Agent Dialing From the phone
   a. Transfers and Conferences are done via the client buttons, not the softkeys on the phone itself
   b. Display of the phone and the client when Agent Sandy Loehrs calls Agent John Armstrong from the Cisco phone:

John Armstrong Client/Phone

Sandy Loehrs Client/Phone
2. **Agent to Agent Dialing From the Client**
   a. Display of the phone and the client when Agent Sandy Loehrs calls Agent John Armstrong from the CaaS Client:

   **John Armstrong Client/Phone**

   ![Image of John Armstrong Client/Phone]

   **Sandy Loehrs Client/Phone**

   ![Image of Sandy Loehrs Client/Phone]
3. **Incoming call from a customer to a workgroup (agent)**
   
a. Call into the WestWestCentral Workgroup from an external Cell phone (answered by agent Sandy Loehrs):

![Image of call interface]

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**Audio Paths**

Inbound/Outbound calls to the PSTN

If using an MGCP gateway:
- **RTP:** Gateway <-> CUCM <-> Media Server <-> Phone
- **Signaling:** Gateway <-> CUCM <-> IC Server <-> CUCM <-> Phone

If using an h.323 gateway:
- **RTP:** Gateway <-> CUCM <-> Media Server <-> Phone
- **Signaling:** Gateway <-> CUCM <-> IC Server <-> CUCM <-> Phone

If using an SIP gateway:
- **RTP:** Gateway <-> Media Server <-> Phone
- **Signaling:** Gateway <-> CUCM <-> IC Server <-> CUCM <-> Phone
Network Architecture for CUCM SIP Integration

Architecture 1 - H.323/MGCP gateway connected to CUCM and SIP gateway connected to IC server

Incoming call to (555-9XXX) from PSTN to MGCP gateway or PSTN to SIP gateway.

1. Call from MGCP gateway is caught by CUCM Routing Pattern (5559XXX) and is routed to IC Server
2. IC Server processes the call from the SIP gateway or from the CUCM and routes it back to the CUCM
3. CUCM routes call to Cisco Phone 7XXX (User 9XXX)
4. Station connects call
Architecture 2- H.323/MGCP gateway connected to CUCM

5. Incoming call to (555-9XXX) from PSTN to MGCP gateway
6. Call is caught by CUCM Routing Pattern (5559XXX) and is routed to IC Server
7. IC Server processes the call and routes it back to the CUCM
8. CUCM routes call to station 7XXX (User 9XXX)
9. Station connects call
Architecture 3- SIP gateway connected to IC server

1. Incoming call to (555-9XXX) from PSTN to SIP gateway
2. Call is routed from the SIP gateway to the IC Server
3. IC Server processes call and routes it to the CUCM
4. CUCM routes call to station 7XXX (User 9XXX)
5. Station connects call
Architecture 4- CUCM station to station call

1. Station 1 places call to Station 2 connected to the CUCM
2. Call is caught by CUCM Routing Pattern (9XXX) and is routed to IC Server
3. IC Server processes call and routes it back to the CUCM
4. CUCM routes call to station 7XXX (User 9XXX)
5. Station connects call
CUCM Stations: Transfer Scenarios—Call Flow

Call from CUCM station to station - Transfer From Phone

Scenario – CUCM Station 1 calls CUCM Station 2 then transfers to CUCM Station 3

1. CUCM Station 1 places call to CUCM Station 2.
2. Call is caught by CUCM Routing Pattern (9XXX) and is routed to IC Server.
3. IC Server processes call and routes it to the CUCM.
4. CUCM routes call to CUCM Station 2 (7XXX)(User 9XXX.)
5. CUCM Station 1 presses “Transfer” on the phone and then places a call to CUCM Station 3.
6. Call is caught by CUCM Routing Pattern (9XXX) and is routed to IC Server.
7. IC Server processes call and routes it to the CUCM.
8. CUCM routes call to CUCM Station 3 (7XXX)(User 9XXX.)
9. CUCM Station 1 presses “Transfer” on the phone again and CUCM Station 2 is connected to CUCM Station 3.

Call from PSTN to CUCM Station - Transfer from Phone

Scenario – PSTN calls CUCM Station 1 then CUCM Station 1 transfers call to CUCM Station 2

(Architecture 2 above)

1. A call comes from the PSTN to CUCM Station 1.
2. Call is caught by CUCM Routing Pattern (9XXX) and is routed to IC Server.
3. IC Server processes call and routes it to the CUCM.
4. CUCM routes call to CUCM Station 1 (7XXX)(User 9XXX.)
5. CUCM Station 1 presses “Transfer” on the phone and then places a call to CUCM Station 2.
6. Call is caught by CUCM Routing Pattern (9XXX) and is routed to IC Server.
7. IC Server processes call and routes it to the CUCM.
8. CUCM routes call to CUCM Station 3 (7XXX)(User 9XXX.)
9. CUCM Station 1 presses “Transfer” on the phone again and PSTN call is connected to CUCM Station 2.
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Call from CUCM station to station - Transfer from IC Client

Scenario – CUCM Station 1 calls CUCM Station2 then transfers to CUCM Station3

1. CUCM Station1 places call from the IC Client to CUCM Station2.
2. IC Server processes call and routes it to the CUCM.
3. CUCM routes call to CUCM Station2 (7XXX)(User 9XXX.)
4. CUCM Station1 presses “Transfer” on the IC Client and types CUCM Station3’s number into the Transfer dialog.
5. IC Server processes call and routes it to the CUCM.
6. CUCM routes call to CUCM Station3 (7XXX)(User 9XXX.)
7. CUCM Station1 presses “Transfer Now” on the IC Client’s Transfer dialog and CUCM Station2 is connected to CUCM Station3.

Call from PSTN to IC Client - Transfer from IC Client

Scenario – PSTN calls CUCM Station1’s IC Client then CUCM Station1 transfers call to CUCM Station2

(Architecture 3 above)

1. A call comes from the PSTN to CUCM Station1’s IC Client.
2. IC Server processes call and routes it to the CUCM.
3. CUCM routes call to CUCM Station1 (7XXX)(User 9XXX.)
4. CUCM Station1 presses “Transfer” on the IC Client and types CUCM Station2’s number into the Transfer dialog.
5. IC Server processes call and routes it to the CUCM.
6. CUCM routes call to CUCM Station2 (7XXX)(User 9XXX.)
7. CUCM Station1 presses “Transfer Now” on the IC Client’s Transfer dialog and PSTN Call is connected to CUCM Station2.
CUFM Stations: Conference Scenarios – Call Flow

Conference between CUCM Stations – Conference from Phone

Scenario – CUCM Station1 calls CUCM Station2 then creates conference from phone with CUCM Station3

1. CUCM Station1 places call to CUCM Station2.
2. Call is caught by CUCM Routing Pattern (9XXX) and is routed to IC Server.
3. IC Server processes call and routes it to the CUCM.
4. CUCM routes call to CUCM Station2 (7XXX)(User 9XXX.)
5. CUCM Station1 presses “Conference” on the phone and then places a call to CUCM Station2.
6. Call is caught by CUCM Routing Pattern (7XXX) and is routed to IC Server.
7. IC Server processes call and routes it to the CUCM.
8. CUCM routes call to CUCM Station3 (7XXX)(User 9XXX.)
9. CUCM Station1 highlights existing call and presses “Conference” on the phone again and is connected to CUCM Station2 and CUCM Station3.

Conference between PSTN and CUCM Stations - Conference from Phone

Scenario – PSTN calls CUCM Station1 then CUCM Station1 conferences with CUCM Station2

(Architecture 2 above)

1. A call comes from the PSTN to CUCM Station1.
2. Call is caught by CUCM Routing Pattern (9XXX) and is routed to IC Server.
3. IC Server processes call and routes it to the CUCM.
4. CUCM routes call to CUCM Station1 (7XXX)(User 9XXX.)
5. CUCM Station1 presses “Conference” on the phone and then places a call to CUCM Station2.
6. Call is caught by CUCM Routing Pattern (9XXX) and is routed to IC Server.
7. IC Server processes call and routed it to the CUCM.
8. CUCM routes call to CUCM Station3 (7XXX)(User 9XXX.)
9. CUCM Station1 highlights the existing call and presses “Conference” on the phone again and the PSTN call is connected to CUCM Station1 and CUCM Station2.
Conference between CUCM Stations - Conference from IC Client

Scenario – CUCM Station 1 calls CUCM Station2 then creates Conference with CUCM Station3

1. CUCM Station1 places call from the IC Client to CUCM Station2.
2. IC Server processes call and routes it to the CUCM.
3. CUCM routes call to CUCM Station2 (7XXX)(User 9XXX.)
4. On CUCM Station1’s IC Client place the current call on Hold by pressing the “Hold” button on the IC Client and then place another call to CUCM Station3 from the IC Client.
5. IC Server processes call and routes it to the CUCM.
6. CUCM routes call to CUCM Station3 (7XXX)(User 9XXX.)
7. CUCM Station1 creates the conference by dragging and dropping the current connected call onto the existing call that is on Hold. CUCM Station1 and CUCM Station2 are both connected to CUCM Station3.

Call from PSTN to IC Client - Conference from IC Client

Scenario – PSTN calls CUCM Station1’s IC Client then CUCM Station1 creates a conf. with CUCM Station2

1. A call comes from the PSTN to CUCM Station1’s IC Client.
2. IC Server processes call and routes it to the
3. CUCM routes call to CUCM Station1 (7XXX)(User 9XXX.)
4. On CUCM Station1’s IC Client place the current call on Hold by pressing the “Hold” button on the IC Client and then place another call to CUCM Station2 from the IC Client.
5. IC Server processes call and routes it to the CUCM.
6. CUCM routes call to CUCM Station2 (7XXX)(User 9XXX.)
7. CUCM Station1 creates the conference by dragging and dropping the current connected call onto the existing call that is on Hold. CUCM Station1 and CUCM Station2 are both connected to the PSTN call.