CISCO UNIFIED COMMUNICATIONS MANAGER SIP INTEGRATION

Validated Integrations:
8.5 with xlC version 3.0 SU-10 and greater

INTEGRATION DOCUMENT

Version 2.03
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## Revision Control

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<th>Change Description</th>
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<td>Initial draft.</td>
<td>Jason Probala</td>
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<td>Modified for 8.5</td>
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<td>Added Audio Paths</td>
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<td>Added Codec support with CUCM regions</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 MGCP, H.323, or 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 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 8.5 with xIC version 3.0 SU-10. This document is provided to show configuration, limitations, additions, and best practices for a successful integration.
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. (pages 15-16)
- The connection address is manually configured because the CUCM SIP trunk does not dynamically register. (page 18)
- 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. (page 18)
- The Station configuration will have Use Proxy for Station Connection selected. (page 20)
- The Line configuration will use TCP and the proxy on the line will have a list of the CUCM’s for the Proxy address. (pages 15-16)
- Interaction Media Server should be used in this configuration and is used on all calls because the line setting is configured as Always-In. (page 14)

CUCM Configuration

- In the CUCM SIP Trunk configuration, MTP should to be unselected to allow audio to flow directly from endpoint to endpoint; bypassing CUCM as an intermediary. (page 10)
- 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-Dial REFERENCE, Accept Unsolicited Notifications, Accept Replaces Header selected. (page 7)
- Create a Route Pattern on the CUCM to route any ININ Extensions (ex. 7XXX, 8XXX) over the SIP Trunk to the IC Server. (page 12)
- 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.” (page 8)
- Setting the Cisco Phone to Auto Answer to Headset is recommended in an agent environment. The SIP Trunk does not support the Talk Event.

Additional Information

- Faxing will need to be directed to the HMP IC server directly from the gateway if possible.
- 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
Extension Details

It is important to understand that there are two significant configuration options a customer has to choose from.

Option 1 (Preferred Option):

This option sends all dialing from CUCM phones across the SIP trunk to IC. This allows every call to appear in the IC client. This 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, IC station extensions will match the CUCM directory numbers, and IC can provide call analysis, reporting, as well as on the phone presence.

Option 2:

This option is for customers that have existing CUCM configurations and do not want to modify CUCM call paths. This would be for integrations that consist mainly of inbound calling and using the Interaction Center as the ACD application. This option will keep CUCM station to station calls from appearing in the IC client and kept local to CUCM.

The following extensions would be used for this option:

<table>
<thead>
<tr>
<th>Extension Type</th>
<th>Extension Number</th>
</tr>
</thead>
<tbody>
<tr>
<td>ININ User Extension</td>
<td>9000</td>
</tr>
<tr>
<td>ININ Station Extension</td>
<td>8000</td>
</tr>
<tr>
<td>Cisco Phone Extension</td>
<td>7000</td>
</tr>
</tbody>
</table>

- **ININ users Extensions [9000]** will be used for internal dialing and external callers via the IVR to connect to the user. The user logs onto a configured ININ station to receive the alerting call. The ININ station is configured with the Cisco phone as the connection address.

- **ININ station Extension [8000]** has an Identification address configured as the Cisco phone extension [7000]. This is to allow outgoing calls from the Cisco phone to be identified as a configured ININ station. Calls needing to connect to external lines will need to be identified as known ININ stations before they will be routed.

- **Cisco Phone Extension [7000]** is configured on the Cisco Unified Communication Manager (CUCM). It is the directory number given during the CUCM configuration.
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
- **SIP Trunk**: Configures MTP and Proxy Destination address

These three 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 (Option 1 and 2)**

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 Profile (Option 1 and 2)

3. Under the DEVICE tab, inside DEVICE SETTINGS, select the SIP Profile option. Select “Conference Join Enabled, RFC 2543 Hold, and Semi Attended Transfer.”
SIP Trunk (Option 1 and 2)

4. Under the DEVICE tab, select the TRUNK option.

5. Click on "Add a New Trunk."

6. Set the Trunk type to "SIP Trunk." Be sure the Device Protocol automatically sets itself to SIP and then click Next.
7. Under the Trunk Configuration dialog, create a Device Name for the trunk and set the Device Pool to Default.

8. Also in the SIP Trunk configuration, MTP needs to remain unselected. This will allow the media to travel from endpoint-to-endpoint (station) instead of forcing the CUCM to be in the audio path.
9. In the Destination Address field, type the IP address of the IC server or SIP Proxy.

10. Set the SIP Trunk Security Profile to the one created in steps 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. Set the appropriate Rerouting Calling Search Space to the partitions containing your CUCM phones and/or gateways.
Route Pattern (Option 1)

This option routes all dialing through the IC server with minimal route pattern configuration.

1. Create a partition for the Trunk (LinePartition)
2. Create a partition for phones (PhonePartition)
3. Create a Calling Search Space for the phones (PhoneSS) - **only include the LinePartition**
4. Create a Calling Search Space for Inbound calls from IC (InboundSS) – include all partitions
5. Set the phones’ Calling search space to LinePartition
6. Create a wildcard route pattern to send all dialed traffic to the SIP trunk and set its route parting to LinePartition
7. On the SIP Trunk itself, set the Inbound Calls \ Calling Search Space to PhoneSS

Route Pattern (Option 2)

11. Setup a Route Pattern in CUCM to allow dialed IC User extension numbers to pass through the CUCM to the IC server across the SIP Trunk. (Ex. 9XXX will route over the SIP Trunk and then back to the CUCM station that the agent is logged into.)

![Cisco Unified CM Administration](image)

**Route Pattern 8888** is setup to allow for the Voicemail Pilot number as well as the Message button on the CUCM station to route across the SIP Trunk. Checking voicemail via the message button is supported.

**Route Pattern 8XXX** is to allow routing of standalone CUCM stations. This will be rare in most integrations.
Route Pattern 9XXX is configured to allow IC user extensions dialed from CUCM phones to route across the SIP Trunk to IC. This is a very important configuration step not to be overlooked.

Route Pattern 90XXXXXXX is configured if dialing external numbers from CUCM phones to PSTN gateways controlled by CUCM is to be represented in the IC client. The CUCM user will dial 91 + ten digit number. IC will need a matching entry in it's dial plan. In the IC Dial-plan entry, the number will be modified to 91XXXXXXX and routed back to CUCM. CUCM will match the 91XXXXXXX dial plan entry and route it to the assigned gateway.

Configure more Route Patterns in CUCM to allow additional IC extension numbers to pass through the CUCM to the IC Server using the SIP Trunk.

Voicemail Pilot (Option 1 and 2)

12. Configure a Voice Mail Pilot number in the CUCM.

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.
Fault Tolerance Configuration (Option 1 and 2)

13. Configuring fault tolerance to allow for redundant IC servers has two parts
   a. Creating a second SIP Trunk as was done in steps 4-10 above.
   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.

Creating additional SIP Trunks

As indicated above, steps 4-10 show 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 9; entering the secondary IC server’s IP address instead of the primary server allows for two SIP Trunks, each pointing to a single IC server.

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 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.

The final step is selecting the Route List for the Route Pattern directed to the IC servers.

Configuring a Route Group:
Name, select Top Down as the algorithm, select the two or (x) number of SIP Trunks to include in the group and save.

Note that any SIP Trunk that is already associated with a Route Patter will not show up in the list.
Configuring a Route List:
Name, select the Route Group, and save.

The last step is to select the configured Route list containing the Route Group of SIP Trunks to the IC servers as the destination of any Route Patterns needing to be directed to IC servers.
Cisco CUCM Cluster Support (Option 1 and 2)

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:

![Device Pool Configuration Diagram]

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)

![Trunk Configuration Diagram]
Additional CUCM Feature Configuration (Option 1 and 2)

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 3.0 SU11. 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-11, the invite message sent to CUCM will include all available codecs configured in the Regionalization container for the Location.
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 3.0 SU6 includes the support for play prompts and is also supported in the CUCM integration.

The Media Server is required for:

- RFC 2833 DTMF Negotiation issues
- Conferencing from the CUCM phone
- Transferring from the CUCM phone to an IC only controlled phone.
- Keeping the audio local at remote sites if using a SIP enabled gateway
### Command Servers:

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<td>2</td>
</tr>
<tr>
<td>2</td>
<td>ICServer2</td>
<td>Active</td>
<td>Yes</td>
<td>2</td>
</tr>
</tbody>
</table>

[Refresh] [Auto-refresh every 10s]
Line Configuration

2. The Audio settings on the Stations and SIP Line need to be configured with Always IN and RFC2833 enabled.

3. The SIP Line and SIP Station Session field should be configured with Normal Media.
4. Audio should also be set to TCP to match the CUCM. This allows for more efficient failover by reducing the time it takes to contact the backup CUCM.

5. Setting the Proxy to the Primary and backup CUCM will allow for redundant CUCM support. If there are more CUCMs 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 - IP430](image-url)

![User Configuration - user1](image-url)
Station Configuration

7. **The Station Extension is set to the ININ Station Extension [8000] to allow calls that are made to an ININ station that does not have a User logged into it.** This setting allows a potential intercom/lobby phone to be used to place and receive calls and it is not required to be associated with an agent.

**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. 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.

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 user extensions to be routed across the SIP trunk and alert the station the intended user is logged into; in this case it will be a CUCM station. The intent of this call flow is to get IC in the call signaling path and Media Server in the Audio Path.

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<td>NT Domain User:</td>
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</tr>
</tbody>
</table>

Exhibit: User Configuration - user1

Enable CC5 Integration

Current Status: Available

Home Site: (CustomerSite)

Current Site: <Not Set>
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

1. Incoming call to (555-9XXX) from PSTN to MGCP gateway or PSTN to SIP gateway.
2. Call from MGCP gateway is caught by CUCM Routing Pattern (5559XXX) and is routed to IC Server.
3. IC Server processes the call from the SIP gateway or from the CUCM and routes it back to the CUCM.
4. CUCM routes call to Cisco Phone 7XXX (User 9XXX).
5. Station connects call.
Architecture 2- H.323/MGCP gateway connected to CUCM

6. Incoming call to (555-9XXX) from PSTN to MGCP gateway
7. Call is caught by CUCM Routing Pattern (5559XXX) and is routed to IC Server
8. IC Server processes the call and routes it back to the CUCM
9. CUCM routes call to station 7XXX (User 9XXX)
10. 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
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 Station2 then transfers to 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 “Transfer” on the phone and then places a call to CUCM Station3.
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 Station3 (7XXX)(User 9XXX.)
9. CUCM Station1 presses “Transfer” on the phone again and CUCM Station2 is connected to CUCM Station3.

Call from PSTN to CUCM Station - Transfer from Phone

Scenario – PSTN calls CUCM Station1 then CUCM Station1 transfers call to 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 “Transfer” 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 routes it to the CUCM.
8. CUCM routes call to CUCM Station3 (7XXX)(User 9XXX.)
9. CUCM Station1 presses “Transfer” on the phone again and PSTN call is connected to CUCM Station2.
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.
CUCM 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 (7XXX) 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.