Add participants to existing conference or space in CMS Cluster deployment with Loadbalancing enabled

Contents

Introduction

This document describes how to add participants to an existing CMS conference in deployment of Clustered CMS with Load Balancing enabled.

Prerequisites

Requirements

Cisco recommends that you have knowledge of these topics:

- CMS Load Balancing (Cisco Meeting Server)
- CUCM ad-hoc conferencing (Cisco Unified Communications Manager)

This document assumes that Load Balacing is already configured for your clustered Callbridges (CB) and working for direct calls to these CMS servers (calling directly to an existing CMS space). This means that these requirements are already configured:

- All the CMS servers that are to be used for Adhoc conferencing are added to **CUCM** > **Media Resources** > **Conference Bridge** and are registered
- A **Media Resource Group List** (**MRGL**) which contains a **Media Resource Group** (**MRG**) is created, and it has the CMS servers only, and is the first group in the **MRGL**
- A **Route List** containing a **Route group** is created, and it has the CMS servers, and the selected **distribution algorithm** is **Circular**

Components Used

The information in this document is based on these software and hardware versions:

- CMS 2.9.1
- CUCM 12.5.1

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

Methods to add participant to existing CMS conference

> **Note**: There are three main methods of adding a participant to an existing CMS conference: add a participant via API, add a participant via Active Control, and add a participant without Active Control.

1. Add a participant via API

To use this method, **LoadbalanceOutgoingCalls** on the **Callbridge Group** has to be enabled.

To add the participant using this method, an **API POST** request has to be made to **/calls/<active-call-id>/participants/.** The **POST** request needs to include the **participantID** of the **participant** which is being added to the conference as value of the **remoteParty** parameter, which is part of this **POST** request.

This **POST** request instructs CMS to make an outgoing call to the participant which is being added. If **LoadbalanceOutgoingCalls** on the **Callbridge Group** is enabled, and if CMS has reached its load limit, it finds a free CMS server in the cluster to make an outgoing call to the participant being added, and a distributed call is created between the two servers. This is the same method used by **CMM** to add participants to a CMS conference.

2. Add a participant via Active Control

To use Active Control participant add, Active Control has to be negotiated first between the CMS server and the user which is adding the participant.

You need to enable Active Control on the **SIP Trunk Profile** that is configured on the **SIP Trunk** connecting CUCM with CMS, to do so enable parameter **Allow IX application media**, and note that the **Standard SIP Profile For TelePresence Conferencing** has it enabled by default. In addition, **LoadbalanceOutgoingCalls** on the **Callbridge Group** has to be enabled.

When a participant is added via Active Control to an existing CMS conference, CMS1 is instructed by the user (via active control message) to make an outgoing call to the new participant. If the load limit value configured on CMS1 is reached and the user tries to add a new participant with active control, CMS1 displays this error message (up to CMS version 2.9.1):

```
add participant "<participant-uri>" request failed: call bridge unavailable
```

This applies to both use cases - when the participant is added to an adhoc conference, and when it is added to an exsiting CMS space via active control.

This is a deffective behaviour and it is being tracked under the defect: CSCvu72374

3. Add a participant without Active Control

When a participant is added without using active control (therefore **Allow IX application media** not enabled on the **SIP Profile**), CUCM makes a call between the user who is initiating the action and the new participant. Then, when the user is ready to join the new participant to the conference, CUCM makes an outgoing call to the adhoc conference running on CMS1. If the load limit is reached on CMS1, the participant cannot be added and CMS1 displays this error message (55 is an example call number):

```
call 55: ending; local teardown, system participant limit reached - not connected after 0:00
```

This error message is a normal error message to be printed by a CMS server when it receives an incoming call and after it has reached its max load limit. It is then up to the call control server (CUCM or VCS) to continue routing the call to other members in the cluster. However, in the case of an adhoc conference, this does not work and it is not possible since CUCM does not have a **Route List** for adhoc conferences.

Configure

This document provides the configuration steps required to use the 3rd way of adding participant to existing conference (**Add a participant without Active Control**).

The behaviour addressed with the configurational steps in this document is:

1. User creates an adhoc conference, CMS1 server is hosting it

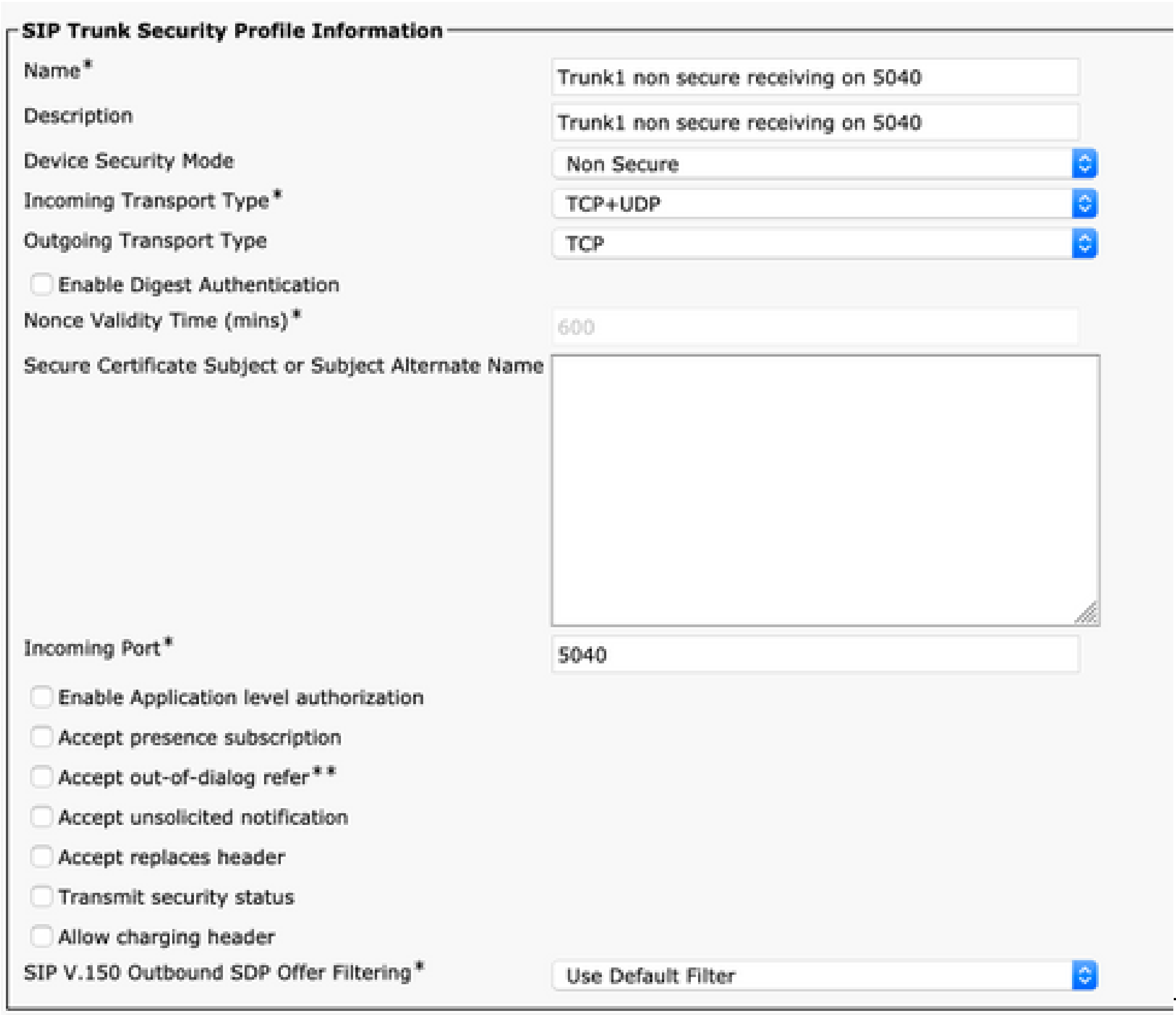
2. After the adhoc conference is established, gradually CMS1 reaches its configured loadlimit (configured over API at **/system/configuration/cluster**)

3. The user tries to add a new participant to the ongoing adhoc conference, however, the new user does not get connected to the conference

> **Note**: This configuration procedure allows for a user to add participants to an existing CMS adhoc conference even if the CMS server hosting the adhoc conference has reached its load limit, and it can be used until the active control defect is fixed. Active Control becomes disabled in that ad-hoc conference.

Step 1. Create a new SIP Trunk Security Profile for Trunk1

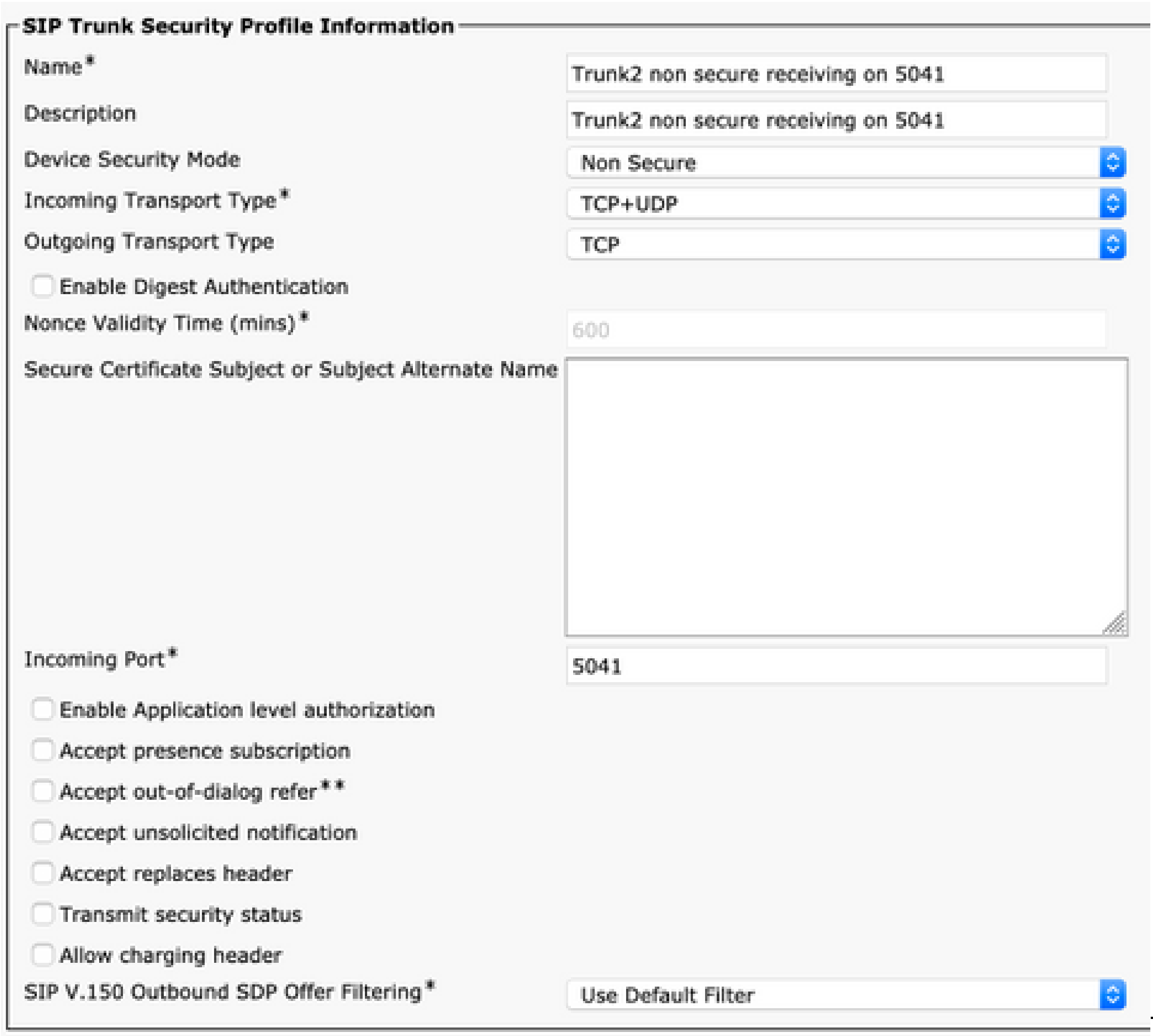
- Navigate to **System** > **Security** > **SIP Trunk Security Profile**
- Select **Add New**
- Set the **Name** to be **Trunk1 non secure receiving on 5040**
- Set the **Device Security Mode** to be **Non secure**
- Set the **Incoming Port** to be **5040**
- Select **Save**



Trunk1 SIP security profile

Step 2. Create a new SIP Trunk Security Profile for Trunk2

- Navigate to **System** > **Security** > **SIP Trunk Security Profile**
- Select **Add New**
- Set the **Name** to be **Trunk2 non secure receiving on 5041**
- Set the **Device Security Mode** to be **Non secure**
- Set the **Incoming Port** to be **5041**
- Select **Save**



Trunk2 SIP se
profile

Step 3. Create a new SIP Normalization Script

- Navigate to **Device** > **Device settings** > **SIP Normalization Scripts**
- Select **Add New**
- Set the **Name** to be **remove_conference_from_call_info_header**
- In the **Content**, use this script

```
M = {}
function M.outbound_INVITE(msg)
    msg:removeHeaderValue("Call-Info", "<urn:x-cisco-remotecc:conference>")
end
return M
```

- Select **Save**

Step 4. Create a new SIP Profile

- Navigate to **Device** > **Device settings** > **SIP profile**
- Select the **Standard SIP Profile For TelePresence Conferencing** and **Copy** it
- Set the **Name** to be **No active control telepresence conferencing**
- Uncheck the **Allow iX Application Media** checkbox at the bottom of the page

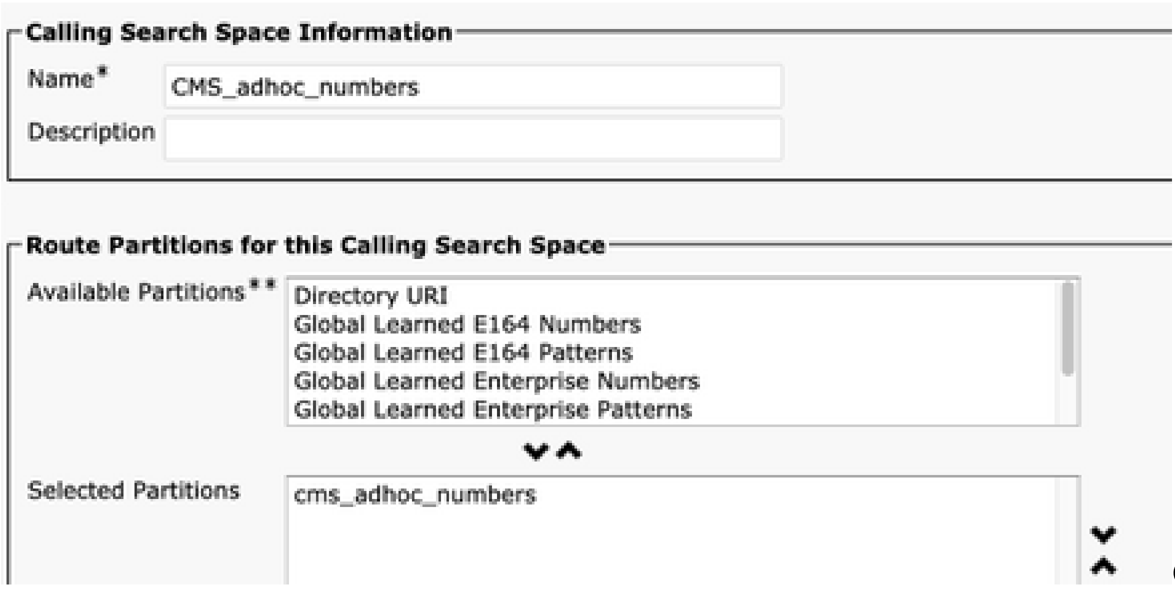
- Select **Save**

Step 5. Create a new Partition

- Navigate to **Call routing** > **Class of Control** > **Partition**
- Select **Add New**
- Set the **Name** to be **cms_adhoc_numbers**
- Select **Save**

Step 6. Create a new Calling Search Space (CSS):

- Navigate to **Call routing > Class of Control > Calling Search Space**
- Select **Add New**
- Set the **Name** to be **CMS_adhoc_numbers**
- Add the partition created in step 5 **cms_adhoc_numbers**
- Select **Save**

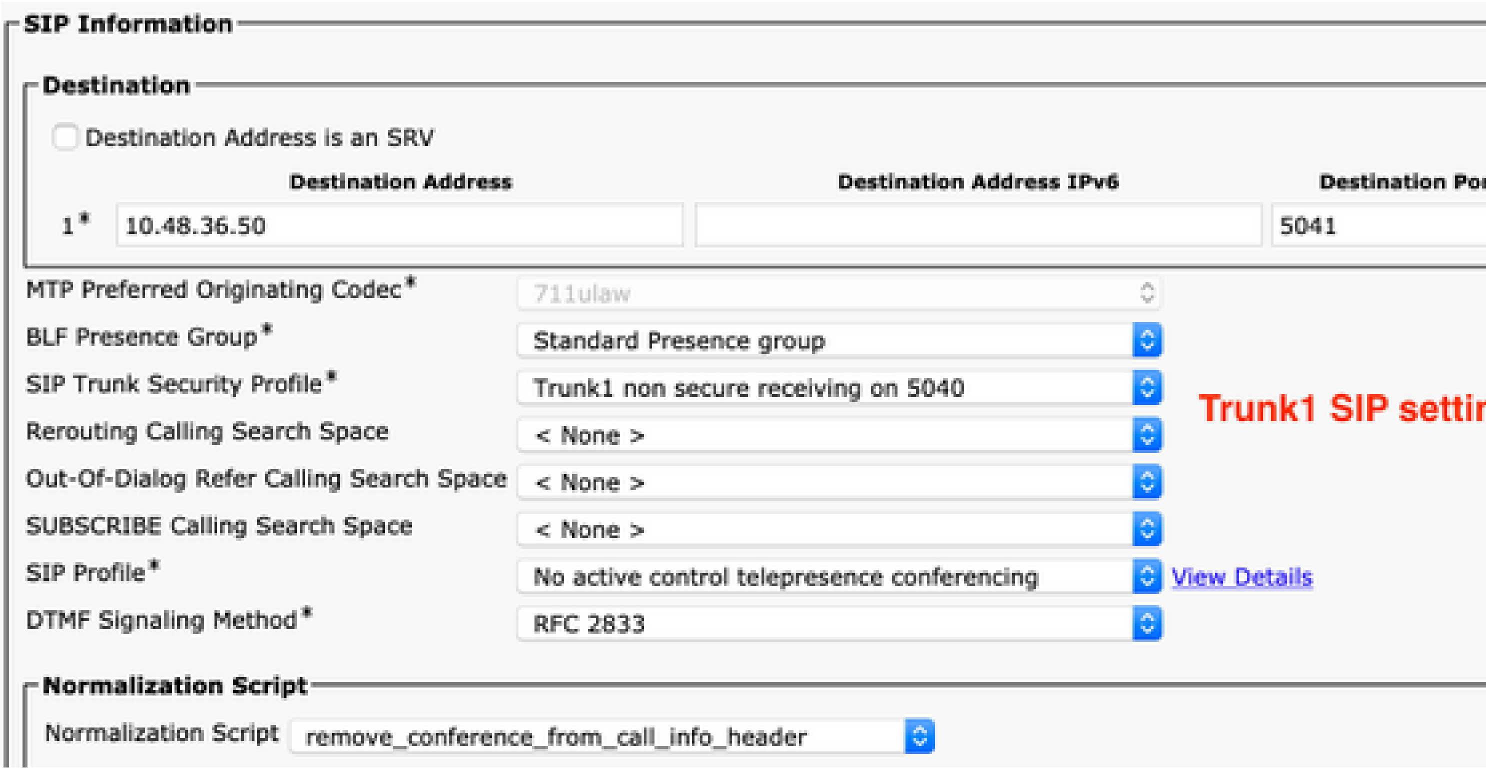


Calling Search Space configuration

Step 7. Create a new SIP trunk, **Trunk1**:

- Navigate to **Device** > **Trunk**
- Select **Add New**
- Select **SIP Trunk** for the **Trunk Type**
- Select **Next**
- Enter these values and **Save**

Device Name	Enter a name for the SIP Trunk, **Trunk1**
Run On All Active Unified CM Nodes	Checked
Destination Address	Enter the IP of the CUCM server itself, for example **10.48.36.50**
Destination Port	Enter the port on which Trunk2 listens on, **5041**
SIP Trunk Security Profile	Select the Profile created in step 1, **Trunk1 non secure receiving on 5040**
SIP Profile	Select the profile created in step 4, **No active control telepresence conferencing**
DTMF Signaling Method	Select **RFC 2833**
SIP Normalization script	Select the script created in step 3, **remove_conference_from_call_info_header**

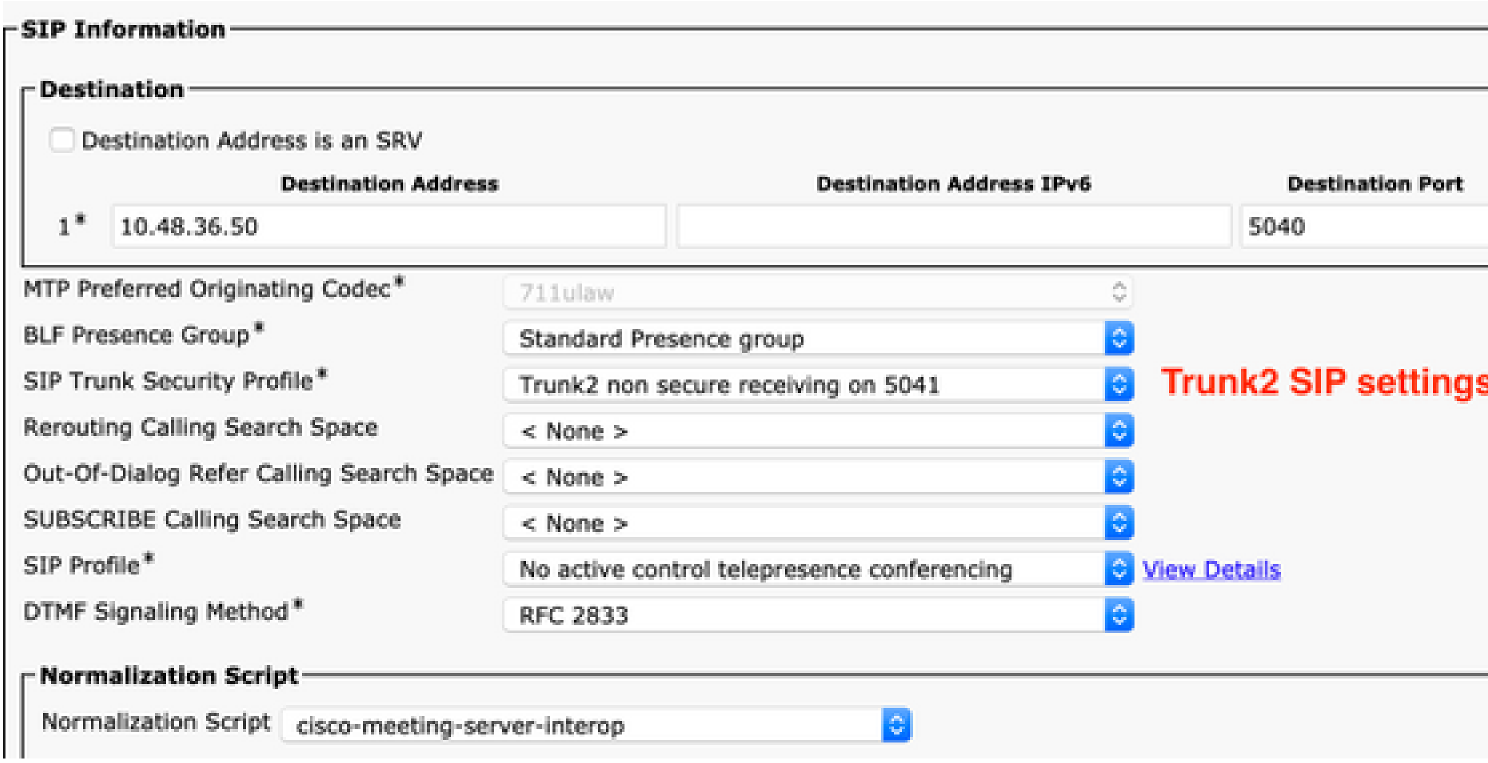


Trunk1 SIP settings

Step 8. Create a new SIP trunk, **Trunk2**:

- Navigate to **Device** > **Trunk**
- Select **Add New**
- Select **SIP Trunk** for the **Trunk Type**
- Select **Next**
- Enter these values and **Save**

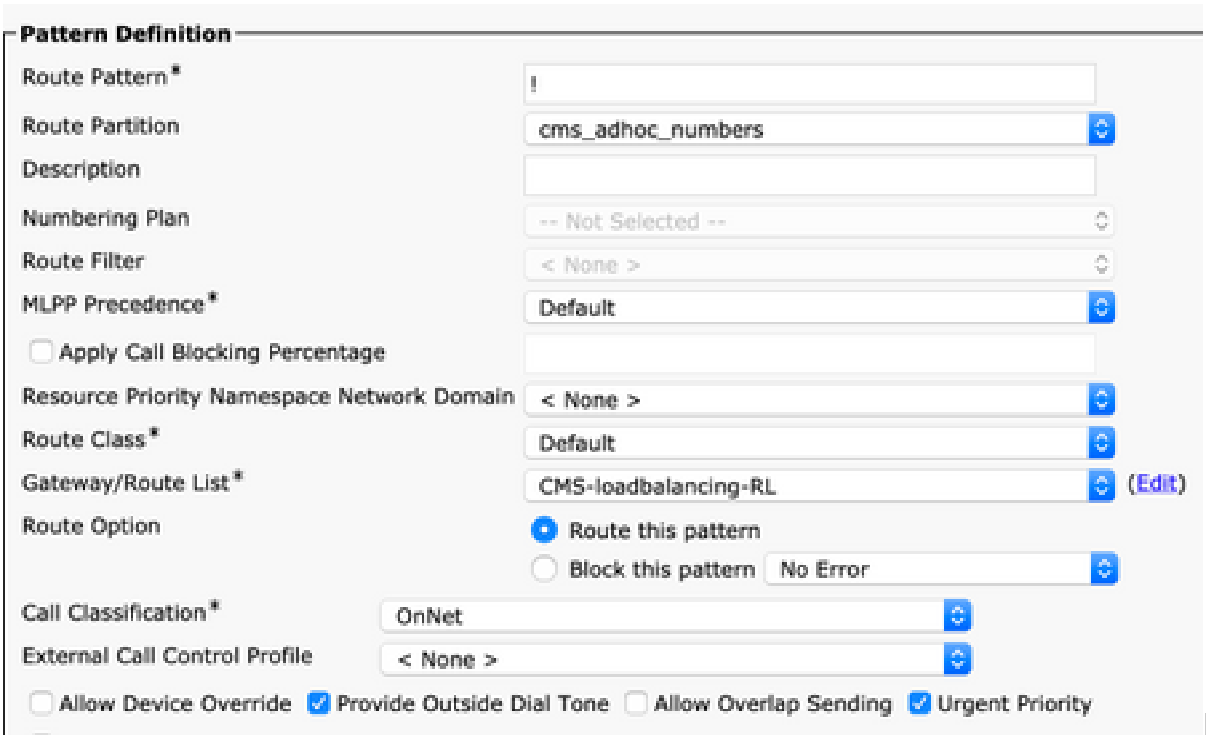
Device Name	Enter a name for the SIP Trunk, **Trunk2**
Run On All Active Unified CM Nodes	Checked
Calling Search Space	Select the CSS created in step 6, **CMS_adhoc_numbers**
Destination Address	Enter the IP address or FQDN of the CUCM server itself, for example **10.48.36.50**
Destination Port	Enter the port on which Trunk1 listens on, **5040**
SIP Trunk Security Profile	Select the Profile created in step 2, **Trunk2 non secure receiving on 5041**
SIP Profile	Select the profile created in step 4, **No active control telepresence conferencing**
DTMF Signaling Method	Select **RFC 2833**
SIP Normalization script	Select the existing normalization script **cisco-meeting-server-interop**



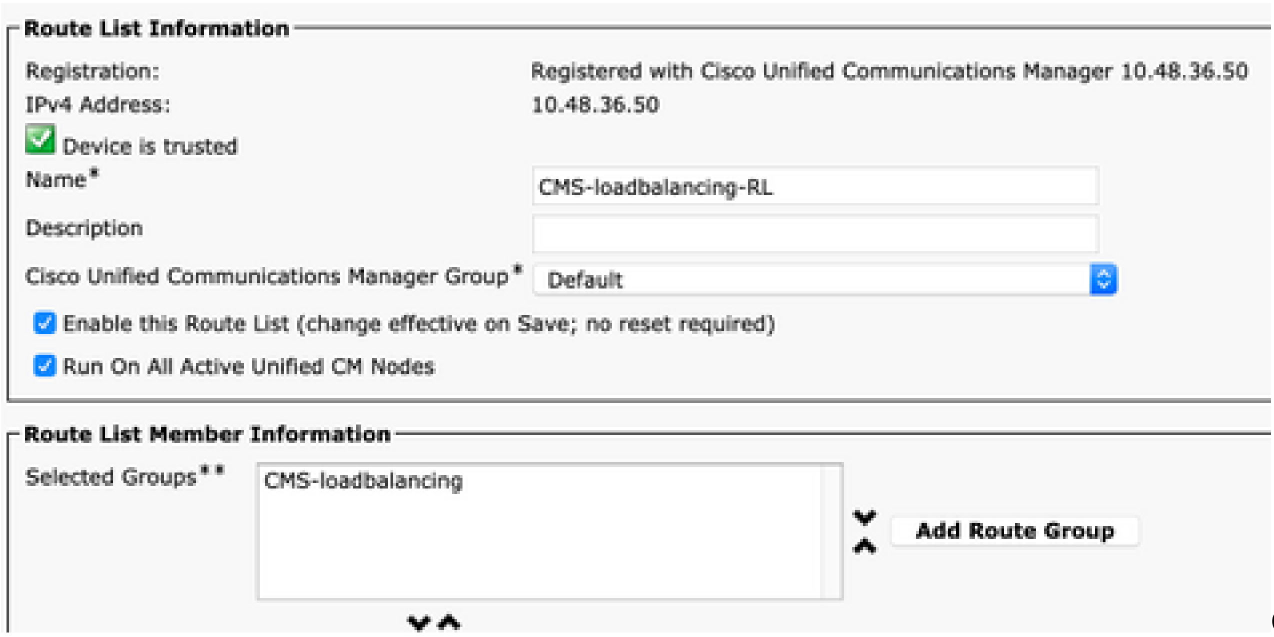
unk2 SIP settings

Step 9. Create a new Route Pattern

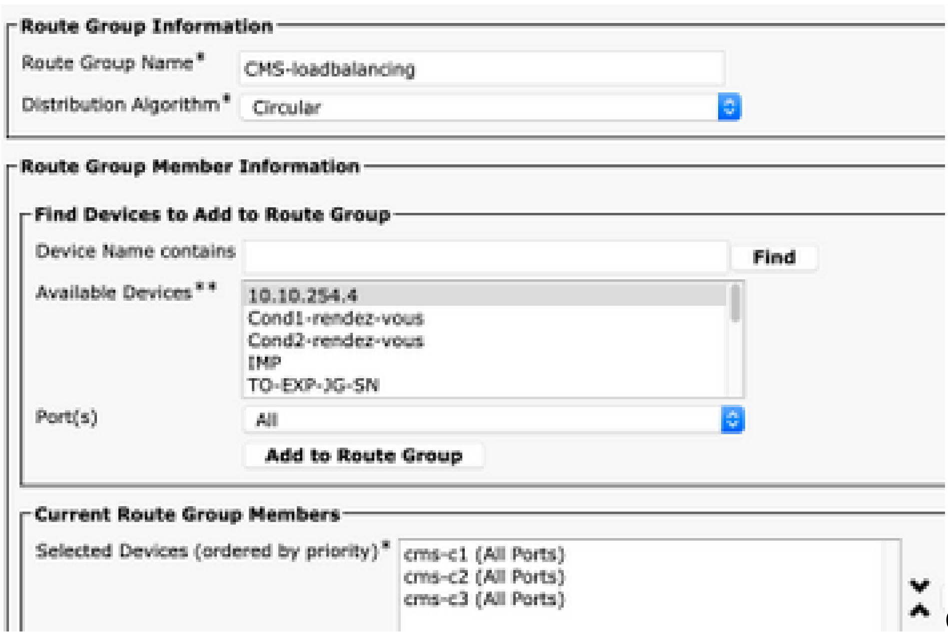
- Navigate to **Call routing** > **Route/Hunt** > **Route Pattern**
- Select **Add New**
- Set the **Route Pattern** to **!**
- Set the **Route Partition** to the partition created in Step 5, **cms_adhoc_numbers**
- Enable the checkbox **Urgent Priority**
- Change **Call Classification** to **OnNet**
- Set the **Gateway/Route List** to be the CMS Route List that is already configured (as mentioned in Requirments section earlier)
- Select **Save**



Route pattern



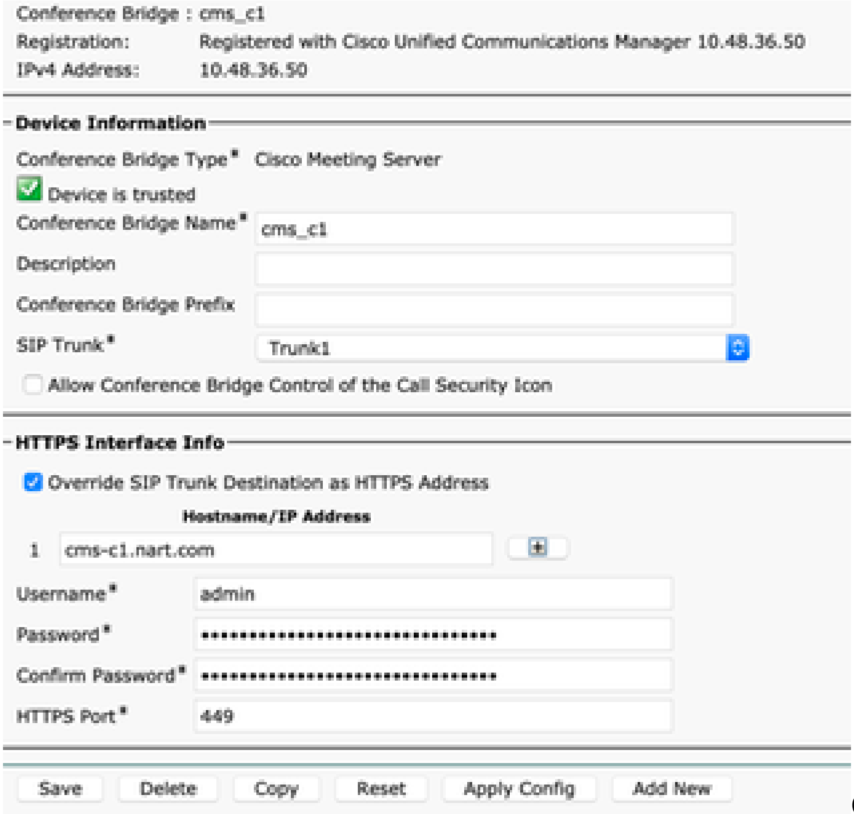
CMS loadbalan
Route list



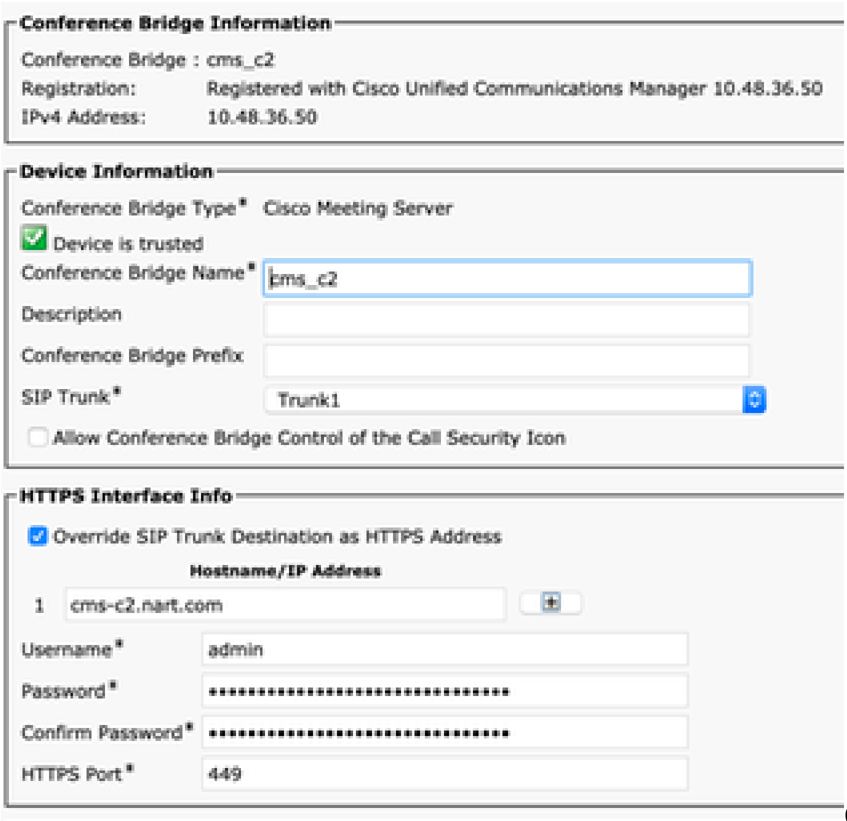
CMS loadbalancing route group

Step 10. Modify the CMS adhoc Conference Bridge configuration

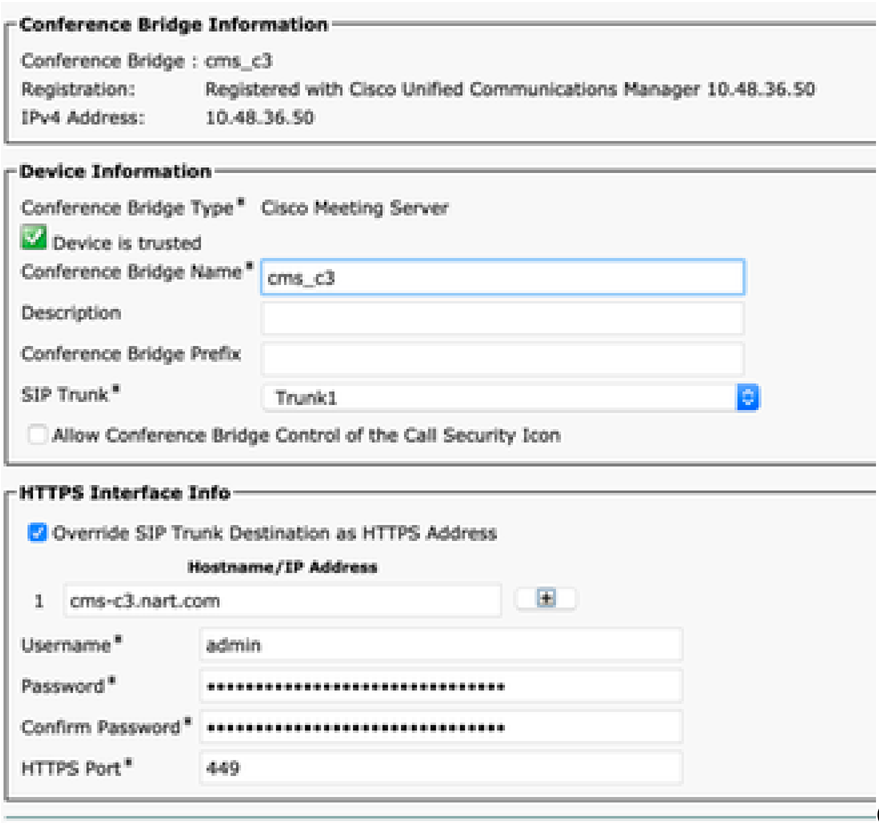
- Navigate to **Media resources** > **Conference bridge**
- Select the first CMS server
- Change the **SIP Trunk** to **Trunk1**, the SIP trunk created in step 7
- Enable the checkbox **Override SIP Trunk Destination as HTTPS Address**
- In the **Hostname/IP Address** field, set the CMS Webadmin **FQDN** for that specific CMS server which must also exist in the Webadmin certificate of that server
- Select **Save**
- Do the same for all other CMS servers, set **Trunk1** to be used on all of them, however change the **Hostname/IP Address** field to the specific **CMS FQDN**



CMS1



CMS2



CMS3

Step 11. Reset SIP trunks **Trunk1** and **Trunk2**

- Navigate to **Device** > **Trunk**
- Select **Trunk1** and **Trunk2**
- Select **Reset selected**
- Wait until both are showing **Full service**

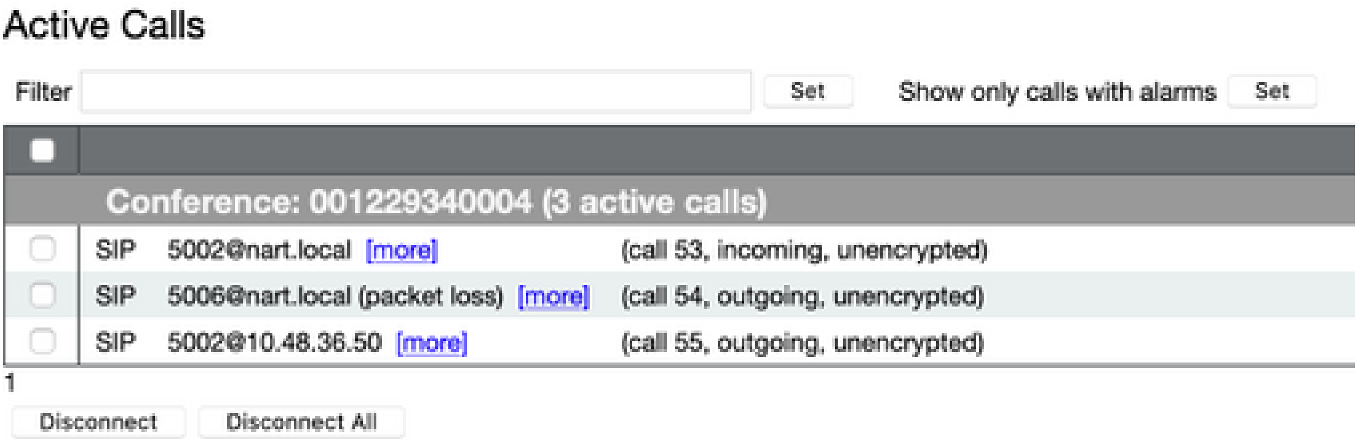
Step 12. Reset CMS adhoc servers

- Navigate to **Media resources** > **Conference bridge**
- Select all CMS servers
- Select **Reset selected**
- Wait until all server are showing **Registered**

Verify

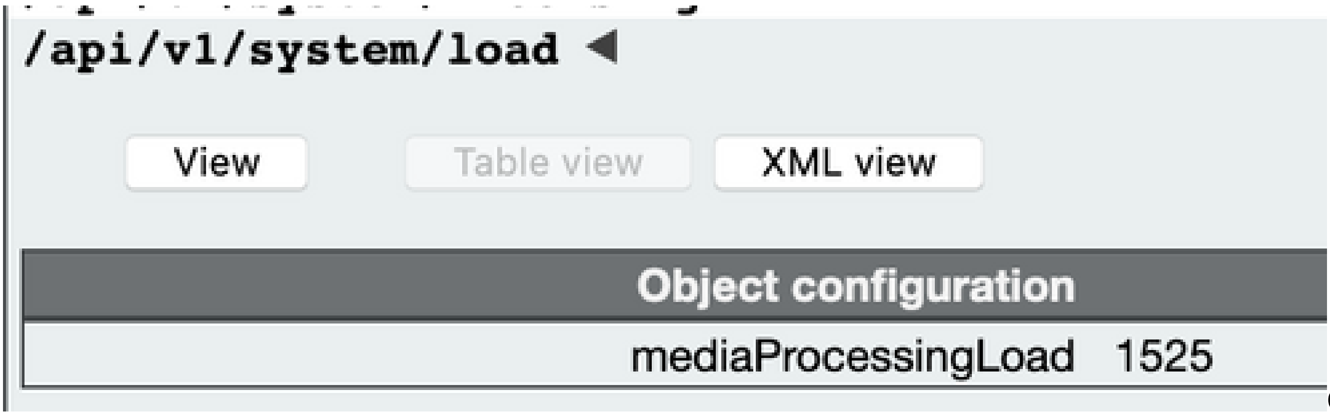
Use this section in order to confirm that your configuration works properly.

- Create an Adhoc conference and check which CMS server is hosting the conference



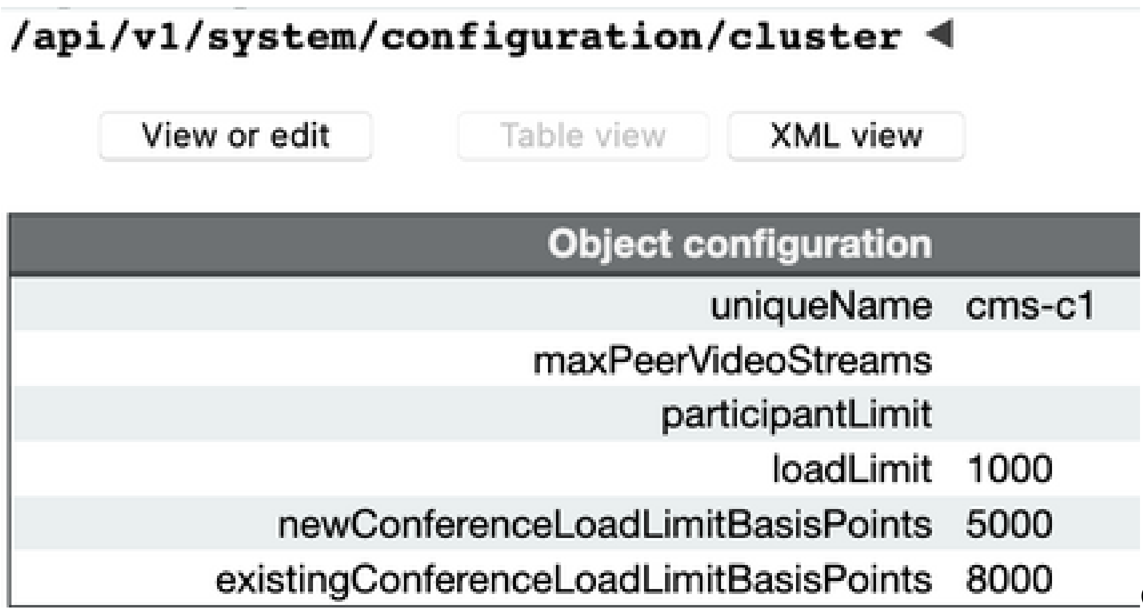
CMS1 h
the adhoc conference

- Check the current **media processing load** on that CMS server, use an **API GET** to **/system/load**



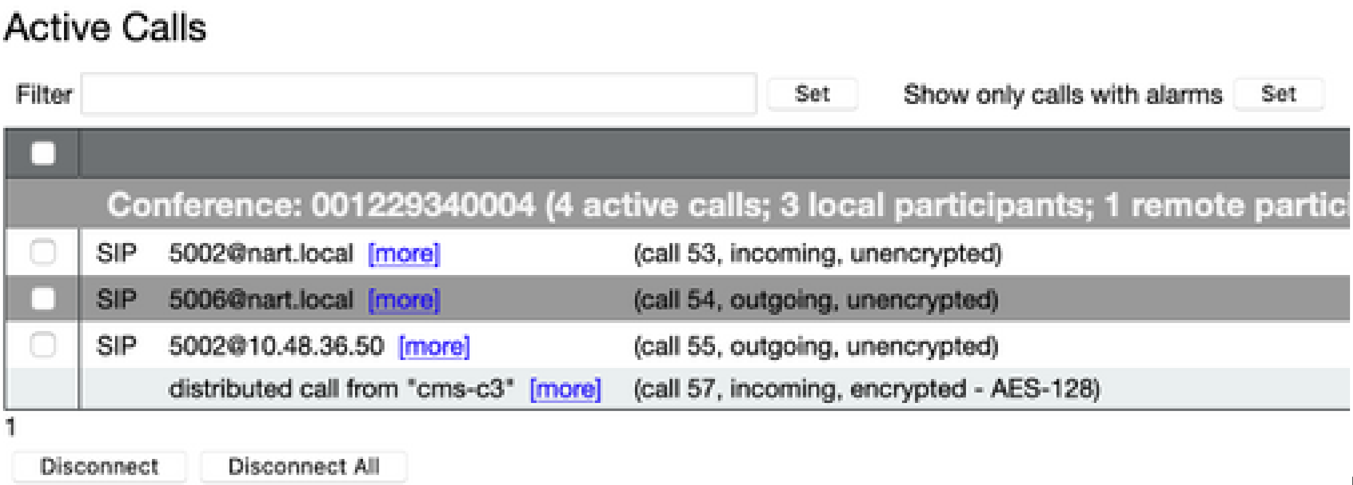
Current me
load

- Set the **load limit** on the server to a value that is lower than the **media processing load** by sending a **POST** to **/system/configuration/cluster** with the paramter **loadlimit**, for example **1000**



Chaning the loadlimit

- Add a new participant to the meeting. The participant gets added and a distributed is created between CMS1 and another CMS server since CMS1 has reached its limit



Distribute

Troubleshoot

There is currently no specific troubleshooting information available for this configuration.

You can use the [Collaboration Solutions Analyser] tool for log analysis.

Related Information

- Load Balancing Logic on Cisco Meeting Server
- CMS configurational documentation
- CMS API and MMP programming guides
- CUCM configurational documentation