Cisco Meeting Server

Deployments with Third Party Call Control

January 02, 2019
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## Change History

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<th>Date</th>
<th>Change Summary</th>
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<tr>
<td>January 02, 2019</td>
<td>No change for version 2.5. Removed version from title.</td>
</tr>
<tr>
<td>July 24, 2018</td>
<td>No change for Cisco Meeting Server 2.4</td>
</tr>
<tr>
<td>May 8, 2017</td>
<td>No change.</td>
</tr>
<tr>
<td>December 20, 2016</td>
<td>No change.</td>
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<tr>
<td>August 03, 2016</td>
<td>Rebranded for Cisco Meeting Server 2.0</td>
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1 Introduction

The Cisco Meeting Server software can be hosted on specific servers based on Cisco Unified Computing Server (UCS) technology as well as on the Acano X-Series hardware, or on a specification-based VM server. Cisco Meeting Server is referred to as the Meeting Server throughout this document.

**Note:** The term Meeting Server in this document means either a Cisco Meeting Server 1000, an Acano X-Series Server or the software running on a virtual host.

This document provides examples of how to configure the Meeting Server to work with third party call control devices from Avaya and Polycom. The examples may need to be adapted according to your specific deployment. These instructions apply equally to all Meeting Server deployment topologies (single server and scaled/resilient deployments).

A separate guide details how to deploy the Meeting Server with Cisco Unified Communications Manager, see [Cisco Meeting Server with Cisco Unified Communications Manager Deployment Guide](#).

1.1 How to Use this Guide

This guide is part of the documentation set (shown in Figure 1) for the Meeting Server.
Deployments with Third Party Call Control: Cisco Meeting Server

Figure 1: Cisco Meeting Server documentation set
1.1.1 Commands

In this document, commands are shown in **black** and must be entered as given – replacing any parameters in <> brackets with your appropriate values. Examples are shown in **blue** and must be adapted to your deployment.

1.1.2 Terminology

Throughout this document the conferencing types mentioned are those as defined in Table 1.

**Table 1: Conferencing Types**

<table>
<thead>
<tr>
<th>Conference type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Rendezvous (also known as personal CMR)</td>
<td>Pre-defined, permanently available addresses that allow conferencing without previous scheduling. The host shares the address with other users, who can call in to that address at any time.</td>
</tr>
<tr>
<td>Ad hoc</td>
<td>Instant or escalated conferencing, for example manually escalated from a point-to-point call to a multiparty call with three or more participants.</td>
</tr>
<tr>
<td>Scheduled</td>
<td>Pre-booked conferences with a start and end time.</td>
</tr>
</tbody>
</table>
2 Configuring a SIP Trunk to an Avaya CM

This appendix provides an example of setting up a SIP trunk between the Cisco Meeting Server and the Avaya Communications Manager (Avaya CM) and may need to be adapted.

**Note:** If you are not your organization's Avaya CM administrator, then Cisco strongly advises you to seek the advice of your local administrator on the best way to implement the equivalent on your server's configuration.

**Note:** Avaya CM is an Avaya PBX, so calls will be audio only, however, the Cisco Meeting Server does not impose this restriction on interoperability with Avaya: therefore a call defined to be type 'avaya' in the Meeting Server does not imply that the call is audio-only.

### 2.1 Configuration Summary

This example deployment assumes that:
- This audio connection between Avaya CM and the Meeting Server is accessed via dialing a prefix 49
- The assigned IVR digits for the Meeting Server are 8320; that is a user from the Avaya environment will dial 498320 to access the Meeting Server IVR
- A DID extension 5328 to route to this same number and allow for PSTN dial-in to the Meeting Server
- Avaya Software Version: CM6 R016x.00.1.510.1 Update: 19940

### 2.2 Cisco Meeting Server Configuration

1. Log in to the Web Admin Interface and go to **Configuration > General**.
2. For IVR Numeric ID, enter 8320.

```
<table>
<thead>
<tr>
<th>IVR</th>
</tr>
</thead>
<tbody>
<tr>
<td>IVR numeric ID</td>
</tr>
</tbody>
</table>
```

These digits will be passed from the Avaya CM to the Meeting Server, and then routed to the Meeting Server IVR.

3. Click **Submit**.
4. Go to **Configuration > Outbound Calls**.
5. Add a dial plan entry for the Avaya CM – see the example below.

The highlighted IP address below matches the C-LAN or Processor Ethernet address on the CM side and represents the CM interface used in the Signaling Group created later.

![Outbound calls table]

6. Click **Add New**.

### 2.3 Avaya CM Configuration

1. Add a node name for the Meeting Server signaling interface.

   ![IP NODE NAMES]

2. Add an Avaya Signaling Group with the following:
   - Group Type = SIP
   - Near-end Node Name = C-LAN or Processor Ethernet interface indicated in the dial plan setting in the previous section
   - Far-end Node Name = Node name for the Meeting Server signaling interface created above.
   - Port settings for both Near-end and Far-end = 5060
   - Far-end Domain = SIP domain associated with the Meeting Server
   - Direct IP-IP Audio Connections = n. This ensures that all traffic from the Avaya CM comes from the Near-end Node
3. Add an Avaya Trunk Group with the following:
   - Group Type = SIP
   - Direction = two way
   - Service Type = tie
   - Additional settings may vary, but see the examples below for possible configuration
### Configuring a SIP Trunk to an Avaya CM

**Group Type:** sip

**TRUNK PARAMETERS**

- **Auto Page Line Retrieval?** n
- **Unicode Name:** auto
- **Redirect On OPTIM Failure:** 5000
- **SCCNP?** n
- **Digital Loss Group:** 18
- **Preferred Minimum Session Refresh Interval (sec):** 600
- **Disconnect Supervision - In?** y
- **Out?** y
- **XDIP Treatment:** auto
- **Delay Call Setup When Accessed Via 1GAR?** n

**TRUNK FEATURES**

- **ACA Assignment?** n
- **Measured:** none
- **Maintenance Tests?** y
- **Numbering Format:** public
- **UUI Treatment:** service-provider
- **Replace Restricted Numbers?** n
- **Replace Unavailable Numbers?** n
- **Modify Tandem Calling Number:** no

- **Show ANSWERED BY on Display?** y
- **DSN Term?** n
PROTOCOL VARIATIONS

Mark Users as Phone? n
Prepend '+' to Calling Number? n
Send Transferring Party Information? n
Network Call Redirection? n
Send Diversion Header? n
Support Request History? y
Telephone Event Payload Type:

Overwrite Calling Identity? n
Convert 180 to 183 for Early Media? n
Always Use re-INVITE for Display Updates? n
Identity for Calling Party Display: P-Asserted-Identity
Enable Q-SIP? n

<table>
<thead>
<tr>
<th>Port</th>
<th>Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>T00001 Cisco</td>
</tr>
<tr>
<td>2</td>
<td>T00002 Cisco</td>
</tr>
<tr>
<td>3</td>
<td>T00003 Cisco</td>
</tr>
<tr>
<td>4</td>
<td>T00004 Cisco</td>
</tr>
<tr>
<td>5</td>
<td>T00005 Cisco</td>
</tr>
<tr>
<td>6</td>
<td>T00006 Cisco</td>
</tr>
<tr>
<td>7</td>
<td>T00007 Cisco</td>
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<tr>
<td>14</td>
<td>T00014 Cisco</td>
</tr>
<tr>
<td>15</td>
<td>T00015 Cisco</td>
</tr>
</tbody>
</table>

4. Add an Avaya Route Pattern to routes calls to trunk group 105 and delete the first two digits (deletes the prefix digits 49).
5. Add a Uniform Dial Plan to provide a routing for a 6-digit number with a prefix of 49. These calls must be set to be routed to AAR tables in Avaya.

6. Add an AAR setting to routes all calls of 6 digits in length and beginning with 49 (i.e. 498320) to route pattern 105 (the Meeting Server Trunk Group).

7. Assign an Extension and DID.

Optionally, in the Uniform Dial Plan you can add a setting for a DID extension (in this example, x5328) to route a call via digits 498320 to the Cisco Systems server.
3 Configuring a Polycom DMA for the Cisco Meeting Server

For calls from a Polycom DMA environment to the Cisco Meeting Server, create an External SIP Peer on the Polycom DMA that will point to the Meeting Server, and then configure a Dial Rule on the Polycom DMA that will direct calls to it.

The following is an example of configuring the Meeting Server for the Polycom DMA, and may need to be adapted. Follow the instructions in the Deployment guides to set up a dial plan rule that points to the Polycom DMA server in the Web Admin Interface Configuration > Outbound Calls page. Also ensure that the correct ports are open (Incoming/Outgoing UDP 32768–65535 – RTP).

Note: If you are not your organization’s Polycom server administrator, then Cisco strongly advises you to seek the advice of your local administrator on the best way to implement the equivalent on your server’s configuration.

3.1 Setting up the External SIP Peer

On the Polycom DMA:

1. Go to Network > External SIP Peer > Add

![External SIP Peer page]

2. In the External SIP Peer page configure the following:
   - Name: Cisco Systems
   - Description: a meaningful phrase, possibly Cisco Systems IP Peer
   - Next hop Address: IP Address of the Meeting Server Call Bridge
3. **Deployments with Third Party Call Control:**

- Port: 5060
- User Route Header: selected
- Type: Other
- Transport Type: TCP

![Image of Edit External SIP Peer](image)

3. Leave the Domain List page blank.

![Image of Edit External SIP Peer](image)

4. In the Postliminary page Header Options section configure the following:
   a. Copy All Parameters: Checked
   b. Format: Use original request's To

5. In the Postliminary page Request URI options section configure the following:
   a. Format: Original user, configured peer’s Destination Network or next hop address
6. In the Authentication page configure the following:
   a. Authentication: Pass authentication
   b. Proxy authentication: Pass Proxy authentication

7. Click **Save**.

### 3.2 Creating the Dial Rule

In the Polycom DMA:
1. Go to **Admin > Call Server > Dial Rules > Add**.

2. In the Edit Dial Rule for Authorized Calls page, configure the following (see below):
   a. Description: Cisco <Description of pattern>

3. Select **Enabled**.

4. Select the Cisco Systems SIP Peer in the left pane and click the arrow to move it to the Selected SIP Peers.

5. In the Preliminary page create a string to represent how calls will match this rule (see below).
Consult the DMA Admin Guide for more detail. The example below matches any call that begins with a 6 and sends it to the [[Undefined variable BrandingTypeVariables.solution or server]].

```java
if(!DIAL_STRING.match(/sip:6/))
{
    return NEXT_RULE;
}
```

6. Click **OK**.

You should now be able to dial from any SIP-enabled Polycom DMA endpoint to the Cisco Meeting Server using the rule created.
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