Cisco Meeting Server

Deployments with Third Party Call Control

January 02, 2019

Cisco Systems, Inc. www.cisco.com

Contents

Change History

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

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.

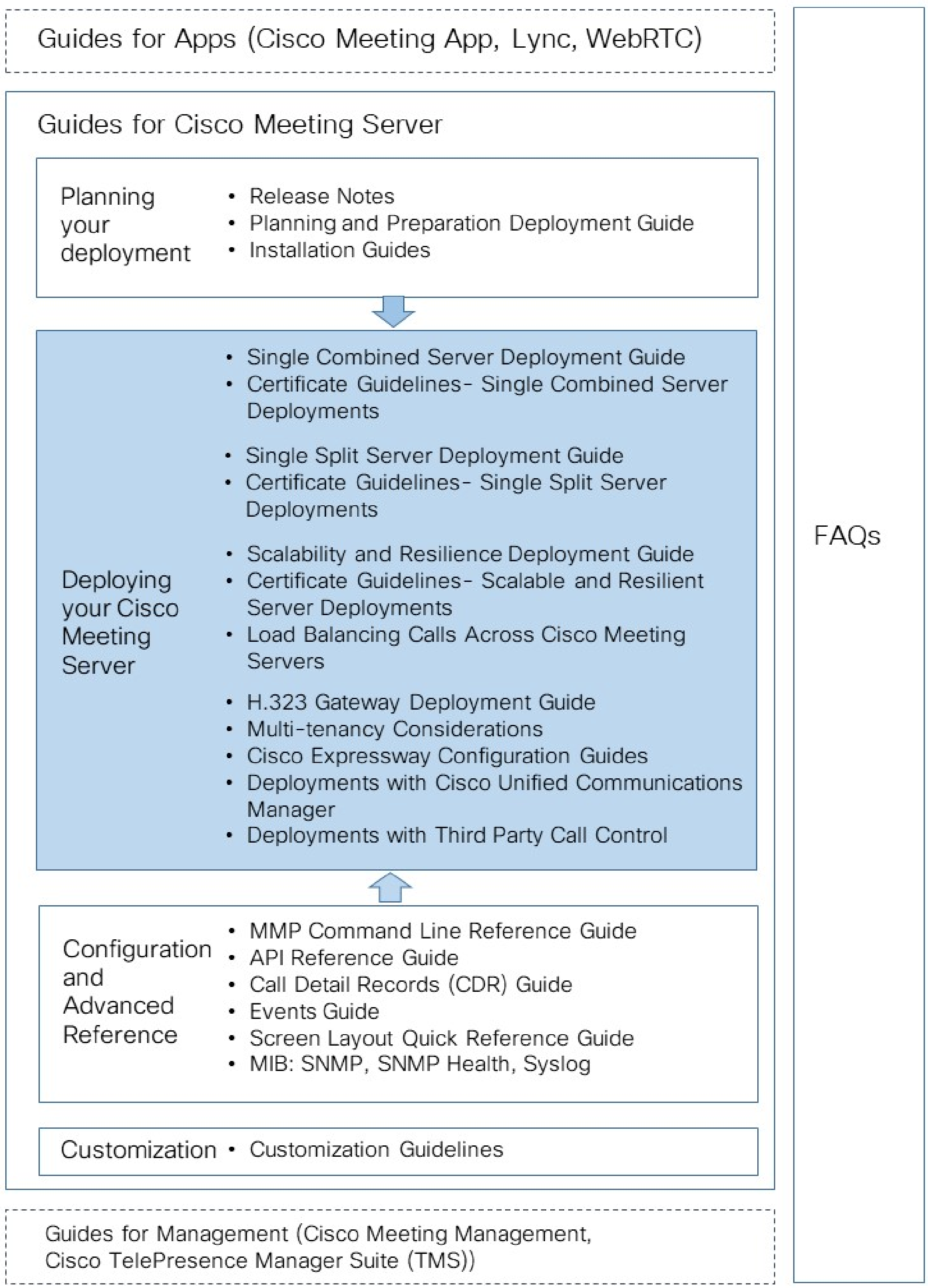
This document provides examples of how to configure the Meeting Server to work with third party call control devices from Avaya and Polycom. The examples may need to be adapted according to your specific deployment. These instructions apply equally to all Meeting Server deployment topologies (single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

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

Conference type	Description
Rendezvous (also known as personal CMR)	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.
Ad hoc	Instant or escalated conferencing, for example manually escalated from a point-to-point call to a multiparty call with three or more participants.
Scheduled	Pre-booked conferences with a start and end time.

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.

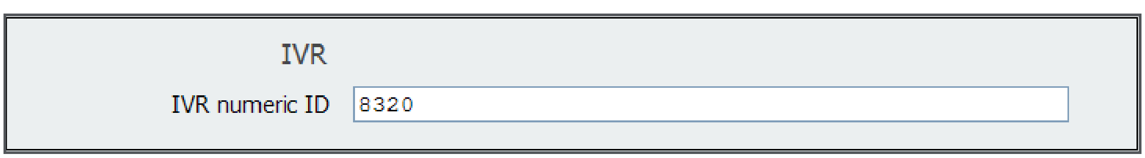
2.1 Configuration Summary

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 theMeeting 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.



   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

Filter Submit

	Domain	SIP proxy to use	Local contact domain	Trunk type	Behavior	Priority	Encryption	
	lync. example.com	<none; call directly>	example.com	Standard SIP	Stop	2	Auto	[edit]
	<match all domains>	10.1.1.77	example.com	Standard SIP	Stop	1	Auto	[edit]
	avaya. example.com	192.168.20.103	example.com	Avaya	Stop	1	Auto	[edit]
				Standard SIP	Stop	0	Auto	Add New Reset

6. Click **Add New**.

2.3 Avaya CM Configuration

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

IP NODE NAMES

Name	IP Address
Cisco	192.168.110.51
App1-AAM	10.22.4.38

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

```
                              SIGNALING GROUP

 Group Number: 105            Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                          SIP Enabled LSP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: Others



   Near-end Node Name: clan-1a11              Far-end Node Name: Cisco
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         Far-end Network Region: 1
                                    Far-end Secondary Node Name:
Far-end Domain: mycompany.com
                                              Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                   RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? n
Session Establishment Timer(min): 3                      IP Audio Hairpinning? y
         Enable Layer 3 Test? y
                                                    Alternate Route Timer(sec): 6
```

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

```
                              TRUNK GROUP

Group Number: 105                    Group Type: sip           CDR Reports: y
  Group Name: Cisco                         COR: 1        TN: 1        TAC: 175
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                      Night Service:
Queue Length: 0
Service Type: tie                      Auth Code? n
                                            Member Assignment Method: auto
                                                     Signaling Group: 105
                                                   Number of Members: 24
```

```
      Group Type: sip

TRUNK PARAMETERS                                       Auto Page Line Retrieval? n

     Unicode Name: auto

                                                      Redirect On OPTIM Failure: 5000

            SCCAN? n                                         Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 600

 Disconnect Supervision - In? y  Out? y


             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? 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
```

```
                              TRUNK GROUP
                                  Administered Members (min/max):   1/24
GROUP MEMBER ASSIGNMENTS                 Total Administered Members:  24

      Port              Name
  1: T00001             Cisco
  2: T00002             Cisco
  3: T00003             Cisco
  4: T00004             Cisco
  5: T00005             Cisco
  6: T00006             Cisco
  7: T00007             Cisco
  8: T00008             Cisco
  9: T00009             Cisco
 10: T00010             Cisco
 11: T00011             Cisco
 12: T00012             Cisco
 13: T00013             Cisco
 14: T00014             Cisco
 15: T00015             Cisco
```

4. Add an Avaya Route Pattern to routes calls to trunk group 105 and delete the first two digits (deletes the prefix digits 49).

```
                    Pattern Number: 105 Pattern Name: Cisco
                             SCCAN? n      Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                          DCS/ IXC
    No          Mrk Lmt List Del  Digits                            QSIG
                             Dgts                                   Intw
 1: 105  0                    2                                      n   user
 2:                                                                  n   user
 3:                                                                  n   user
 4:                                                                  n   user
 5:                                                                  n   user
 6:                                                                  n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                      none
 2: y y y y y n  n            rest                                      none
 3: y y y y y n  n            rest                                      none
 4: y y y y y n  n            rest                                      none
 5: y y y y y n  n            rest                                      none
 6: y y y y y n  n            rest                                      none
```

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.

```
48                   6    0                  aar  n
49                   6    0                  aar  n
5004                 4    4    5316          ext  y
```

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).

```
          Dialed            Total     Route    Call    Node   ANI
          String           Min  Max  Pattern   Type    Num    Reqd
     49                    6    6     105      aar            n
     5                     7    7     999      aar            n
```

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.

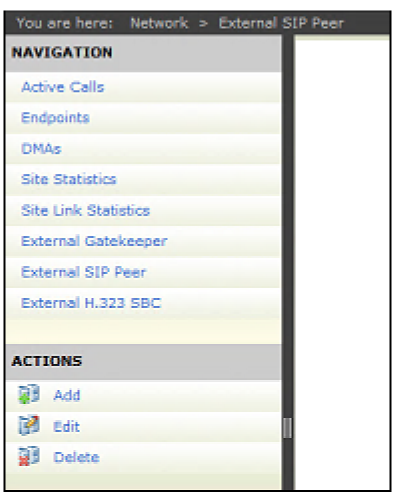
The following is an example of configuring theMeeting Server for the Polycom DMA, and may need to be adapted. Follow the instructions in the Deployment guides to set up a dial plan rule that points to the Polycom DMA server in the Web Admin Interface **Configuration > Outbound Calls** page. Also ensure 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

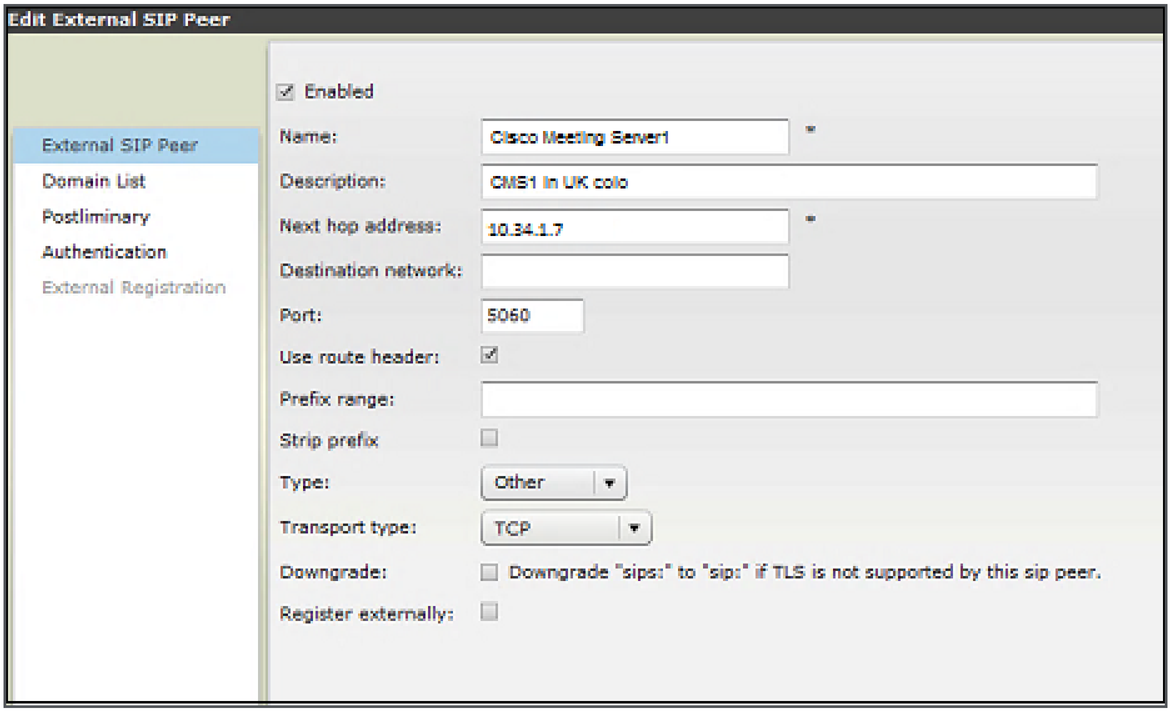
On the Polycom DMA:

1. Go to **Network > External SIP Peer > Add**

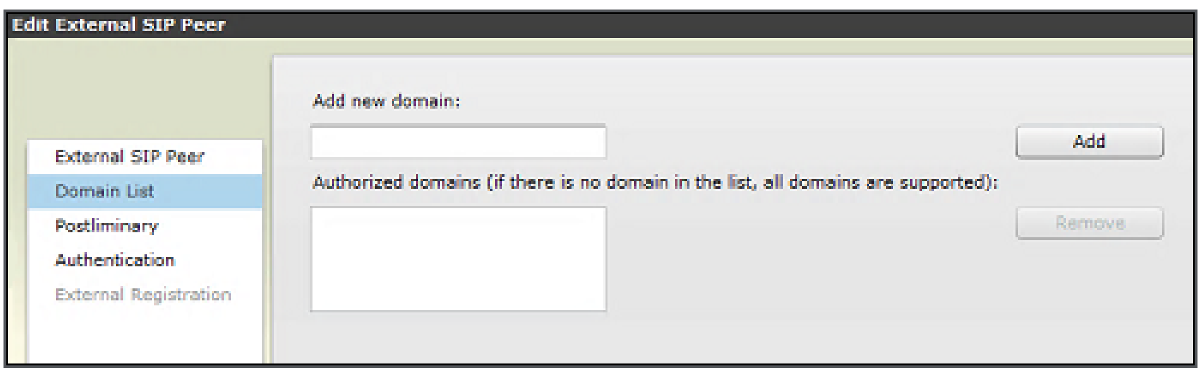


2. In the External SIP Peer page configure the following:
   - Name: Cisco Systems
   - Description: a meaningful phrase, possibly Cisco Systems IP Peer
   - Next hop Address: IP Address of the Meeting Server Call Bridge

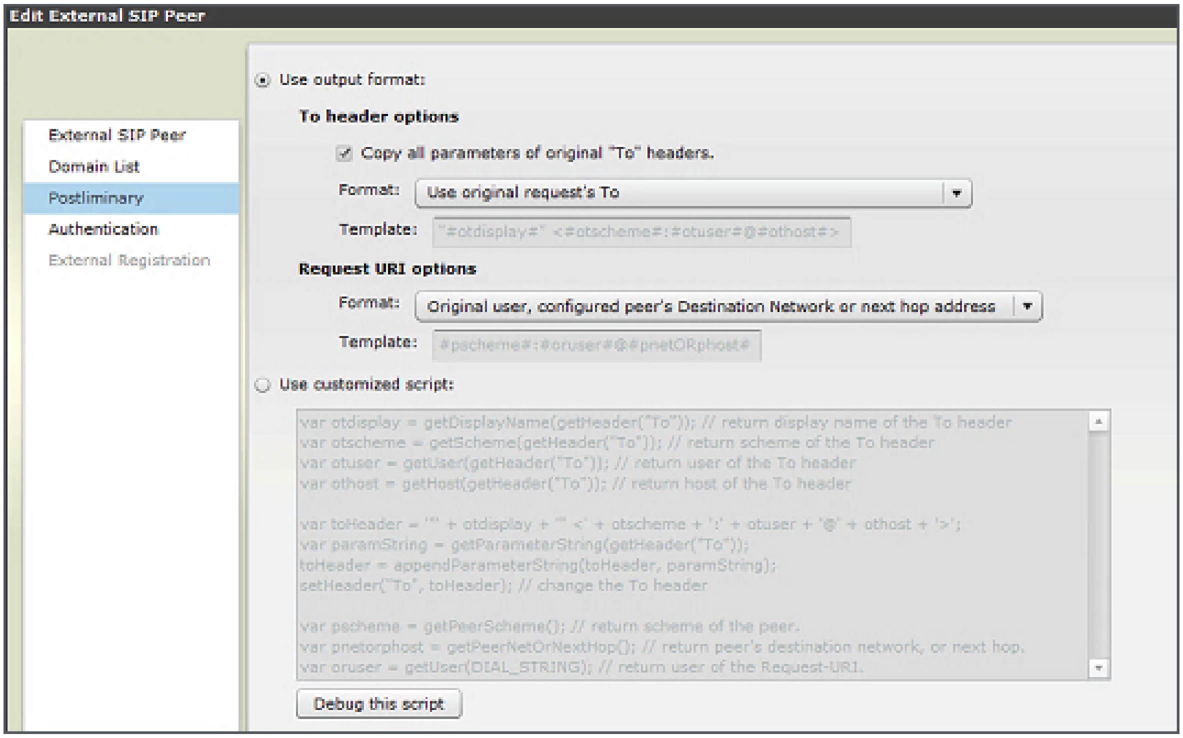
- Port: 5060
- User Route Header: selected
- Type: Other
- Transport Type: TCP



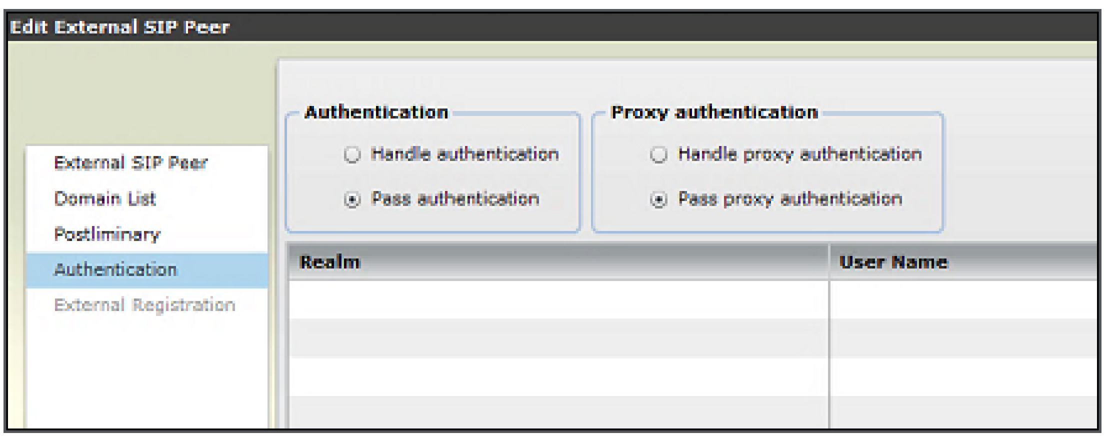
3. Leave the Domain List page blank.



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

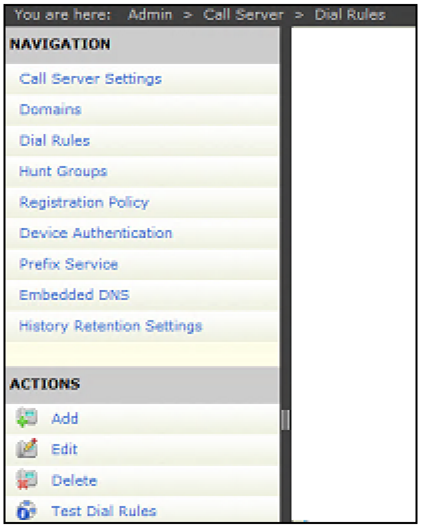


7. Click **Save**.

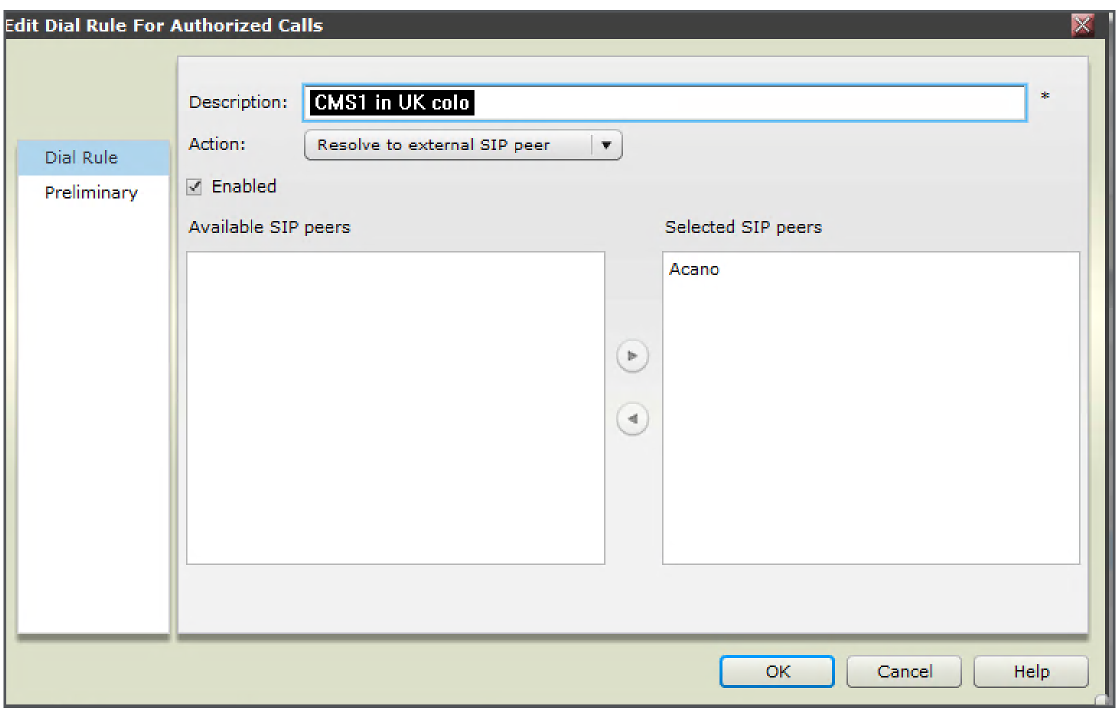
3.2 Creating the Dial Rule

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.

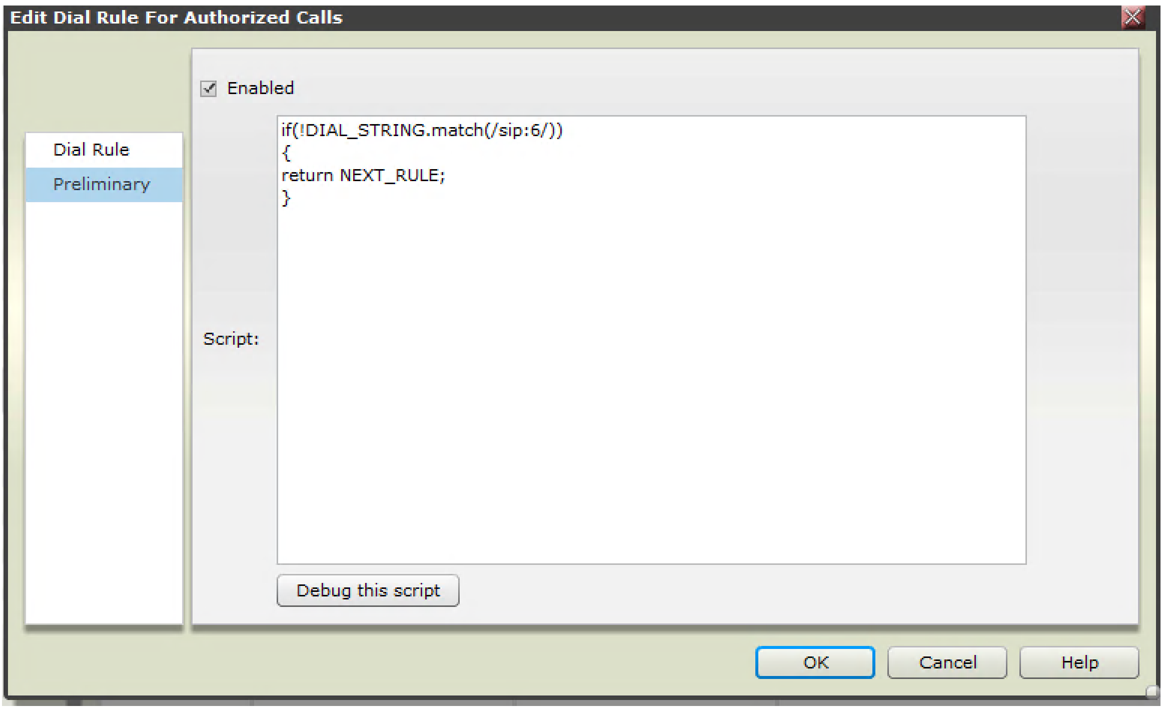


5. In the Preliminary page create a string to represent how calls will match this rule (see below).

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]]].

```
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.

Cisco Legal Information

Cisco Trademark