Pexip Infinity and Polycom DMA

Deployment Guide

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Contents

Introduction

Polycom DMA is a SIP/H.323 registrar and call control device. This guide describes how to integrate the Pexip Infinity solution with a deployment based around Polycom DMA, so that SIP and H.323 calls can be routed to and from the Pexip Infinity platform and endpoints registered to the DMA.

Prerequisites

This guide assumes that you are familiar with DMA.

It assumes that Pexip Infinity and DMA have been deployed successfully, configured with basic settings (such as an IP address, DNS and NTP servers) and are able to route calls. It also assumes that Pexip Infinity has been configured with Virtual Meeting Rooms and associated aliases appropriate to your dial plan. For complete information on how to configure your Pexip Infinity solution, see the Pexip Infinity technical documentation website at docs.pexip.com.

Example deployment scenario

The examples used in this guide assume a dial plan where all Virtual Meeting Room aliases are in one of the following two formats:

- meet.<name>@example.com
- 555<extension>@example.com

It assumes that a Virtual Meeting Room, named **meet.alice**, has been configured with two aliases: **meet.alice@example.com** and **555123@example.com**.

Note that Pexip Infinity supports aliases that include a domain (e.g. **555123@example.com**), as well as aliases without a domain (e.g. just **555123**). To match a received destination alias that includes a domain, the aliases configured within Pexip Infinity must also include the same domain. If an alias configured on Pexip Infinity does not include a domain:

- calls to **<alias>** without any domain portion (e.g. **555123**) will be matched
- calls to **<alias>@<IPaddress>** (e.g. **555123@192.0.2.0**) will be matched
- calls to **<alias>@<domain>** (e.g. **555123@example.com**) will not be matched.

Pexip Infinity configuration

In these steps, you configure Pexip Infinity to use DMA as the SIP proxy and H.323 gatekeeper for outbound calls. It involves:

- Adding a SIP proxy
- Adding an H.323 gatekeeper
- Assigning the SIP proxy and H.323 gatekeeper to a location

Adding a SIP proxy

To add DMA as a SIP proxy:

1. Go to **Call Control > SIP Proxies**.
2. Select **Add SIP proxy**.
3. Complete the following fields:

Name	Enter the name you want to use to refer to this SIP proxy. This example uses **DMA**.
Description	Enter a description of the SIP proxy. This example uses **SIP proxy to DMA**.
Address	Enter the IP address or FQDN of the DMA.
Port / Transport	Depending on your security policy, select either: ◦ **Port** of ***5060*** and **Transport** of ***TCP*** ◦ **Port** of ***5061*** and **Transport** of ***TLS***

4. Select **Save**.

Adding an H.323 gatekeeper

To add DMA as an H.323 gatekeeper:

1. Go to **Call Control > H.323 Gatekeepers**.
2. Select **Add H.323 gatekeeper**.
3. Complete the following fields:

Name	Enter the name you want to use to refer to this H.323 gatekeeper. This example uses **DMA**
Description	Enter a description of the H.323 gatekeeper. This example uses **H.323 gatekeeper to DMA**.
Address	Enter the IP address or FQDN of the DMA.
Port	Leave as the default **1719**.

4. Select **Save**.

Assigning the SIP proxy and H.323 gatekeeper to a location

This is only required if the DMA is the only route for outgoing calls from Pexip Infinity for the location.

To nominate DMA as the SIP proxy and H.323 gatekeeper to be used for outbound calls from a Pexip Infinity location:

1. Go to **Platform > Locations**.
2. Select the location.
3. From the **H.323 gatekeeper** drop-down menu, select the name of the H.323 gatekeeper added earlier (**DMA** in this example).
4. From the **SIP proxy** drop-down menu, select the name of the SIP proxy added earlier (**DMA** in this example).
5. Select **Save**.

Polycom DMA configuration

This section lists the tasks required to configure DMA so that it can be integrated with one Pexip Infinity Conferencing Node in a single location:

- Setting up a SIP peer
- Adding a dial rule

- Setting up an H.323 gatekeeper

This guide is based on Polycom DMA 7000. If you are using other versions of DMA, you will need to perform the same set of tasks, but the menus and options may differ slightly from those described here.

Setting up a SIP peer

In this step we configure a single Pexip Infinity Conferencing Node as a SIP peer in DMA.

1. From the DMA web interface, go to **Network > External SIP Peer**.
2. From the **Actions** panel on the left, select **Add**.

   The **Add External SIP Peer** dialog opens.
3. From the panel on the left, select **External SIP Peer** and complete the following fields:

Enabled	Select this option.
Name	Enter a name for the peer. This example uses **Pexip Infinity Node_1**.
Description	You can optionally enter a description.
Next hop address	Enter the DNS name or IP address of the Pexip Infinity Conferencing Node.
Port	This depends on what you select in the **Transport type** field below: ◦ For TLS, enter **5061** ◦ For TCP, enter **5060**
Use route header	Select this option.
Type	Select ***Other***.
Transport type	Select ***TLS*** (if supported); otherwise select ***TCP***.

4. From the panel on the left, select **Domain List**. This can be left blank.
5. From the panel on the left, select **Postliminary** and complete the following fields:

Use Output format	Select this option.
To header options	
Copy all parameters of original "To" headers	Select this option.
Format	Select ***Use original request's To***.
Request URI options	
Format	Select ***Use original request's URI (RR)***.

6. Select **OK**.

Adding a dial rule

In this step we add a dial rule that routes all calls starting with **meet.** to the Pexip Infinity Conferencing Node that we have just added as a SIP peer.

1. From the DMA web interface, go to **Admin > Call Server > Dial Rules**.
2. From the **Actions** panel on the left, select **Add**.

   The **Add Dial Rule For Authorized Calls** dialog opens.
3. From the panel on the left, select **Dial Rule** and complete the following fields:

Description	Enter a description. This example uses **Route calls starting with "meet." to Pexip Infinity**.
Action	Select ***Resolve to external SIP peer***.
Enabled	Select this option.
Available SIP peers	Select the Pexip Infinity Conferencing Node added earlier and add it to the list of **Selected SIP peers**.

4. From the panel on the left, select **Preliminary** and complete the following fields:

Enabled	Select this option.

Script	Enter a DMA script that will match the calls you want to route to Pexip Infinity. In this example, we want to match all calls starting with **meet.** so we use the following script:
	`if(!DIAL_STRING.match (/sip:meet.*/))`
	`{`
	`  return NEXT_RULE;`
	`}`

5. Select **OK**.

Note that an alternative approach to using a dial rule if, for example, you are not limited to specific call scenarios, is to make use of DMA's default dial plan and to assign a dial string prefix to the SIP peer instead (as per the H.323 gatekeeper example below).

Setting up an H.323 gatekeeper

In this step we configure a single Pexip Infinity Conferencing Node as an H.323 gatekeeper in DMA, and configure it so that all calls starting **555** are routed to that Conferencing Node.

1. From the DMA web interface, go to **Network > External Gatekeeper**.
2. From the **Actions** panel on the left, select **Add**.
   The **Add External Gatekeeper** dialog opens.
3. From the panel on the left, select **External Gatekeeper** and complete the following fields:

Enabled	Select this option.
Name	Enter a name. This example uses **Pexip Infinity Node_1**.
Description	You can optionally enter a description.
Address	Enter the DNS name or IP address of the Pexip Infinity Conferencing Node.
RAS port	Leave as the default **1719**.
Prefix range	Enter the numeric prefix used in your dial plan for Pexip Infinity services. This example uses **555**.
Strip prefix	Leave this box unselected.

4. Select **OK**.

Testing the deployment

To confirm that you have successfully integrated Pexip Infinity and DMA, you need to test that endpoints registered to DMA can make calls to, and receive calls from, the Pexip Infinity platform.

Calls to Pexip Infinity

- From a SIP endpoint registered to DMA, place a call to one of your Pexip Infinity Virtual Meeting Room aliases. Use at least one other endpoint to place a call to the same Virtual Meeting Room. In this example, you would call **meet.alice@example.com**.
- From an H.323 endpoint registered to DMA, place a call to one of your Pexip Infinity Virtual Meeting Room aliases. Use at least one other endpoint to place a call to the same Virtual Meeting Room. In this example, you would call **555123@example.com**.
- Each endpoint should connect to the Virtual Meeting Room and be able to send and receive audio and video from all of the other participants.

Calls from Pexip Infinity

There are a number of ways that Pexip Infinity Conferencing Nodes can be prompted to make outbound calls. For a full list, see Automatically dialing out to a participant from a conference and Manually dialing out to a participant from a conference. For the purposes of this test, we will place the call manually using the Administrator interface, as follows:

1. From the Pexip Infinity Administrator interface, go to **Services > Virtual Meeting RoomS** and select the name of the Virtual Meeting Room from which you want to place the call. In this example we select **meet.alice**.
2. At the bottom left of the screen, select **Dial out to participant**.
3. Complete the following fields:

Field	Description
System location	Select the system location to which the Conferencing Node that you added as a SIP peer/H.323 gatekeeper belongs.
Service alias	This lists all of the aliases that have been configured for the selected Virtual Meeting Room or Virtual Auditorium. The participant will see the incoming call as coming from the selected alias.
Destination alias	The alias of the endpoint that you want to dial.
Protocol	Select either ***SIP*** or ***H.323***, depending on which protocol you wish to test.

4. Select **Dial out to participant**.

The call should be received by the destination endpoint, with the call showing as coming from the selected alias. On answer, the endpoint should connect to the selected Virtual Meeting Room and be able to send and receive audio and video from all of the other participants.

Limitations

Known limitations between Pexip Infinity and Polycom DMA:

- Starting and stopping a presentation many times may stop a Polycom HDX endpoint from sending the presentation stream.