A MITEL
PRODUCT
GUIDE

OpenScape Solution Set V11

Alphanumeric SIP URI Configuration Guide

Administrator Documentation

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Mitel

Contents

1 Minimum version requirements

OpenScape Alphanumeric SIP URI feature is released as PSR with the minimum versions for the below products:

- OSV: V10R1.7.5
- SBC: V10 R1.01.00 10.01.01.00-2
- UC: V10 R2.1.0 040100
- Fusion Client: V2.R1.1.0 Build 102

Product Management must be contacted to request a Project Specific Release (PSR) to use OpenScape Alphanumeric SIP URI.

2 Introduction to Alphanumeric SIP URI Dialing

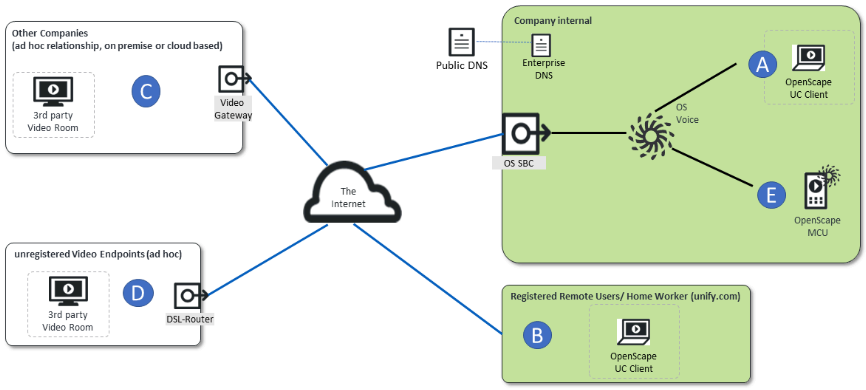
Foreign Peer Domains enabled

Alphanumeric SIP URI dialing, with Foreign Peer Domains enabled, allows users to attend a video conference which is organized from another company and they cannot not invite other companies to participate to their conferences. An invitation to a video conference always provides a SIP URI string to give the participants the possibility to join the conference using the company-preferred video client and avoid using other web access links or applications. Routing based functionality on the entire SIP URI rather than just the userinfo part of the SIP URI is especially useful when the Request URI of an incoming SIP request does not contain numbers within the domain for which the OpenScape Voice is responsible or contains alphanumeric characters in the userinfo part of the URI. This feature supports the two following cases:

- Video dial in: From internet to OpenScape UC conference server - to invite other companies to attend OS UC video conferences
- Video dial out: Using OS UC client for video call - to join video conferences from other companies (routed through internet)

The following figure illustrates the Video SIP-URI architecture

Alphanumeric SIP URI for Video



Third party video endpoints, behind cloud destinations, can place or accept video calls and reach OpenScape System through the internet. The SIP URI concept is duplicated for other UCC call controls (for example OS 4000 with OS UC). A simple case is where an internal participant plans a UC Video conference and invites external guests by sending a SIP URI address and PIN. Another simple case is when an incoming video call, only one SIP URI address is published for dial-in for security reasons and screening, is routed to the OS UC MCU/ video conference portal for further processing where the user must enter the conference PIN to join.

Peer Domains enabled

Alphanumeric SIP URI dialing, with Peer Domains enabled, allows users to attend a video conference which is organized from their own company by using intranet. This feature supports the following two cases:

- Video dial in: From intranet to OpenScape UC conference server

- Video dial out: Using OS UC client for video call - to join video conferences from own company (routed through intranet)

NOTICE: UC to UC calls are not supported.

3 Foreign Peer Domains

3.1 Use cases and Restrictions

3.1.1 Dial in

Dial in supported formats

The supported dial in SIP URI formats are the following

1) number@domain, for example: 4989700712345@unify.com
2) alphanumeric@domain, for example: user@polycom.com
3) domain, for example: Unify.com or mcu10.unify.com

The IP address in the domain part of the SIP URI is not allowed, for example user@10.11.12.13 or IP 10.11.12.13

Dial in to subscribers

Video dial-in calls from the internet to subscribers are not supported and the system rejects this kind of requests.

Responsible Domain

Responsible domains are used to allow foreign domains to connect to a UC conference. You can add only domains and IP are not accepted. Also alphanumeric SIP URI with userinfo@domain format are not allowed.

The wildcard *domain.com* can be used to match a domain *mcu.domain.com* or alphanumeric SIPURI *mcu@domain.com* as part of *domain.com*.

When a call is received and the domain doesn't match with the Responsible Domain entries, then the call is rejected.

Restrictions:

- IPs are not allowed
- Alphanumeric SIP URI with userinfo@domain format is not allowed

Domain Code

The Domain code accepts an exact match entry in the form of *user@domain*

A wildcard can be used to cover the user part. For example, the wildcard *unify.com* can match *mcu@unify.com*

The wildcard on both user and domain parts is * and it allows all alphanumeric sip uri.

SBC Whitelist

The SBC White list allows the following cases:

- Only exact match: *sip:user@domain* or *sip:user@IPAddress*
- Wildcard match **@video.unify.com* or **.video.unify.com* for all video devices in the video.unify.com realm

- Regular expression, for example *sip/:sub/d/d/d/d/d@video/.unify/.com* to match a string *sip:sub12345@video.unify.com*
- The wildcard on both user and domain parts is not allowed for complete fields meaning *@*

3.1.2 Dial out

Dial out supported formats

The supported dial out SIP URI formats are the following

1) number@domain, for example: 4989700712345@unify.com
2) alphanumeric@domain, for example: user@polycom.com
3) Alice@10.11.12.13 (public IP)
4) 10.11.12.13 (public IP)
5) domain, for example: Unify.com or mcu10.unify.com

Dial out to subscribers

Video dial-out calls to internet subscribers is supported and depends on the external subscriber configuration.

Responsible Domain

A responsible domain is not needed for dial out. In case of used Dial in and Dial out in parallel, the only case needing attention is the numeric sip uri format, for example *1000@unify.com*. When the *unify.com* domain is added in the responsible domain for dial in purposes and the user tries to dial out *1000@unify.com*, then the called party is determined by translating the number in the userinfo (1000) and the call is routed internally to the OpenScape voice.

Domain Code

The Domain code accepts an exact match entry in the form of *user@domain*

A wildcard can be used to cover the user part. For example, the wildcard *unify.com* can match *mcu@unify.com*

The wildcard on both user and domain parts is * and it allows all alphanumeric SIP URI.

Remote Subscribers

Remote subscribers can dial alphanumeric SIP URI addresses. OSV users can register as remote subscribers in the SBC and can execute dial out calls in video conference systems. For security reasons, add extra interface in the SBC for remote subscribers.

3.2 Dial in configuration

3.2.1 OpenScape Voice configuration steps for Dial in

Follow the instructions below to configure the OpenScape Voice for dial in. Set the parameters specified with the exact values given and keep all other parameters with their default values or set whichever value you want. You can keep the names provided to the different parameters for easier navigation throughout the instructions or give the names your prefer.

New Class of service

First create a new class of service in Global Translation and Routing

1) Log in to the **CMP** and navigate to **Configuration** > **OpenScape Voice** > **Global Translation and Routing** > **Translation** > **Classes of Service**
2) Click **Add**
3) A pop-up window **Add Class of Service** appears. Configure the following parameters:
   - **Name**: SipUriRestrict

New Traffic Type

Create a new traffic type in Global Translation and Routing

1) Navigate to **Configuration** > **OpenScape Voice** > **Global Translation and Routing** > **Translation** > **Traffic Types**
2) Click **Add**
3) A pop-up window **Add Traffic Type** appears. Configure the following parameters:
   - **ID**: 20
   - **Name**: VoiceVideoCall

Endpoint profile for Originating Endpoint

Call originations from the foreign SIP server are restricted based on the CoS defined above. You must create a new Endpoint Profile to assign it to an endpoint. In the endpoint profile assign the class of service created earlier and set **SIP Privacy Support** to Full.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Profiles** > **Endpoint**
2) Click **Add**
3) A pop-up window **Add Endpoint Profile** appears. Configure the following parameters:
   - **Name**: ForeignDomainEPP
   - **Class of Service**: SipUriRestrict
   - **SIP Privacy Support**: Full

Endpoint Profile for Multipoint Control Unit (MCU)

The MCU Endpoint Profile is necessary for the SIP endpoint for routing purposes. No CoS needs to be assigned. Set **SIP Privacy Support** to Full.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Profiles** > **Endpoint**
2) Click **Add**
3) A pop-up window **Add Endpoint Profile** appears. Configure the following parameters:
   - **Name**: MCU
   - **SIP Privacy Support**: Full

Endpoint for Foreign Domains

Create the endpoint for Foreign Domains and assign the endpoint profile created previously. Use Central SBC in the endpoint template.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Members** > **Endpoints**
2) Click **Add**
3) A pop-up window **Add Endpoint** appears.

   Configure the following parameters in the **General** tab:

   - **Name**: foreignAccess
   - **Registered**: Checked
   - **Profile**: Click "**...**" and select ForeignDomainEPP
   - **Endpoint Template**: Click "**...**" and select Central SBC

   Configure the following parameters in the **SIP** tab:

   - **Endpoint Type**: SIP Trunking
   - **Endpoint Address**: The IP of the SBC
   - **Port**: The core realm port
   - **Transport Protocol**: TCP

   Enable the following attributes in the **Attributes** tab:

   - SIP Proxy
   - Central SBC
   - Route via Proxy
   - Send alphanumeric SIP URI when available
   - Support Foreign Peer Domain

   In the **Aliases** tab, click **Add** and give the IP and the port of the SBC:

   - **Name**: 10.10.170.119:50505 (example)

Endpoint for MCU

This is the Multipoint Control Unit that offers video conference. In this example, use the UC conference and the endpoint registered in the UC media server. Assign the MCU endpoint profile in the endpoint.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Endpoint Management** > **Endpoints**
2) Click **Add**

3) A pop-up window **Add Endpoint** appears.

Configure the following parameters in the **General** tab:

- **Name**: MCU
- **Registered**: Checked
- **Profile**: Click "**...**" and select MCU

Use IP or FQDN for the media server, port and protocol. Configure the following parameters in the **SIP** tab:

- **Endpoint Type**: SIP Trunking
- **Signaling Address Type**: IP Address or FQDN
- **Endpoint Address**: mcu.sipuri.com (FQDN example)
- **Port**: 5060 (example)
- **Transport Protocol**: TCP

Use the default attributes for the media server in the **Attributes** tab

In the **Aliases** tab, click **Add** and create the alias mcu.sipuri.com:

- **Name**: mcu.sipuri.com (example)

Responsible Domain

You can add trusted domains in the responsible domain to allow foreign domain access in UC conference. In this example, the domain *sbcu01.ucaasft.global-intra.net* has been used.

You can use a wildcard to match a domain, for example *global-intra.net* should be matched to the *sbcu01.ucaasft.global-intra.net* domain.

When a call received in OSV with the endpoint attribute **Support Foreign Peer Domains** set and the domain is not matched to Responsible domain, the call will be rejected.

Depending on your need, you can add the MCU domain in the responsible domain (for example, when the MCU belongs to a foreign domain)

1) Navigate to **Configuration** > **OpenScape Voice** > **Administration** > **Signaling Management** > **SIP**
2) A pop-up window **SIP Settings** appears.

Configure the following parameters in the **Responsible Domains** tab:

- **Enable**: Checked

Click **Add** and give the Responsible Domain’s name:

- **Name**: sbcu01.ucaasft.global-intra.net (example)

Destination for MCU Endpoint

Create a destination for the MCU endpoint

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Destinations and Routes** > **Destinations**
2) A pop-up window **Add Destination** appears.

Click **Add** and give the Destination’s name:

- **Name**: VoiceVideoAuthenticator (example)

Click **Save**

3) Click on the created destination and the **Edit Destination** window appears

   In the **Routes** tab, add the MCU endpoint in the destination

Destination for ForeignAccess Endpoint

Create a destination for the ForeignAccess endpoint

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Destinations and Routes** > **Destinations**
2) A pop-up window **Add Destination** appears.

   Click **Add** and give the Destination's name:

   - **Name**: foreignAccess_Dest (example)

   Click **Save**
3) Click on the created destination and the **Edit Destination** window appears

   In the **Routes** tab, add the foreignAccess endpoint in the destination

Domain Code for Calls

Create a Domain code to route incoming sip uri calls from foreign domain to the mcu video conference.

The Domain code accepts exact match entry: user@domain.

You can use a wildcard to cover user part. For example, the wildcard *sbcu01.ucaasft.global-intra.net* could match the *mcu@sbcu01.ucaasft.global-intra.net*. In this example, the sip uri *mcu@sbcu01.ucaasft.global-intra.net* is forwarded to the voicevideoAuthenticator destination and routed to the mcu. Use the Class of service SipUriRestricted with traffic type VoiceVideoCall.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Translation** > **Domain Codes**
2) Click **Add**
3) A pop-up window **Add Domain Code** appears.

   In the **General** tab configure the following parameters:

   - **Domain Code**: mcu@sbcu01.ucaasft.global-intra.net (example)
   - **Class Of Service**: SipUriRestrict
   - **Select Traffic Type**: VoiceVideoCall
   - **Destination Name**: VoiceVideoAuthenticator

Domain Code for internal subscribers

Create a domain code for internal subscribers.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Translation** > **Domain Codes**
2) Click **Add**
3) A pop-up window **Add Domain Code** appears.

   In the **General** tab configure the following parameters:

   - **Domain Code**: mcu@sbcu01.ucaasft.global-intra.net (example)
   - **Select Traffic Type**: VoiceVideoCall
   - **Destination Name**: VoiceVideoAuthenticator

3.2.2 Session Border Controler configuration steps for Dial in

Follow the instructions below to configure the Session Border Controller for dial in. Set the parameters specified with the exact values given and keep all other parameters with their default values or set whichever value you want. You can keep the names provided to the different parameters for easier navigation throughout the instructions or give the names your prefer.

Add domain FQDN in the SBC

You must a registered FQDN matched with the public IP of the SBC. The FQDN should be resolvable by an external DNS server. In this example *sbcu01.ucaasft.global-intra.net* used for dial in sip uri calls.

1) Log in to the **CMP** and navigate to **Configuration** > **OpenScape SBC** > **Administration**

Select an OpenScape SBC from the OpenScape SBC pull-down located in the Navigation Tree. Navigate to **Configuration > Network/Net Services > DNS** and configure the following parameters:

- **Alias**: Click **Add** and select sbcu01.ucaasft.global-intra.net

Create a remote endpoint for Dial in

Follow the steps below to create a remote endpoint for dial in

1) Navigate to **Configuration** > **OpenScape SBC** > **Features** > **Enable Remote Endpoints** > **Configure**

2) In the **Remote endpoint configuration** area click **Add**

3) A pop-up window **Remote endpoint configuration** appears. Configure the following parameters:

- **Type**: Gateway
- **Profile**: Gateway
- **Support Foreign Peer Domains**: Enable

  When a remote endpoint with this flag is found and there are entries in the whitelist, the entries must be checked against R-URI. Once match criteria are satisfied, the call is routed to the OSV. No entries in the white list means that OSV blocks everything.

  NOTICE:

  You can configure only one remote endpoint with the flag "Support Foreign Peer Domains" in the system.

- **Signaling address type**: DNS SRV or IP/FQDN
- Select **White list** and add the domain for the incoming call in the field **White list for trusted peers**.

  White list allows exact match entry in form sip:user@domain, wildcard match *@video.unify.com or *.video.unify.com for all video devices in video.unify.com

  The wildcard on both user and domain part is not allowed for complete fields i.e *@*
- **Core realm port**: Add the core realm port for OSV

4) Click **Save**

3.2.3 UC/ Media Server configuration steps for Dial in

Follow the instructions below to configure the UC/ Media Server for dial in. Set the parameters specified with the exact values given and keep all other parameters with their default values or set whichever value you want. You can keep the names provided to the different parameters for easier navigation throughout the instructions or give the names your prefer.

Enable SIP URI in OSV CSTA applications

First, enable the SIP URI in the CSTA Application

1) Log in to the **CMP** and navigate to **Configuration** > **OpenScape Voice** > **Administration** > **Signaling Management** > **CSTA**

2) A new pop-up window **CSTA Settings** appears

3) In the **Applications** tab, select a UC Backend server and click **Edit**. In the new pop-up window **Modify CSTA Application**, select the **Interface Options** tab and enable the following Interface attribute:

- Enable SIP URI

4) Click **Save**

WebUI enable SIPURI calling

- Use SSH to log in as root user to the UC Backend server
- Navigate to `/opt/siemens/HiPathCA/config/common`
- Edit the file `WebClientUI.cfg`
- In the bottom line search for the parameter **SearchOrDial.EnableSipUriDialing** and verify whether it is set to true. If not, change it to **SearchOrDial.EnableSipUriDialing** = true
- Save the file, exit and restart the webclient service to apply changes

SIP URI terminal for media server

Here you must select a Media server in the Unified Communications tab and create a new conference terminal for SIP URI.

In case you want to use more than one media servers for redundancy, repeat the steps below for the redundant media servers

1) Navigate to **Configuration** > **Unified Communications** > **Configuration** > **Media Server**

Select a Media Server from the **Media Server Nodes** list. A pop-up window **Node administration** appears.

2) In the **Terminals** tab click **Add**. A new pop-up window **Add a Terminal** appears. Configure the following parameters:

- **Terminal ID**: conference-sipuri (example)
- **Application**: Conferencing#welcome
- **Addresses**: Click **Add** and create a sip uri address in the form of: *sip: user@domain* or add sip uri without user part. In this example *sip:mcu@sbcu01.ucaasft.global-intra.net* is used but you can add additional entries. Select **Single number** and click **OK**
- **Streams**: Click **Add** and add also **Video** as Media Type.

3) Click **Save**

Conference sip uri bridge numbers

Follow the steps below to ass conference sip uri bridge numbers

1) Navigate to **Configuration** > **Unified Communications** > **Configuration** > **Conferences**

   A new pop-up window **Conferences** appears
2) In the **Configuration** tab, click **Bridge numbers**
3) A pop-up window **Number configuration** appears. In the **Bridge numbers** tab, click **Add** and configure the following parameters:
   - **Display name**: sip uri bridge1 (example)
   - **Number**: sip:mcu@sbcu01.ucaasft.global-intra.net (example)

   NOTICE:

   The sip uri bridge numbers must be in the form *sip:mcu@sbcu01.ucaasft.global-intra.net*.You can add one or more sip uri addresses as a bridge numbers.
4) Click **Save**

OSV connections configuration for video calls

1) Navigate to **Configuration** > **Unified Communications** > **Configuration** > **Connections** > **OS Voice**
2) Select an OS Voice.
3) In the **OpenScape Voice Connection** window:
   - **Version**: OpenScape Voice VX video
4) Click **Save**

3.3 Dial out configuration

3.3.1 OpenScape Voice configuration steps for Dial out

Follow the instructions below to configure the OpenScape Voice for dial out. Set the parameters specified with the exact values given and keep all other parameters with their default values or set whichever value you want. You can keep the names provided to the different parameters for easier navigation throughout the instructions or give the names your prefer.

Endpoint profile for Originating Endpoint

Create endpoint profile for the originating endpoint. A new Endpoint Profile is needed to assign it in the endpoint.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Profiles** > **Endpoint**
2) Click **Add**
3) A pop-up window **Add Endpoint Profile** appears. Configure the following parameters:
   - **Name**: DialOut_EPP
   - **SIP Privacy Support**: Full

Endpoint for Dial Out

Create the endpoint for foreign Domains and assign the endpoint profile created (DialOut_EPP). Use Central SBC in the endpoint template.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Members** > **Endpoints**
2) Click **Add**
3) A pop-up window **Add Endpoint** appears.

   Configure the following parameters in the **General** tab:

   - **Name**: DialOut_EP
   - **Registered**: Checked
   - **Profile**: Click "**...**" and select DialOut_EPP
   - **Endpoint Template**: Click "**...**" and select Central SBC
   - **Endpoint Type**: Central SBC

   Configure the following parameters in the **SIP** tab:

   - **Endpoint Type**: SIP Trunking
   - **Endpoint Address**: The IP of the SBC
   - **Port**: The core realm port
   - **Transport Protocol**: TCP

   Enable the following attributes in the **Attributes** tab:

   - SIP Proxy
   - Central SBC
   - Route via Proxy
   - Send alphanumeric SIP URI when available
   - Support Foreign Peer Domain

   In the **Aliases** tab, click **Add** and give the IP and the port of the SBC:

   - **Name**: 10.10.170.119:50505 (example)

Responsible Domain

Generally, Responsible Domain is not needed for dial out. In case of used Dial in and Dial out in parallel, the only case you must take care is the numeric sip uri format, for example 1000@unify.com. When the unify.com domain is added in the responsible domain for dial in purposes and the user tries to dial out 1000@unify.com, the called party is determined by translating the number in the userinfo (1000) and the call is routed internally to the OpenScape Voice.

Destination for Dial out Endpoint

Create a destination for the Dial out endpoint

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Destinations and Routes** > **Destinations**
2) A pop-up window **Add Destination** appears.

   Click **Add** and give the Destination's name:

   - **Name**: DialOut_Dest (example)

   Click **Save**
3) Click on the created destination and the **Edit Destination** window appears

   In the **Routes** tab, add the DialOut_EP endpoint in the destination

Subscriber Configuration

Create subscribers and fill in the desired Alphanumeric SIP URI for each one of them.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Members** > **Subscribers**
2) Click **Add**
3) A pop-up window **Add Subscriber** appears.

   Configure the following parameter in the **General** tab:

   - **Alphanumeric SIP URI**: User@unify.com (example)

   Enable the following attributes in the **Routing** tab:

   - Video Call Allowed
   - Send alphanumeric SIP URI when available

Domain Code for Dial out

Create a Domain code to route outgoing sip uri calls to internet MCU video conference. In this example, sip uri *mcu@meetingsemea8.webex.com* will be forwarded to the DialOut_Dest destination and routed to the webex video conference from DialOut_EP endpoint.

The Domain code accepts only exact match entries in the form user@domain.

Wildcards can be used to cover the user part, for example wildcard *meetingsemea8.webex.com* could match *mcu@meetingsemea8.webex.com*. The wildcard on both user and domain part is * and it allows all alphanumeric sip uri.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Translation** > **Domain Codes**
2) Click **Add**
3) A pop-up window **Add Domain Code** appears.

   In the **General** tab configure the following parameters:

   - **Domain Code**: mcu@meetingsemea8.webex.com (example)
   - **Traffic Type**: None
   - **Destination Name**: DialOut_Dest

3.3.2 UC configuration steps for Dial out

Follow the instructions below to configure the UC for dial out. Set the parameters specified with the exact values given and keep all other parameters with their default values or set whichever value you want. You can keep the names provided to the different parameters for easier navigation throughout the instructions or give the names your prefer.

Enable SIP URI in OSV CSTA applications

First, enable the SIP URI in the CSTA Application

1) Log in to the **CMP** and navigate to **Configuration** > **OpenScape Voice** > **Administration** > **Signaling Management** > **CSTA**
2) A new pop-up window **CSTA Settings** appears

3) In the **Applications** tab, select a UC Backend server and click **Edit**. In the new pop-up window **Modify CSTA Application**, select the **Interface Options** tab and enable the following Interface attribute:
   - Enable SIP URI
4) Click **Save**

WebUI enable SIPURI calling

- Use SSH to log in as root user to the UC Backend server
- Navigate to `/opt/siemens/HiPathCA/config/common`
- Edit the file `WebClientUI.cfg`
- In the bottom line search for the parameter **SearchOrDial.EnableSipUriDialing** and verify whether it is set to true. If not, change it to **SearchOrDial.EnableSipUriDialing** = true
- Save the file, exit and restart the webclient service to apply changes

SIP URI terminal for media server

Here you must select a Media server in the Unified Communications tab and create a new conference terminal for SIP URI.

In case you want to use more than one media servers for redundancy, repeat the steps below for the redundant media servers

1) Navigate to **Configuration** > **Unified Communications** > **Configuration** > **Media Server**

   Select a Media Server from the **Media Server Nodes** list. A pop-up window **Node administration** appears.
2) In the **Terminals** tab click **Add**. A new pop-up window **Add a Terminal** appears. Configure the following parameters:
   - **Terminal ID**: conference-sipuri (example)
   - **Application**: Conferencing#welcome
   - **Addresses**: Click **Add** and create a sip uri address in the form of: *sip: user@domain* or add sip uri without user part. In this example *sip:mcu@sbcu01.ucaasft.global-intra.net* is used but additional entries can be added. Select **Single number** and click **OK**
   - Streams: Click **Add** and add also **Video** as Media Type.
3) Click **Save**

Conference sip uri bridge numbers

Follow the steps below to ass conference sip uri bridge numbers

1) Navigate to **Configuration** > **Unified Communications** > **Configuration** > **Conferences**

   A new pop-up window **Conferences** appears
2) In the **Configuration** tab, click **Bridge numbers**

3) A pop-up window **Number configuration** appears. In the **Bridge numbers** tab, click **Add** and configure the following parameters:

- **Display name**: sip uri bridge1 (example)
- **Number**: sip:mcu@sbcu01.ucaasft.global-intra.net (example)

> **NOTICE:**
>
> The sip uri bridge numbers must be in the form *sip:mcu@sbcu01.ucaasft.global-intra.net*.You can add one or more sip uri addresses as a bridge numbers.

- Select **White list** and add the domain for the incoming call in the field **White list for trusted peers**.

  White list allows exact match entry in form sip:user@domain, wildcard match *@video.unify.com or *.video.unify.com for all video devices in video.unify.com

  The wildcard on both user and domain part is not allowed for complete fields i.e *@*
- **Core realm port**: Add the core realm port for OSV

4) Click **Save**

OSV connections configuration for video calls

1) Navigate to **Configuration** > **Unified Communications** > **Configuration** > **Connections** > **OS Voice**
2) Select an OS Voice.
3) In the **OpenScape Voice Connection** window:

- **Version**: OpenScape Voice VX video

4) Click **Save**

3.3.3 Session Border Controler configuration steps for Dial out

Follow the instructions below to configure the Session Border Controller for dial iout. Set the parameters specified with the exact values given and keep all other parameters with their default values or set whichever value you want. You can keep the names provided to the different parameters for easier navigation throughout the instructions or give the names your prefer.

Add DNS in the SBC

You must select DNS from the Network/Net service menu of the SBC.

1) Log in to the **CMP** and navigate to **Configuration** > **OpenScape SBC** > **Administration**

   Select an OpenScape SBC from the OpenScape SBC pull-down located in the Navigation Tree. Navigate to **Configuration > Network/Net Services > DNS** and configure the following parameters:

   - **DNS server IP address**: 10.6.0.254 (example)

Create a remote endpoint for Dial out

Follow the steps below to create a remote endpoint for dial out

1) Navigate to **Configuration** > **OpenScape SBC** > **Features** > **Enable Remote Endpoints** > **Configure**
2) In the **Remote endpoint configuration** area click **Add**
3) A pop-up window **Remote endpoint configuration** appears. Configure the following parameters:
   - **Type**: Gateway
   - **Profile**: Gateway
   - **Access realm profile**: Main-Access-Realm-ipv4
   - **Core realm profile**: Main-Access-Realm-ipv4
   - **Support Foreign Peer Domains**: Enable

     When a remote endpoint with this flag is found and there are entries in the whitelist, the entries must be checked against R-URI. Once match criteria are satisfied, the call is routed to the OSV. No entries in the white list means that OSV blocks everything.

     NOTICE:

     You can configure only one remote endpoint with the flag "Support Foreign Peer Domains" in the system.
   - **Signaling address type**: DNS SRV or IP/FQDN

     NOTICE:

     SBC checks for SRV or FQDN entries in the DNS. When DNS query is successful, then the call is routed to the external video conference from the access SBC interface.
   - **Core realm port**: Add the core realm port for OSV
4) Click **Save**

3.3.4 Supported Clients for Dial out

The supported UC clients for dial out are OpeScape Fusion for office, Webclient in WebRTC mode and the OpenScape Mobile client.

4 Peer Domains

4.1 Use cases and Restrictions

4.1.1 Dial in

Dial in supported formats

The supported dial in SIP URI formats are the following

1) alphanumeric@domain, for example: user@polycom.com
2) domain, for example: Unify.com or mcu10.unify.com

For the case when userinfo contains number (number@domain, for example 4989700712345@unify.com) and the domain is added in the responsible domain, the called party is determined by translating the number in the userinfo and the call is routed from E164.

Dial in to subscribers

Video dial-in calls from trusted domains subscribers is supported.

Responsible Domain

Responsible domain is used to verify whether the call is trusted or not.
Add only domains, IPs are not accepted. Also alphanumeric SIP URI with userinfo@domain format are not allowed.

The wildcard *domain.com* can be used to match a domain *mcu.domain.com* or alphanumeric SIPURI *mcu@domain.com* as part of *domain.com*.

Restrictions:

- IPs are not allowed
- Alphanumeric SIP URI with userinfo@domain format is not allowed

Domain Code

The Domain code accepts an exact match entry in the form of *user@domain*

A wildcard can be used to cover the user part. For example, the wildcard *unify.com* can match *mcu@unify.com*

The wildcard on both user and domain parts is * and it allows all alphanumeric sip URI

4.1.2 Dial out

Dial out supported formats

The supported dial out SIP URI formats are the following

1) alphanumeric@domain, for example: user@polycom.com
2) domain, for example: Unify.com

For the case when the userinfo contains number (number@domain, for example 4989700712345@unify.com) and the domain is added in the responsible domain, the called party is determined by translating the number in the userinfo after stripping out unnecessary characters and the call is routed from E164.

Dial out to subscribers

Video dial-out calls trusted domains subscribers is supported.

Responsible Domain

In case of a Dial out numeric SIP URI format, for example 1000@unify.com, when the *unify.com* domain is added in the responsible domain, the called party is determined by translating the number in the userinfo (1000) and the call is routed internally to the OpenScape Voice

Domain Code

The Domain code accepts an exact match entry in the form of *user@domain*

A wildcard can be used to cover the user part. For example, the wildcard *unify.com* can match *mcu@unify.com*

The wildcard on both user and domain parts is * and it allows all alphanumeric SIP URI.

4.2 Dial in configuration

4.2.1 OpenScape Voice configuration steps for Dial in

Follow the instructions below to configure the OpenScape Voice for dial in. Set the parameters specified with the exact values given and keep all other parameters with their default values or set whichever value you want. You can keep the names provided to the different parameters for easier navigation throughout the instructions or give the names your prefer.

Endpoint profile for Originating Endpoint

You must create a new Endpoint Profile to assign it to an endpoint for peer domains. In the endpoint profile assign the class of service created earlier and set **SIP Privacy Support** to Full.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Profiles** > **Endpoint**
2) Click **Add**
3) A pop-up window **Add Endpoint Profile** appears. Configure the following parameters:
   - **Name**: Peer_Domain_SBC
   - **SIP Privacy Support**: Full

Endpoint for Peer Domains

Create the endpoint for Peer Domains and assign the endpoint profile created previously. Use Central SBC in the endpoint template.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Members** > **Endpoints**
2) Click **Add**
3) A pop-up window **Add Endpoint** appears.

   Configure the following parameters in the **General** tab:

   - **Name**: Peer_SBC
   - **Registered**: Checked
   - **Profile**: Click "**...**" and select Peer_Domain_SBC
   - **Endpoint Template**: Click "**...**" and select Central SBC

   Configure the following parameters in the **SIP** tab:

   - **Endpoint Type**: SIP Trunking
   - **Endpoint Address**: The IP of the SBC
   - **Port**: The core realm port
   - **Transport Protocol**: TCP

   Enable the following attributes in the **Attributes** tab:

   - SIP Proxy
   - Central SBC
   - Route via Proxy
   - Send alphanumeric SIP URI when available
   - Support Peer Domain

   In the **Aliases** tab, click **Add** and give the IP and the port of the SBC:

   - **Name**: 10.10.185.66:50100 (example)

Endpoint for UC conference

This is the endpoint used for UC video conference. In this example the endpoint is registered in UC media server. Assign MCU endpoint profile in the endpoint.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Endpoint Management** > **Endpoints**
2) Click **Add**
3) A pop-up window **Add Endpoint** appears.

   Configure the following parameters in the **General** tab:

   - **Name**: MCU
   - **Registered**: Checked
   - **Profile**: Click "**...**" and select MCU

   Use IP or FQDN for the media server, port and protocol. In this example an FQDN is used, for example mcu.sipuri.com. Configure the following parameters in the **SIP** tab:

   - **Endpoint Type**: SIP Trunking
   - **Signaling Address Type**: IP Address or FQDN
   - **Endpoint Address**: mcu.sipuri.com
   - **Port**: 5060 (example)
   - **Transport Protocol**: TCP

   Use the default attributes for the media server in the **Attributes** tab

   In the **Aliases** tab, click **Add** and create the alias mcu.sipuri.com:

   - **Name**: mcu.sipuri.com (example)

Responsible Domain

You can add trusted domains in the responsible domain to allow peer domains access in UC conference or subscriber. In this example, the domain *unify.com* has been used.

You can use a wildcard to match a domain, for example *unify.com* should be matched to the *mcu.unify.com* domain.

1) Navigate to **Configuration** > **OpenScape Voice** > **Administration** > **Signaling Management** > **SIP**
2) A pop-up window **SIP Settings** appears.

   Configure the following parameters in the **Responsible Domains** tab:

   - **Enable**: Checked

   Click **Add** and give the Responsible Domain's name:

   - **Name**: unify.com (example)

Destination - VoiceVideoAuthenticator

Create a destination for the MCU endpoint

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Destinations and Routes** > **Destinations**
2) A pop-up window **Add Destination** appears.

   Click **Add** and give the Destination's name:

   - **Name**: VoiceVideoAuthenticator (example)

   Click **Save**
3) Click on the created destination and the **Edit Destination** window appears

   In the **Routes** tab, add the MCU endpoint in the destination

Destination for PeerAccess Endpoint

Create a destination for the PeerAccess endpoint

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Destinations and Routes** > **Destinations**
2) A pop-up window **Add Destination** appears.

   Click **Add** and give the Destination's name:

   - **Name**: PeerAccess_Dest (example)

   Click **Save**
3) Click on the PeerAccess_Dest destination and the **Edit Destination** window appears

   In the **Routes** tab, add the Peer_SBC endpoint in the destination

Domain Code for Incoming Calls

Create a Domain code to route incoming SIP URI calls from peer domain to the MCU video conference.

The Domain code accepts exact match entry: user@domain.

You can use a wildcard to cover the user part. For example, the wildcard *unify.com* could match the *conf@unify.com*. In this example, the SIP URI *conf@unify.com* is forwarded to the voicevideoAuthenticator destination and routed to the UC conference.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Translation** > **Domain Codes**
2) Click **Add**
3) A pop-up window **Add Domain Code** appears.

   In the **General** tab configure the following parameters:

   - **Domain Code**: conf@unify.com
   - **Select Traffic Type**: VoiceVideoCall
   - **Destination Name**: VoiceVideoAuthenticator

4.2.2 Session Border Controler configuration steps for Dial in

Follow the instructions below to configure the Session Border Controller for dial in. Set the parameters specified with the exact values given and keep all other parameters with their default values or set whichever value you want. You can keep the names provided to the different parameters for easier navigation throughout the instructions or give the names your prefer.

Add DNS in the SBC

Select DNS from the Network/Net service menu of the sbc. Add DNS server in the sbc.

1) Log in to the **CMP** and navigate to **Configuration** > **OpenScape SBC** > **Administration**

   Select an OpenScape SBC from the OpenScape SBC pull-down located in the Navigation Tree. Navigate to **Configuration > Network/Net Services > DNS** and configure the following parameters:

   - **DNS server IP address**: Click **Add** and select the IP address of the DNS

Create a remote endpoint for Dial in

Follow the steps below to create a remote endpoint for dial in

1) Navigate to **Configuration** > **OpenScape SBC** > **Features** > **Enable Remote Endpoints** > **Enable** > **Configure**
2) In the **Remote endpoint configuration** area click **Add**
3) A pop-up window **Remote endpoint configuration** appears. Configure the following parameters:

   - **Type**: Gateway
   - **Profile**: Gateway
   - **Support Peer Domains**: Enable
   - **Enable access control**: Enable

NOTICE:

When the flag **Enable access control** is used, the incoming calls are allowed only when the protocol, port, and IP/FQDN configured in one of the Remote Location

Domain list are matched with the Via header on the INVITE message.

NOTICE:

When the **Enable access control** flag is disabled and the remote endpoint is configured with the flag **Support Peer Domains**, the incoming calls are accepted from unknown peers when the MTLS negotiations are successful. The Remote Location Domain list contains exactly one entry with a blank remote URL in this case.

- **Signaling address type**: DNS SRV
- **Core realm port**: Add the core realm port for OSV

4) Click **Save**

4.2.3 UC/ Media Server configuration steps for Dial in

Follow the instructions below to configure the UC/ Media Server for dial in. Set the parameters specified with the exact values given and keep all other parameters with their default values or set whichever value you want. You can keep the names provided to the different parameters for easier navigation throughout the instructions or give the names your prefer.

Enable SIP URI in OSV CSTA applications

First, enable the SIP URI in the CSTA Application

1) Log in to the **CMP** and navigate to **Configuration** > **OpenScape Voice** > **Administration** > **Signaling Management** > **CSTA**

2) A new pop-up window **CSTA Settings** appears

3) In the **Applications** tab, select a UC Backend server and click **Edit**. In the new pop-up window **Modify CSTA Application**, select the **Interface Options** tab and enable the following Interface attribute:

- Enable SIP URI

4) Click **Save**

WebUI enable SIPURI calling

- Use SSH to log in as root user to the UC Backend server
- Navigate to `/opt/siemens/HiPathCA/config/common`
- Edit the file `WebClientUI.cfg`
- In the bottom line search for the parameter **SearchOrDial.EnableSipUriDialing** and verify whether it is set to true. If not, change it to **SearchOrDial.EnableSipUriDialing** = true
- Save the file, exit and restart the webclient service to apply changes

SIP URI terminal for media server

Here you must select a Media server in the Unified Communications tab and create a new conference terminal for SIP URI.

In case you want to use more than one media servers for redundancy, repeat the steps below for the redundant media servers

1) Navigate to **Configuration** > **Unified Communications** > **Configuration** > **Media Server**

   Select a Media Server from the **Media Server Nodes** list. A pop-up window **Node administration** appears.
2) In the **Terminals** tab click **Add**. A new pop-up window **Add a Terminal** appears. Configure the following parameters:
   - **Terminal ID**: conference-sipuri (example)
   - **Application**: Conferencing#welcome
   - **Addresses**: Click **Add** and create a sip uri address in the form of: *sip: user@domain* or add sip uri without user part. In this example *sip:conf@unify.com* is used but you can add additional entries. Select **Single number** and click **OK**
   - **Streams**: Click **Add** and add also **Video** as Media Type.
3) Click **Save**
4) In the **Address Binding** tab, check that the SIP URI entries have been added.

Conference sip uri bridge numbers

Follow the steps below to ass conference SIP URI bridge numbers

1) Navigate to **Configuration** > **Unified Communications** > **Configuration** > **Conferences**

   A new pop-up window **Conferences** appears
2) In the **Configuration** tab, click **Bridge numbers**
3) A pop-up window **Number configuration** appears. In the **Bridge numbers** tab, click **Add** and configure the following parameters:
   - **Display name**: sip uri (example)
   - **Number**: sip:conf@unify.com (example)

   NOTICE:

   The sip uri bridge numbers must be in the form *sip:conf@unify.com*.You can add one or more sip uri addresses as a bridge numbers.
4) Click **Save**

OSV connections configuration for video calls

1) Navigate to **Configuration** > **Unified Communications** > **Configuration** > **Connections** > **OS Voice**
2) Select an OS Voice.
3) In the **OpenScape Voice Connection** window:
   - **Version**: OpenScape Voice VX video
4) Click **Save**

4.3 Dial out configuration

4.3.1 OpenScape Voice configuration steps for Dial out

Follow the instructions below to configure the OpenScape Voice for dial out. Set the parameters specified with the exact values given and keep all other parameters with their default values or set whichever value you want. You can keep the names provided to the different parameters for easier navigation throughout the instructions or give the names your prefer.

Endpoint profile for Originating Endpoint

Create endpoint profile for the originating endpoint. A new Endpoint Profile is needed to assign it in the endpoint.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Profiles** > **Endpoint**
2) Click **Add**
3) A pop-up window **Add Endpoint Profile** appears. Configure the following parameters:
   - **Name**: Peer_Domain_SBC
   - **SIP Privacy Support**: Full

Endpoint for Dial Out

Create the endpoint for foreign Domains and assign the endpoint profile created (Peer_Domain_SBC). Use Central SBC in the endpoint template.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Members** > **Endpoints**
2) Click **Add**

3) A pop-up window **Add Endpoint** appears.

   Configure the following parameters in the **General** tab:

   - **Name**: Peer_SBC
   - **Registered**: Checked
   - **Profile**: Click "**...**" and select Peer_Domain_SBC
   - **Endpoint Template**: Click "**...**" and select Central SBC
   - **Endpoint Type**: Central SBC

   Configure the following parameters in the **SIP** tab:

   - **Endpoint Type**: SIP Trunking
   - **Endpoint Address**: The IP of the SBC
   - **Port**: The core realm port
   - **Transport Protocol**: TCP

   Enable the following attributes in the **Attributes** tab:

   - SIP Proxy
   - Central SBC
   - Route via Proxy
   - Send alphanumeric SIP URI when available
   - Support Peer Domains

   In the **Aliases** tab, click **Add** and give the IP and the port of the SBC:

   - **Name**: 10.10.170.119:50505 (example)

Responsible Domain

Generally, Responsible Domain is not needed for dial out. In case of used Dial in and Dial out in parallel, the only case you must take care is the numeric SIP URI format, for example 1000@unify.com. When the unify.com domain is added in the responsible domain for dial in purposes and the user tries to dial out 1000@unify.com, the called party is determined by translating the number in the userinfo (1000) and the call is routed internally to the OpenScape Voice.

Destination for Dial out Endpoint

Create a destination for the Dial out endpoint

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Destinations and Routes** > **Destinations**
2) A pop-up window **Add Destination** appears.

   Click **Add** and give the Destination’s name:

   - **Name**: DialOut_Dest (example)

   Click **Save**
3) Click on the created destination and the **Edit Destination** window appears

   In the **Routes** tab, add the DialOut_EP endpoint in the destination

Subscriber Configuration

Create subscribers and fill in the desired Alphanumeric SIP URI for each one of them.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Members** > **Subscribers**

2) Click **Add**
3) A pop-up window **Add Subscriber** appears.

   Configure the following parameter in the **General** tab:

   - **Alphanumeric SIP URI**: User@unify.com (example)

   Enable the following attributes in the **Routing** tab:

   - Video Call Allowed
   - Send alphanumeric SIP URI when available

Domain Code for Dial out

Create a Domain code to route outgoing SIP URI calls to internet MCU video conference. In this example, sip uri *user1@yourdomain.com* will be forwarded to the DialOut_Dest destination and routed to the video conference system or subscriber from the DialOut_EP endpoint.

The Domain code accepts only exact match entries in the form user@domain.

Wildcards can be used to cover the user part, for example wildcard *yourdomain.com* could match *user1@yourdomain.com*. The wildcard on both user and domain part is * and it allows all alphanumeric SIP URI.

1) Navigate to **Configuration** > **OpenScape Voice** > **Business Group** > **Translation** > **Domain Codes**
2) Click **Add**
3) A pop-up window **Add Domain Code** appears.

   In the **General** tab configure the following parameters:

   - **Domain Code**: user1@yourdomain.com (example)
   - **Traffic Type**: None
   - **Destination Type**: Destination
   - **Destination Name**: DialOut_Dest

4.3.2 UC configuration steps for Dial out

Follow the instructions below to configure the UC for dial out. Set the parameters specified with the exact values given and keep all other parameters with their default values or set whichever value you want. You can keep the names provided to the different parameters for easier navigation throughout the instructions or give the names your prefer.

Enable SIP URI in OSV CSTA applications

First, enable the SIP URI in the CSTA Application

1) Log in to the **CMP** and navigate to **Configuration** > **OpenScape Voice** > **Administration** > **Signaling Management** > **CSTA**
2) A new pop-up window **CSTA Settings** appears
3) In the **Applications** tab, select a UC Backend server and click **Edit**. In the new pop-up window **Modify CSTA Application**, select the **Interface Options** tab and enable the following Interface attribute:

   - Enable SIP URI
4) Click **Save**

WebUI enable SIP URI calling

- Use SSH to log in as root user to the UC Backend server
- Navigate to `/opt/siemens/HiPathCA/config/common`
- Edit the file `WebClientUI.cfg`
- In the bottom line search for the parameter **SearchOrDial.EnableSipUriDialing** and verify whether it is set to true. If not, change it to **SearchOrDial.EnableSipUriDialing** = true
- Save the file, exit and restart the webclient service to apply changes

4.3.3 Session Border Controler configuration steps for Dial out

Follow the instructions below to configure the Session Border Controller for dial iout. Set the parameters specified with the exact values given and keep all other parameters with their default values or set whichever value you want. You can keep the names provided to the different parameters for easier navigation throughout the instructions or give the names your prefer.

Add DNS in the SBC

You must select DNS from the Network/Net service menu of the SBC.

1) Log in to the **CMP** and navigate to **Configuration** > **OpenScape SBC** > **Administration**

   Select an OpenScape SBC from the OpenScape SBC pull-down located in the Navigation Tree. Navigate to **Configuration > Network/Net Services > DNS** and configure the following parameters:

   - **DNS server IP address**: 10.6.0.254 (example)

Create a remote endpoint for Dial out

Follow the steps below to create a remote endpoint for dial out

1) Navigate to **Configuration** > **OpenScape SBC** > **Features** > **Enable Remote Endpoints** > **Enable** > **Configure**
2) In the **Remote endpoint configuration** area click **Add**
3) A pop-up window **Remote endpoint configuration** appears. Configure the following parameters:

   - **Type**: Gateway
   - **Profile**: Gateway
   - **Access realm profile**: Main-Access-Realm-ipv4
   - **Core realm profile**: Main-Access-Realm-ipv4
   - **Support Peer Domains**: Enable
   - **Enable access control**: Enable

   NOTICE:

   When the flag **Enable access control** is used, the incoming calls are allowed only when the protocol, port, and IP/FQDN configured in one of the Remote Location

Domain list are matched with the Via header on the INVITE message.

NOTICE:

When the **Enable access control** flag is disabled and the remote endpoint is configured with the flag **Support Peer Domains**, the incoming calls are accepted from unknown peers when the MTLS negotiations are successful. The Remote Location Domain list contains exactly one entry with a blank remote URL in this case.

- **Signaling address type**: DNS SRV
- **Core realm port**: Add the core realm port for OSV

4) Click **Save**

4.3.4 Supported Clients for Dial out

The supported UC clients for dial out are OpeScape Fusion for office, Webclient in WebRTC mode and the OpenScape Mobile client.