A MITEL PRODUCT GUIDE

Unify OpenScape Solution Set V10

OpenScape Voice with MS Teams and OpenScape SBC, Configuration Guide

Administrator Documentation

10/2025

A31003-S1100-M122-09-7631

Mitel

Contents

History of Changes

Issue	Date	Summary
1	03/2017	First issue of the guide
2	03/2021	Minor Changes
3	01/2022	v1.8 Updates
4	02/2022	v1.9 Updates
5	11/2022	Minor Change
6	06/2023	Minor Change
7	08/2023	Minor Change
8	03/2024	Added Chapter 1.3 Microsoft Teams & AudioCodecs SBC
9	09/2025	Improvements and enhancements throughout the document. Added RTP parameter configurations for **Non-Media Bypass Mode** and **Media Bypass Mode.**

1 Introduction

Microsoft Teams Direct Routing solution with Unify OpenScape SBC (and OpenScape Voice) is available and released for both Media Bypass ON and OFF Teams tenant configurations.

This document describes how to connect the Unify OpenScape SBC (with OpenScape Voice) to Microsoft Teams Direct routing configuration. Microsoft Teams Direct Routing configuration isn't included in current document and may be found under https://docs.microsoft.com/en-us/microsoftteams/direct-routing-configure.

Product	SW Version	Media Bypass OFF Support	Media Bypass ON Support
OpenScape Voice	**V10R2.13.1**	✓	✓
OpenScape SBC	**V10R1.2.0**	✓	
OpenScape SBC	**V10R2.2.0**	✓	✓
OpenScape Apps	**V10 R3.10.1**	✓	✓

1.1 Additional Support Information

In the current Unify product software implementation:

- OpenScape SBC with OpenScape Voice solution is supported.
- OpenScape SBC with OpenScape 4000 solution is supported.
- SBC standalone (without PBX) is not currently supported.
- Domain-based MS Teams multi-tenancy is supported.
- Comfort Noise generation is not currently supported by OpenScape SBC.
- The History-Info header is not currently supported by OpenScape Voice & OpenScape 4000.
- The OSEE environment with SBC-THIG and Teams is not currently supported.

1.2 Network Topology

The block diagrams in **fig.1** and **fig.2** below show the Teams Direct Routing connection topology along with the SIP / Media flows for Media Bypass OFF and Media Bypass ON cases.

While Teams Phone System is directly connected to OpenScape SBC, the PSTN access is possible to be available through OpenScape Voice, too.

The example configuration presented in current document covers the PSTN (SSP) SIP trunk connectivity to OS SBC directly; PSTN access through OS Voice is out of scope.

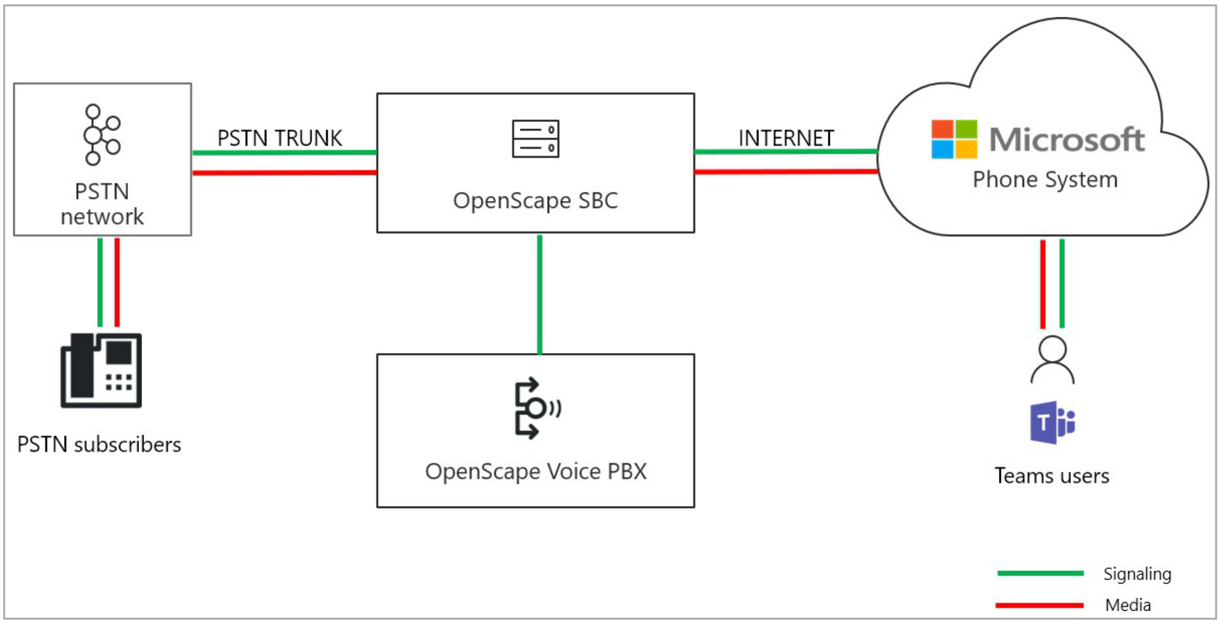


Figure 1: Topology Block Diagram SIP and Media Flows – Media Bypass OFF

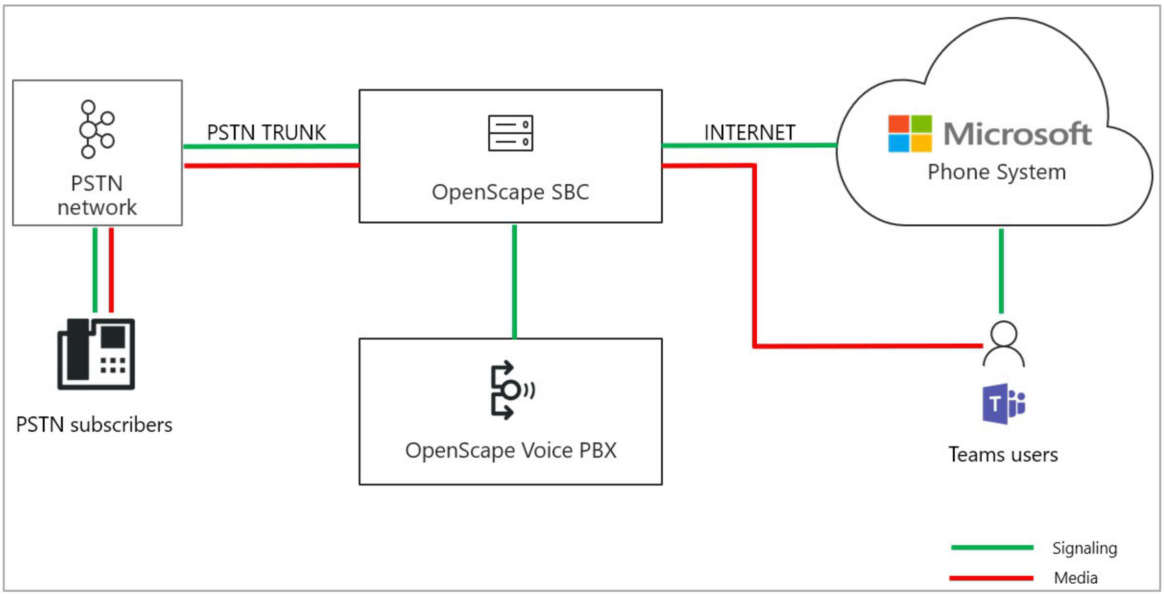


Figure 2: Topology Block Diagram SIP and Media Flows – Media Bypass ON

When media bypass is deactivated and a Teams client makes or receives a call, both signaling and media flow between the SBC, the Microsoft Phone System, and the Teams client, as shown in **fig.1**.

On the other hand, when media bypass is activated and a Teams client makes or receives a call, the signaling continues to flow between the SBC, the Microsoft Phone System, and the Teams client, as in the case of Media Bypass OFF. However, in the Media Bypass ON case, the media flow is kept directly between the Teams client and the SBC, as shown in **fig.2**.

The OpenScape SBC isn't a B2BUA, and it has limited SIP message manipulation capabilities. The OpenScape Voice IP-PBX provides call routes, enhanced SIP message manipulation, and number modification facilities. Thus, SIP signaling for incoming and outgoing calls to the Teams client will always pass through the OS Voice service.

1.3 Microsoft Teams & AudioCodes SBC

For the current certification, Microsoft Teams Direct Routing configuration is utilized to set up the testing environment.

The prerequisites for Direct Routing are:

1. MS Team's users of Direct Routing must have the following licenses assigned in Office 365:
    - *Office 365 Enterprise E3 (including SfB Online Plan2, Exchange Plan2, and Teams)*
    - *Phone System licenses or Office 365 Enterprise E5 (including SfB Online Plan2, Exchange Plan2, Teams, Phone System, and Audio Conferencing)*.
2. MS Teams certified SBC (https://docs.microsoft.com/en-us/microsoftteams/direct-routing- border-controllers).
3. A publicly registered domain name. A public domain name, such as *onmicrosoft.com,* is not a possibility for direct routing.
4. Public trusted certificate for the SBC with a SAN record with the host name of the SBC. The certificate must be from one Microsoft's approved root CAs (https://docs.microsoft.com/en-us/MicrosoftTeams/direct-routing-plan#public-trusted-certificate-for-the-sbc).
5. Public IP address for SBC WAN connection and appropriate firewall rules for signaling.
6. "SBC MS Direct Routing" License should be obtained and applied to OpenScape SBC.

2 OpenScape Voice Configuration

This chapter describes the OpenScape Voice configuration for connecting to OpenScape SBC. The purpose of this connectivity is to enable OpenScape Voice to provide the necessary SIP message manipulation and call routing functions to OpenScape SBC, allowing it to interconnect with Teams Direct Routing and facilitate calls between Teams clients and PSTN subscribers.

In OpenScape Voice, the connection to the OpenScape SBC must be set up, along with the signaling paths to Microsoft Teams datacenters and the SSP (PSTN provider).

In addition, call routing must be configured according to the numbering plan of Teams users and PSTN subscribers.

As an example:

Items	Example
SBC IP	10.8.242.72 TCP 5060
Signaling path to Teams FQDN 1: sip.pstnhub.microsoft.com	10.8.242.72 TCP 50001
Signaling path to Teams FQDN 2: sip2.pstnhub.microsoft.com	10.8.242.72 TCP 50002
Signaling path to Teams FQDN 3: sip3.pstnhub.microsoft.com	10.8.242.72 TCP 50003
Signaling path to Teams FQDN ALL: sip-all.pstnhub.microsoft.com **see note**	10.8.242.72 TCP 50004
Signaling path to PSTN provider: BCOM	10.8.242.72 TCP 50010
Teams user number ranges (reachable from PSTN)	31850080xxx
PSTN subscriber number ranges	498970072xxxx

For OS Voice installation or General configuration, refer to the Unify customer documentation site.

Important:

Per Microsoft's announcement, support for the *"sip-all.pstnhub.microsoft.com"* FQDN will end in March 2022.

Although Microsoft recommends using the three FQDNs for Direct Routing connection points — **"sip.pstnhub.microsoft.com"**, **"sip2.pstnhub.microsoft.com"**, and **"sip3.pstnhub.microsoft.com"** — the **"sip-all.pstnhub.microsoft.com"** FQDN was originally used in Unify component configurations due to DNS resolution issues in some

countries.
However, there have been reported cases where the **"sip-all.pstnhub.microsoft.com"** FQDN can cause incorrect certificate negotiation between OpenScape SBC and the Microsoft Teams tenant.

Therefore, do NOT configure the SIP trunk to point to "sip-all.pstnhub.microsoft.com" in Unify components unless explicitly recommended by Unify support.

For completeness, this document still presents the configuration of the **"sip-all.pstnhub.microsoft.com"** SIP trunk.

Configuring the Endpoints

An **Endpoint** is a network component, such as an originating or terminating device, and in our case, OpenScape SBC. An endpoint can be a DN (Directory Number) that does not have a number associated with it yet. An **Endpoint Profile** enables the administrator to set parameters for that endpoint.

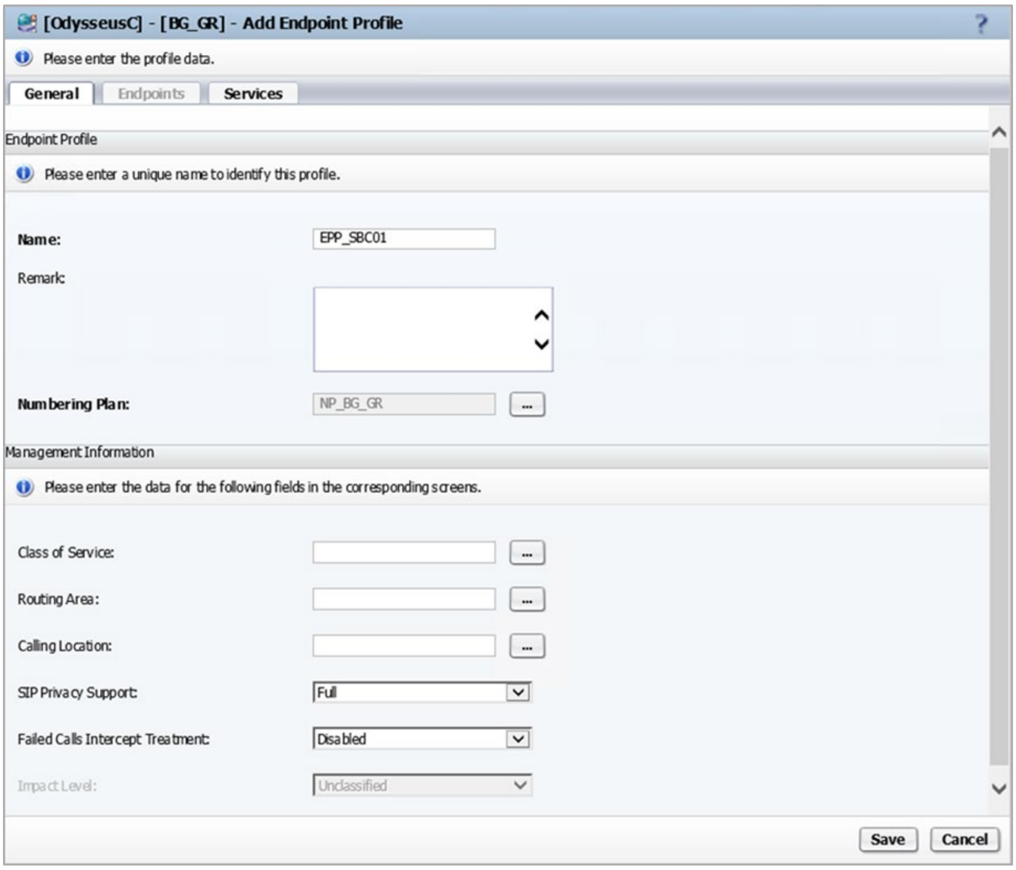
2.1.1 SBC Endpoint Configuration

1. Go to **OpenScape Common Management Platform > Configuration > OpenScape Voice > Business Group > Profiles > Endpoint** to configure the Endpoint Profile.

2. Click **Add.**

3. In the **General** tab, configure the following:

   - **Name**: EPP_Teams
     (a common-sense name)
   - **SIP Privacy Support**: Full



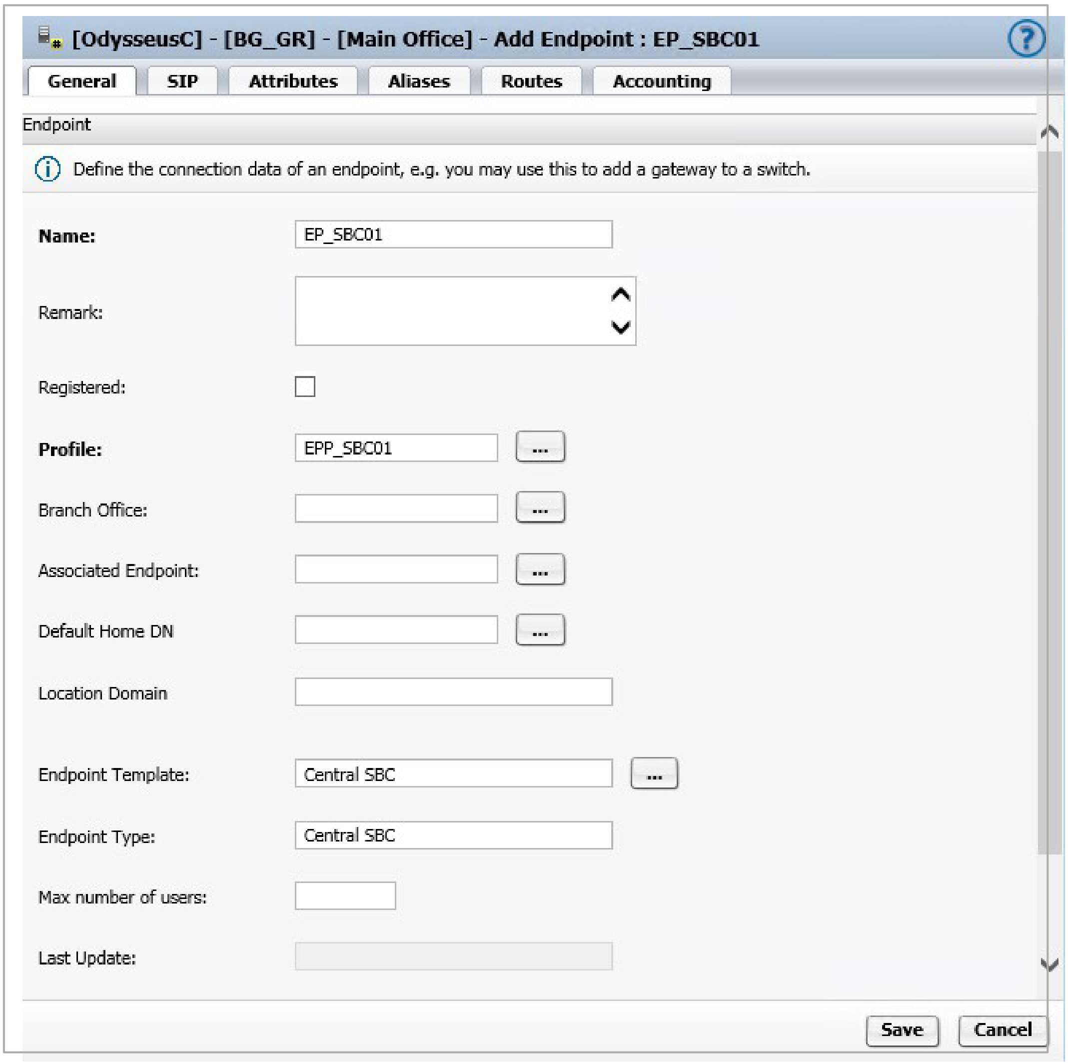
4. In the **Services** tab, from the **Call Transfer** drop-down menu, select **Yes**.
5. Click **Save**.

6. Navigate to **OpenScape Common Management Platform > Configuration > OpenScape Voice > Business Group > Members > Endpoints** to configure the **Endpoint**.

7. Click **Add**.

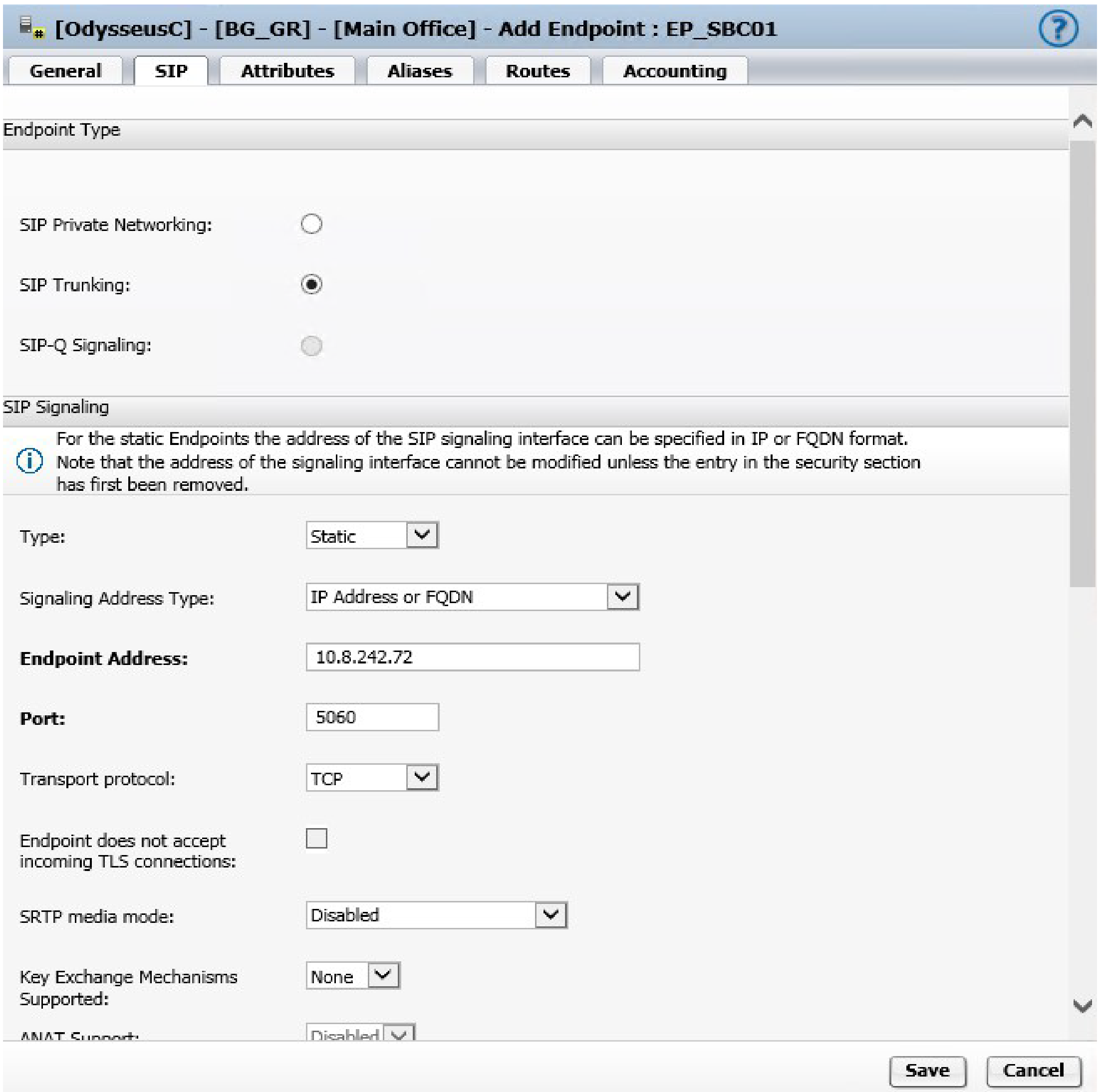
8. In the **General** tab, enter the following:

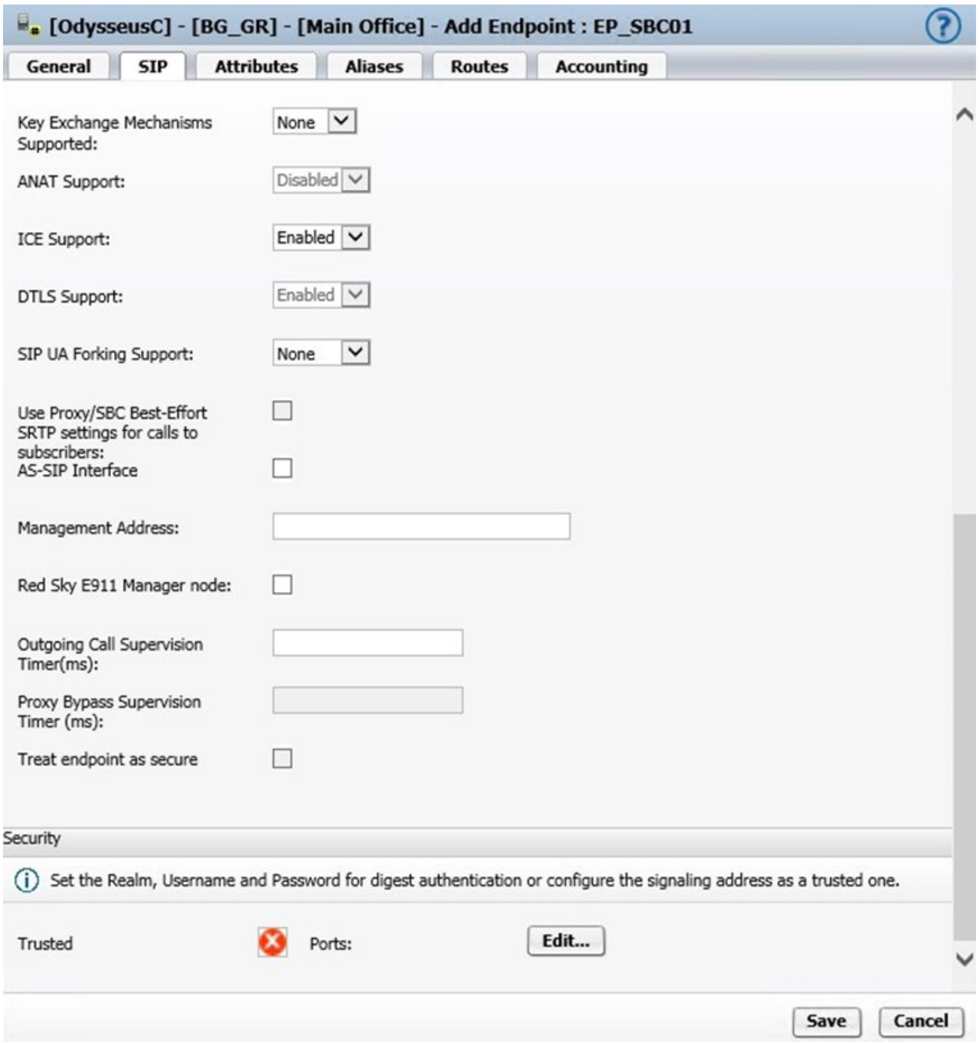
   - **Name**: EP_SBC01
     (a common-sense name).
   - **Profile**: EPP_SBC01
     (select previously created endpoint profile).
   - **Endpoint Template**: Central SBC (set of pre-configured endpoint attributes)



9. Select the **SIP** tab and configure the following:

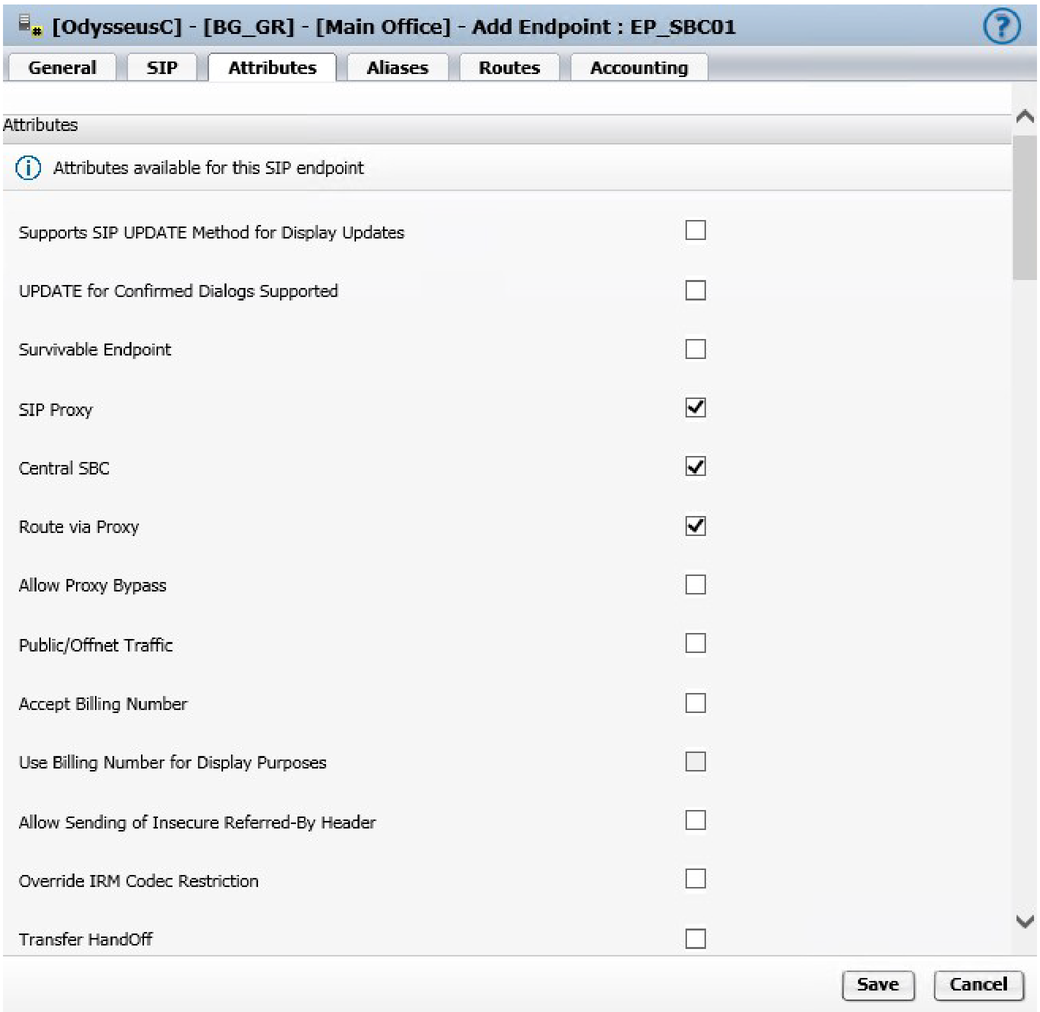
- **SIP Trunking**: Activated
- **Type**: Static
  (it can be enabled only if the **SIP Proxy** attribute is enabled)
- **Signaling Address Type:** IP Address or FQDN (route the calls via proxy)
- **Endpoint Address:** 10.8.242.72 (SBC LAN interface)
- **Port**: 5060
- **Transport protocol:** TCP (UDP, TLS or MTLS are also possible)
- **SRTP media mode**: Disabled
- **Trusted Ports**: All (click **Edit** and **Add** all Ports)

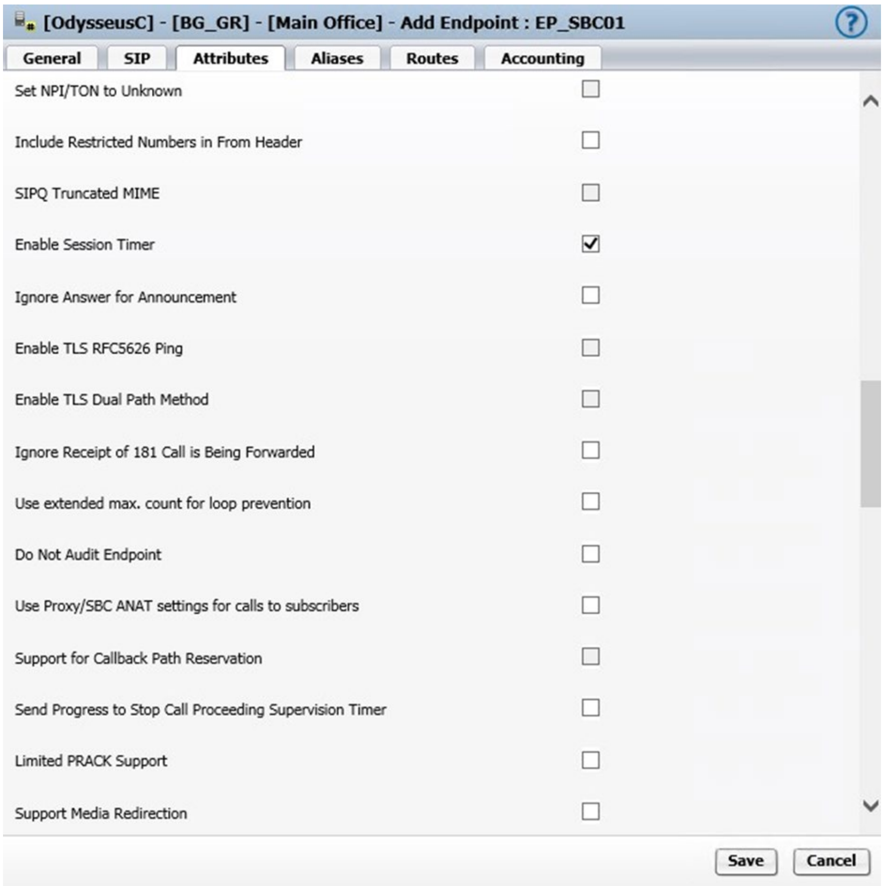



[OdysseusC] - [BG_GR] - [Main Office] - Add Endpoint : EP_SBC01
General
SIP
Attributes
Aliases
Routes
Accounting
Key Exchange Mechanisms Supported:
None
ANAT Support:
Disabled
ICE Support:
Enabled
DTLS Support:
Enabled
SIP UA Forking Support:
None
Use Proxy/SBC Best-Effort SRTP settings for calls to subscribers:
AS-SIP Interface
Management Address:
Red Sky E911 Manager node:
Outgoing Call Supervision Timer(ms):
Proxy Bypass Supervision Timer (ms):
Treat endpoint as secure
Security
Set the Realm, Username and Password for digest authentication or configure the signaling address as a trusted one.
Trusted
Ports:
Edit...
Save
Cancel

10. The **Attributes** tab is populated automatically since the "Central SBC" template was selected in the **General** tab. Make sure that the following are selected:

- **SIP Proxy**: Activated
- **Central SBC**: Activated
- **Route via Proxy**: Activated
- **Enable Session Timer:** Activated





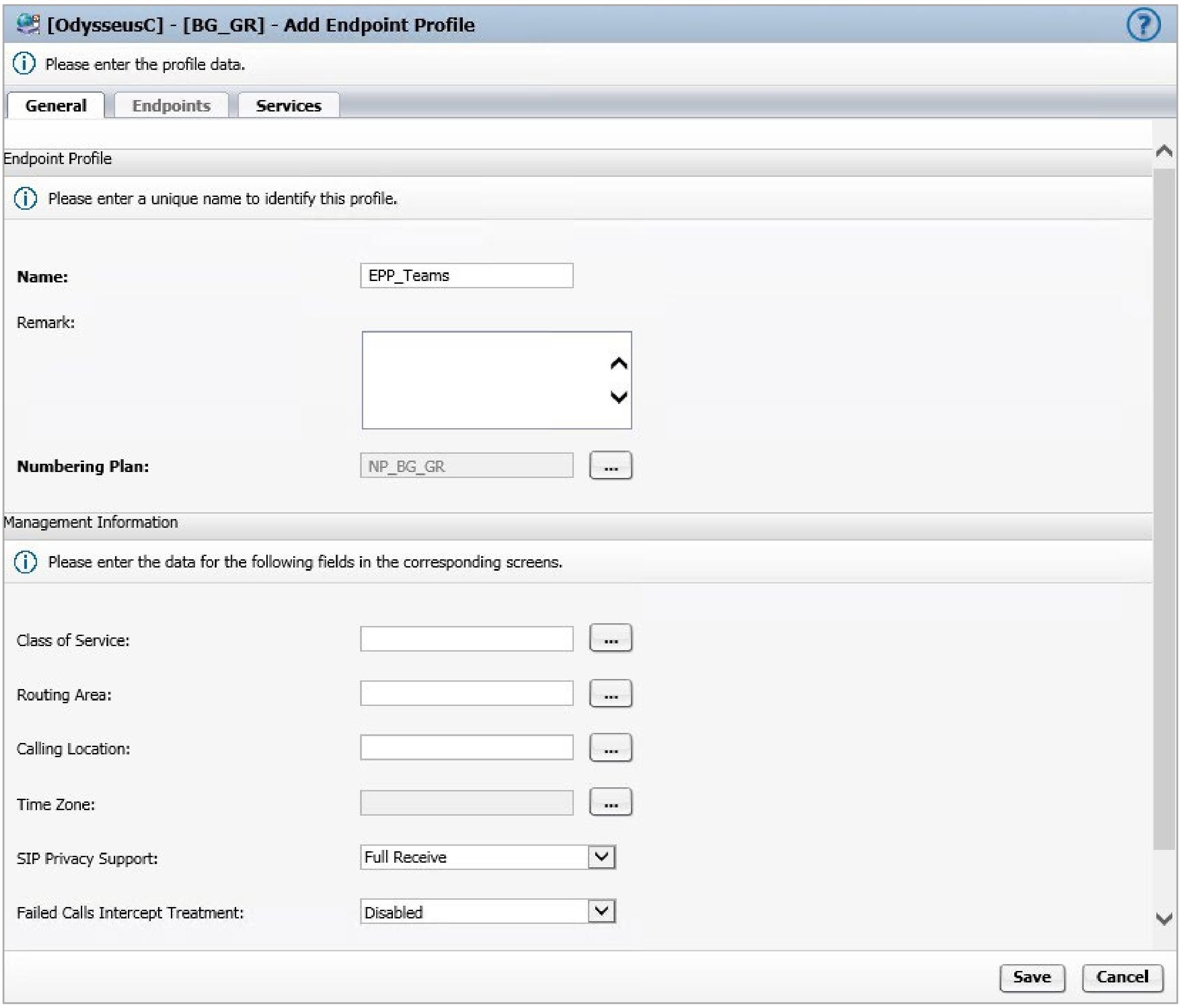
11. Select the **Aliases** tab and click **Add**. Enter the following:

- **Name**: 10.8.242.72
  (the SBC LAN interface for incoming SIP traffic; if there is a need to restrict the port 5060, the value 10.8.242.72:5060 should be entered, instead).

12. Click **OK** and then click **Save**.

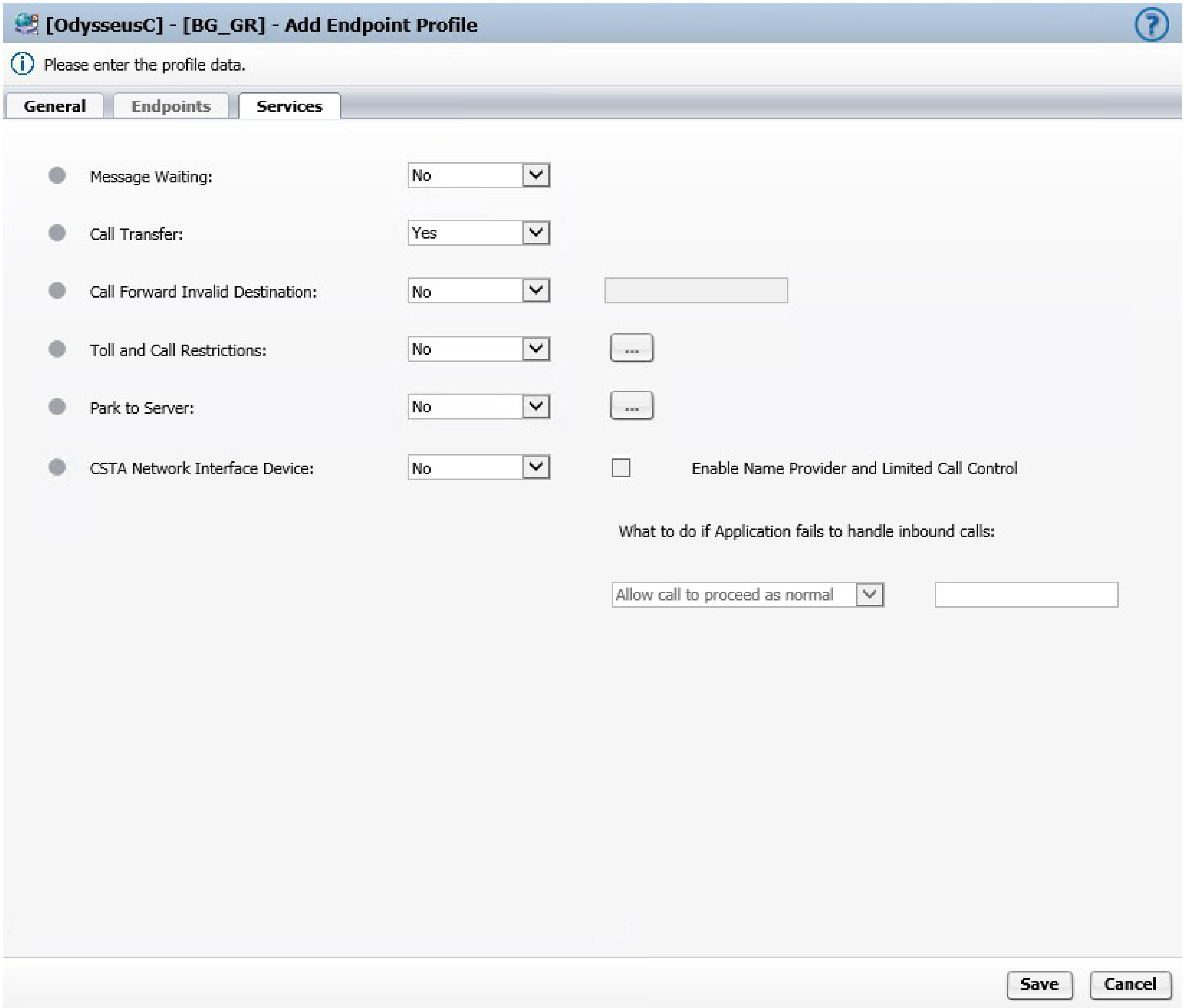
2.1.2 MS Teams Endpoint Configuration

1. Go to **OpenScape Common Management Platform > Configuration > OpenScape Voice > Business Group > Profiles > Endpoint** to configure the Endpoint Profile.
2. Click **Add.**



3. In the **General** tab, configure the following:
   - **Name**: EPP_Teams (a common-sense name)
   - **Numbering Plan:** NP_BG_GR
     (All translations, destinations and routes regarding MS teams should be assigned to the Numbering Plan of Teams Endpoint Profiles)
   - **SIP Privacy Support**: Full Receive
     (the OS Voice does not send a P-Asserted-Identity header field in the messages (requests or responses) to the endpoint. However, the OS Voice SHALL accept any received P-Asserted-Identity header fields)

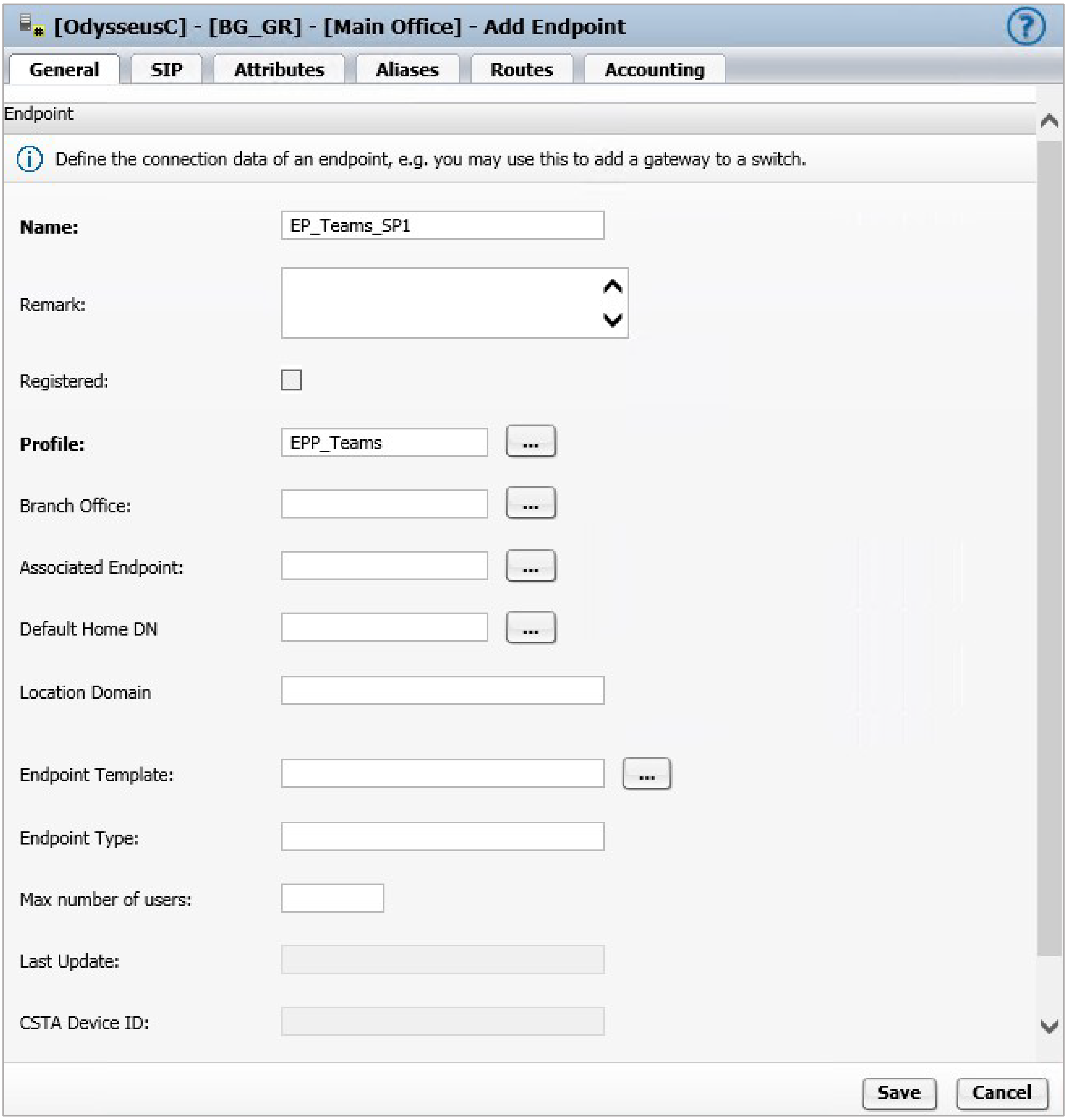
4. In the **Services** tab, from the **Call Transfer** drop-down menu, select Yes.



5. Click **Save**.

6. Go to **OpenScape Common Management Platform > Configuration > OpenScape Voice > Business Group > Members > Endpoints** to configure the Endpoint.

7. Click **Add**.

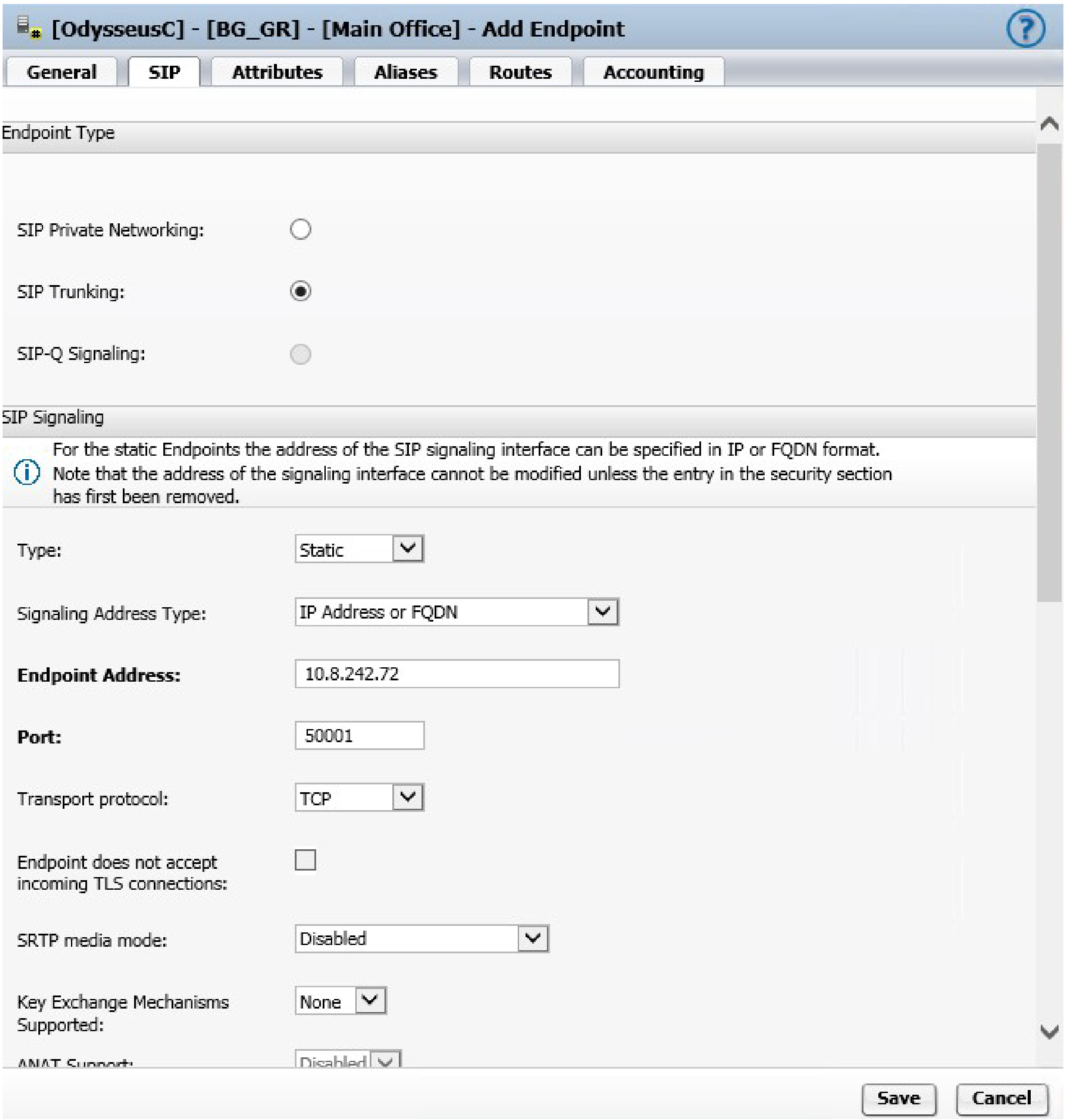


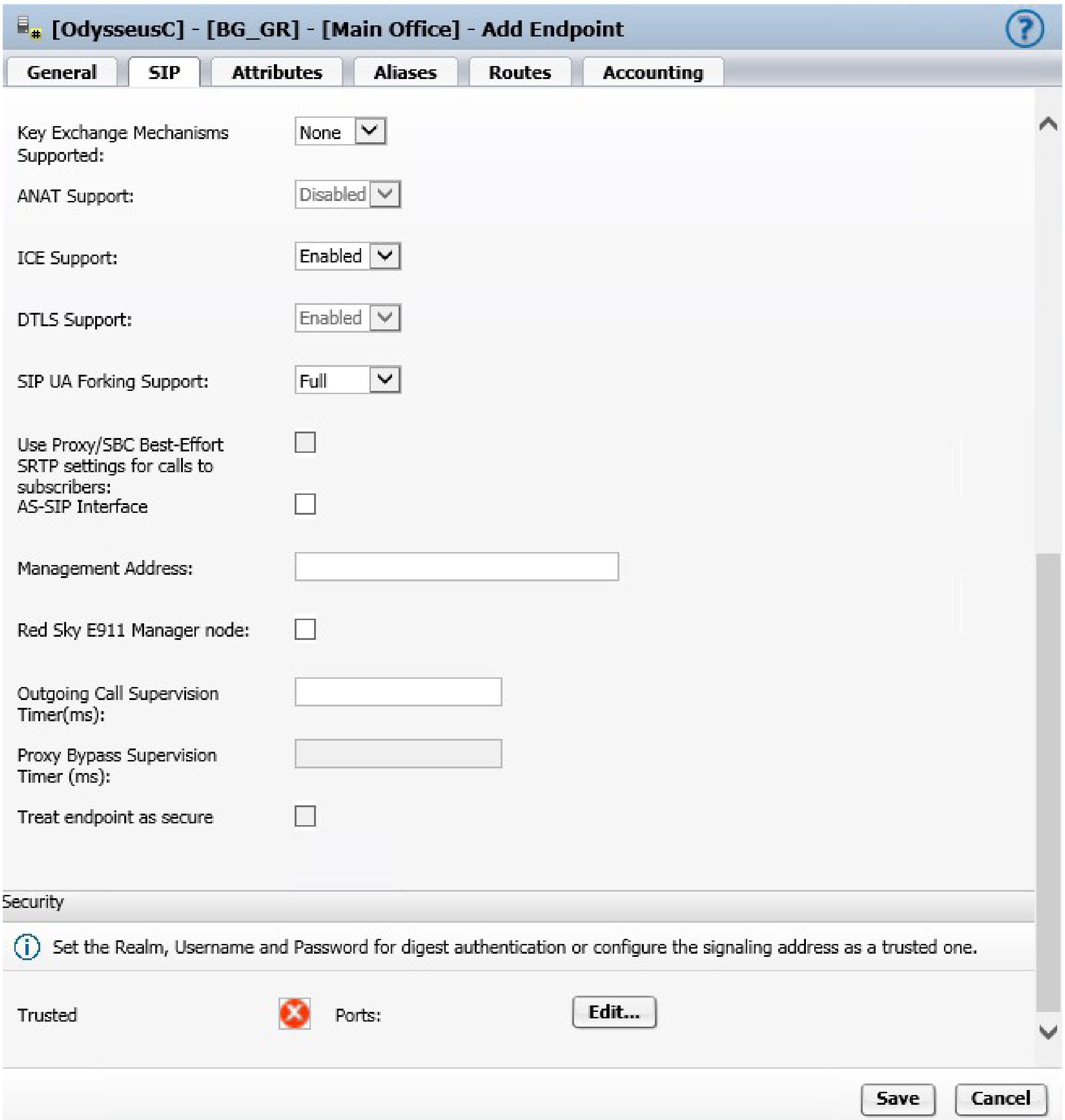
8. In the **General** tab, enter the following:

- **Name**: EP_Teams_SP1 (a common-sense name).
- **Profile**: EPP_Teams (select previously created endpoint profile).

9. Select the **SIP** tab and enter the following:

- **SIP Trunking**: Activated
- **Type**: Static
- **Signaling Address Type**: IP Address or FQDN
- **Endpoint Address:** 10.8.242.72 (SBC LAN interface)
- **Port:** 50001
  (will be configured in SBC for sip.pstnhub.microsoft.com trunk)
- **Transport protocol:** TCP (UDP, TLS or MTLS are also possible)
- **SRTP media mode**: Disabled
- **ICE Support:** Enabled (default value for static endpoints)
- **SIP UA Forking Support**: Full
  (If activated, SIP UA fully complies with RFC3261 SDP offer/answer rules)
- **Trusted** Ports: All
  (click **Edit** and **Add** all Ports)



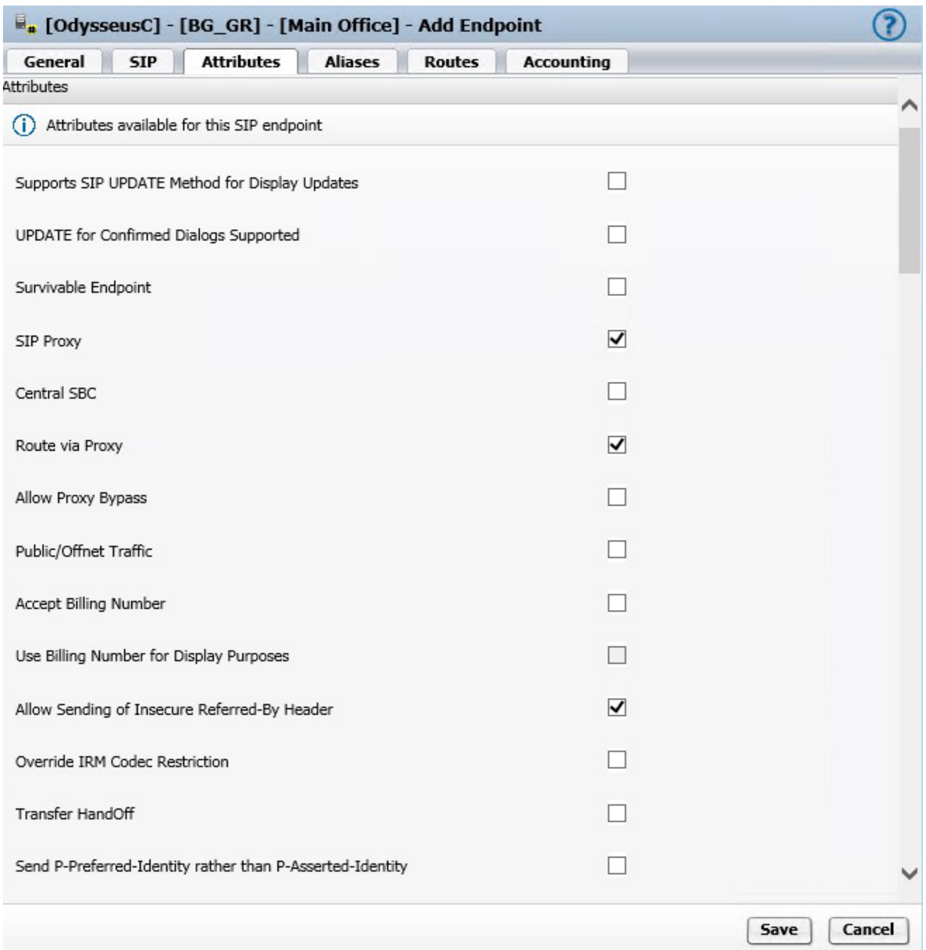

[OdysseusC] - [BG_GR] - [Main Office] - Add Endpoint
General
SIP
Attributes
Aliases
Routes
Accounting
Key Exchange Mechanisms Supported:
None
ANAT Support:
Disabled
ICE Support:
Enabled
DTLS Support:
Enabled
SIP UA Forking Support:
Full
Use Proxy/SBC Best-Effort SRTP settings for calls to subscribers:
AS-SIP Interface
Management Address:
Red Sky E911 Manager node:
Outgoing Call Supervision Timer(ms):
Proxy Bypass Supervision Timer (ms):
Treat endpoint as secure
Security
Set the Realm, Username and Password for digest authentication or configure the signaling address as a trusted one.
Trusted
Ports:
Edit...
Save
Cancel

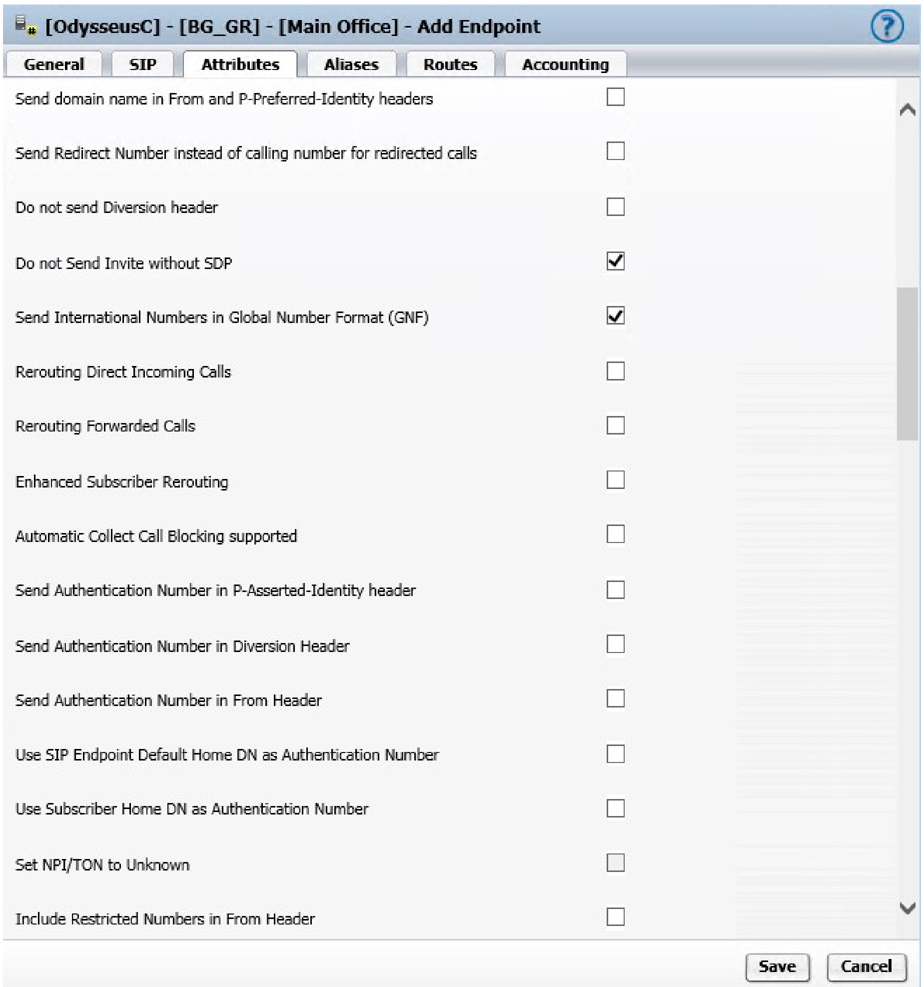
10. Go to the **Attributes** tab and configure the following:

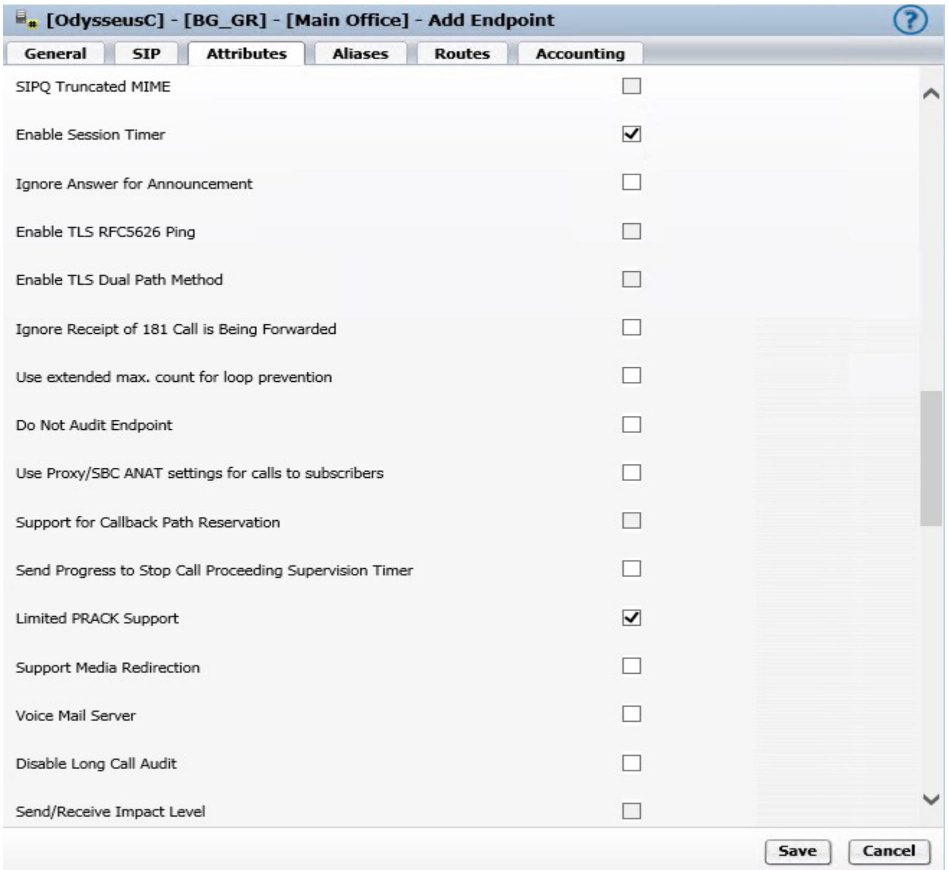
For both **Non-Media Bypass Mode** and **Media Bypass Mode,** select the following attributes to activate them:

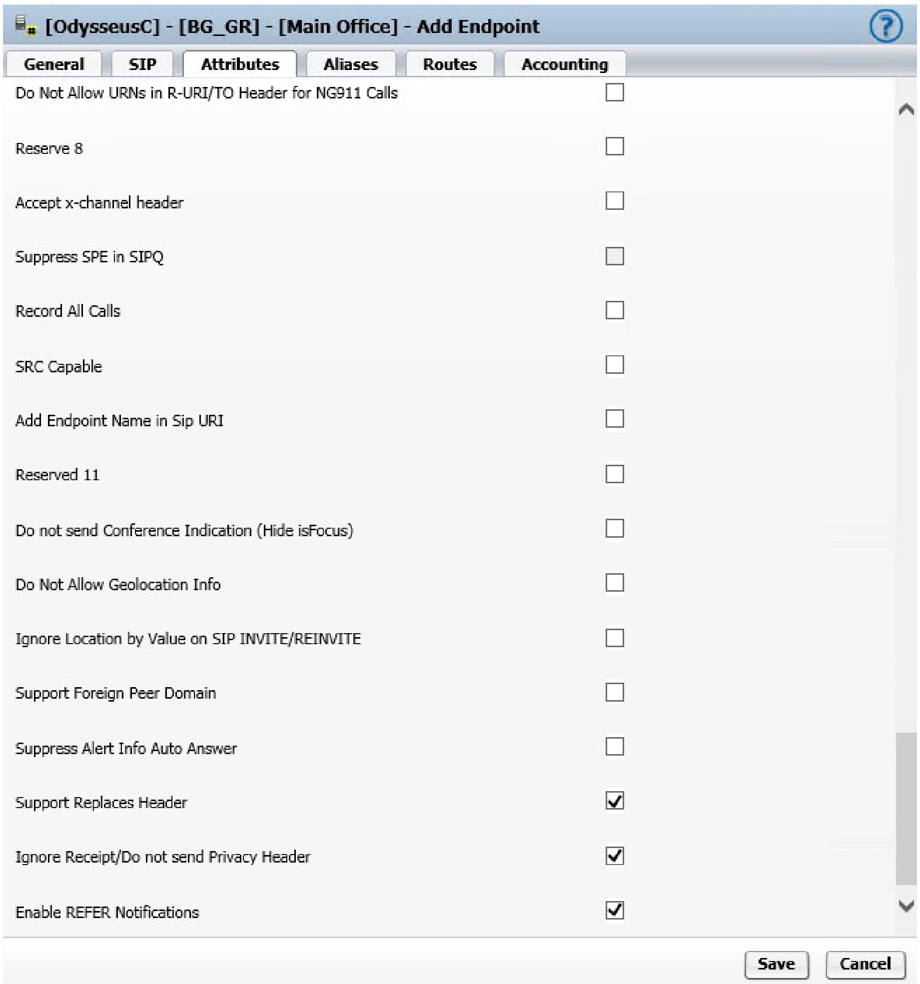
- **SIP Proxy**: Activated
- **Route via Proxy:** Activated
  (route the calls via proxy)

- **Allow Sending of Insecure Referred-By Header:** Activated
  (this attribute makes sure, that to the right call account)
- **Do not Send Invite without SDP:** Activated
  (SIP re-INVITE requests that do not include SDP will not be sent during redirection procedures; OSV will reuse the SDP previously received from the endpoint to send as an SDP offer to the new partner endpoint; when the SDP answer is received the new SDP will be sent in a re-INVITE and the 200 OK answer will be consumed by OS Voice)
- **Send International Numbers in GNF**: Activated
  (the OS Voice adds a '+' in front of all numbers which have NPI = PUBLIC / NOA = INTERNATIONAL and to do this, both the Translation and the Display Number Modification tables MUST be provisioned to send numbers with NPI = PUBLIC / NOA = INTERNATIONAL to this endpoint)
- **Enable Session Timer:** Activated
- **Limited PRACK Support**: Activated
  (the PRACK-Lite feature provides a limited form of RFC3262 PRACK within OS Voice, supporting PRACK on a half-call basis and only for SIP network-network interfaces)
- **Ignore Receipt/Do not send Privacy Header**: Activated
  (when PAI is activated in Teams Direct Routing configuration for the SBC endpoint, Teams sends a *"Privacy:id"* header which causes anonymous call display at called party and with this attribute OS strips this header from PAI)




[OdysseusC] - [BG_GR] - [Main Office] - Add Endpoint
General
SIP
Attributes
Aliases
Routes
Accounting
Send domain name in From and P-Preferred-Identity headers
Send Redirect Number instead of calling number for redirected calls
Do not send Diversion header
Do not Send Invite without SDP
Send International Numbers in Global Number Format (GNF)
Rerouting Direct Incoming Calls
Rerouting Forwarded Calls
Enhanced Subscriber Rerouting
Automatic Collect Call Blocking supported
Send Authentication Number in P-Asserted-Identity header
Send Authentication Number in Diversion Header
Send Authentication Number in From Header
Use SIP Endpoint Default Home DN as Authentication Number
Use Subscriber Home DN as Authentication Number
Set NPI/TON to Unknown
Include Restricted Numbers in From Header
Save
Cancel


[OdysseusC] - [BG_GR] - [Main Office] - Add Endpoint
General
SIP
Attributes
Aliases
Routes
Accounting
SIPQ Truncated MIME
Enable Session Timer
Ignore Answer for Announcement
Enable TLS RFC5626 Ping
Enable TLS Dual Path Method
Ignore Receipt of 181 Call is Being Forwarded
Use extended max. count for loop prevention
Do Not Audit Endpoint
Use Proxy/SBC ANAT settings for calls to subscribers
Support for Callback Path Reservation
Send Progress to Stop Call Proceeding Supervision Timer
Limited PRACK Support
Support Media Redirection
Voice Mail Server
Disable Long Call Audit
Send/Receive Impact Level
Save
Cancel



For **Media Bypass Mode** only:

a. Select the **Enable REFER Notifications** attribute**.**
b. Go to **OpenScape Common Management Platform > Configuration > OpenScape Voice > Administration > General Settings > RTP**.
c. Click on the name of the "*Srx/Sip/PassiveForkingEnable*" parameter.

   The **Edit RTP parameter** window opens.

d. Set the value of the "*Srx/Sip/PassiveForkingEnable*" parameter to **True**.

For **Non-Media Bypass Mode** only:

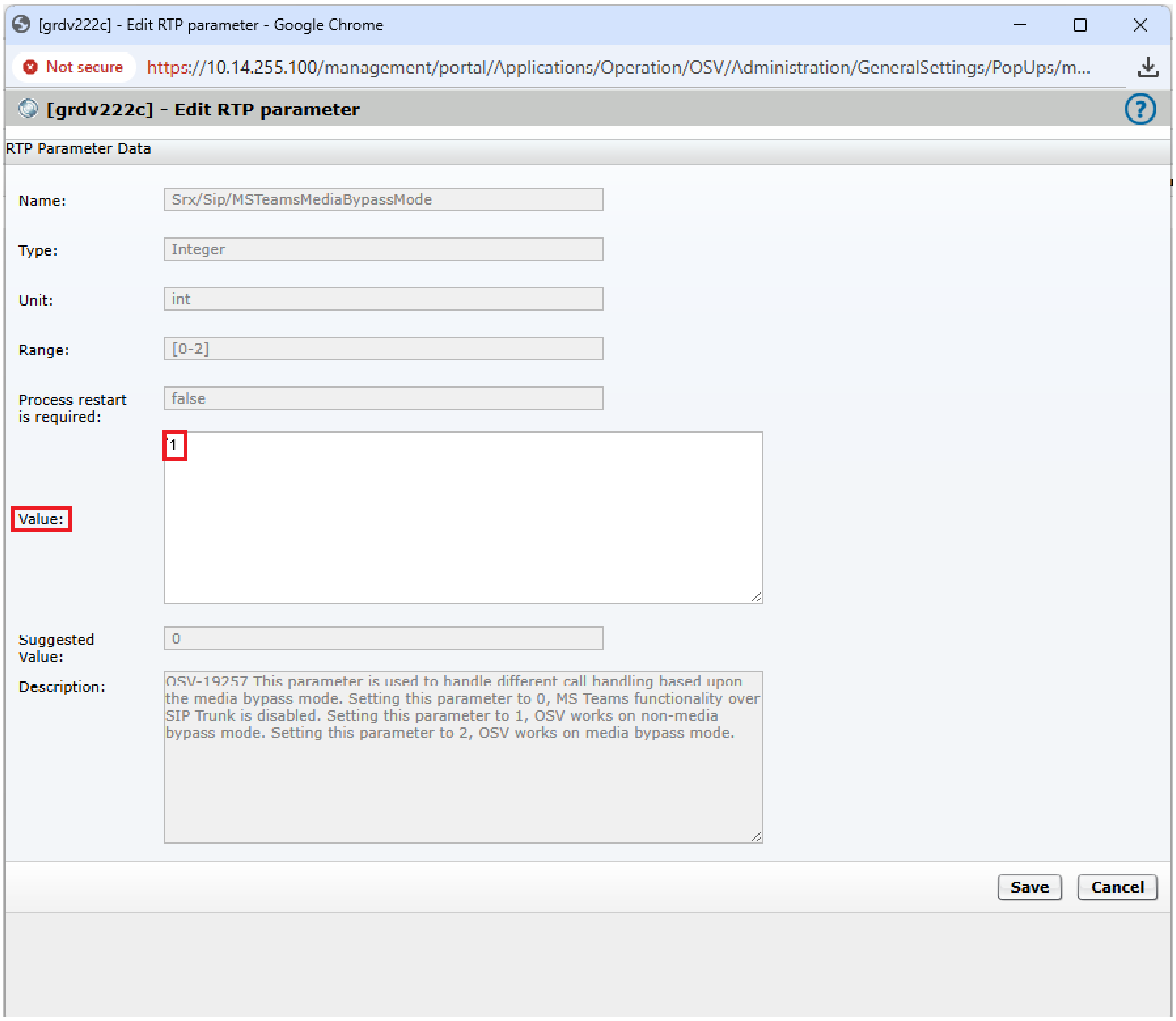
a. Select the **Support Replaces Header** attribute**.**

b. Go to **OpenScape Common Management Platform > Configuration > OpenScape Voice > Administration > General Settings > RTP**.

c. Click on the name of the "*Srx/Sip/MSTeamsMediaBypassMode*" parameter.

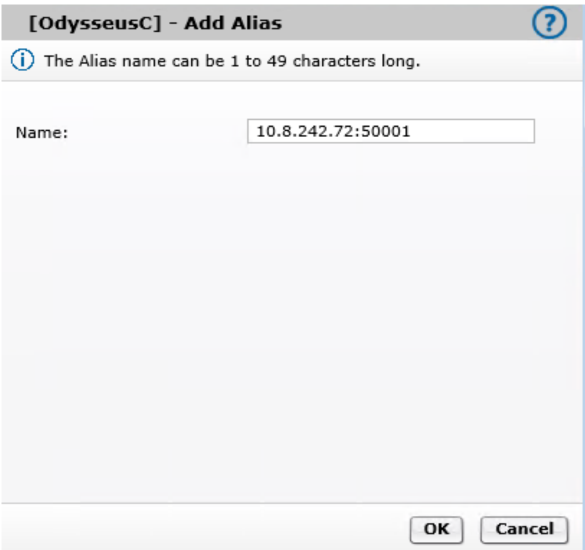
   The **Edit RTP parameter** window opens.

d. Set the value of the "*Srx/Sip/MSTeamsMediaBypassMode*" parameter to "**one" (1).**



11. Select the **Aliases** tab and click **Add**.

- In the **Name** field, enter 10.8.242.72:50001
(the SBC LAN interface for incoming SIP traffic restricted for port 50001, that corresponds to Teams SIP Proxy 1 related traffic)



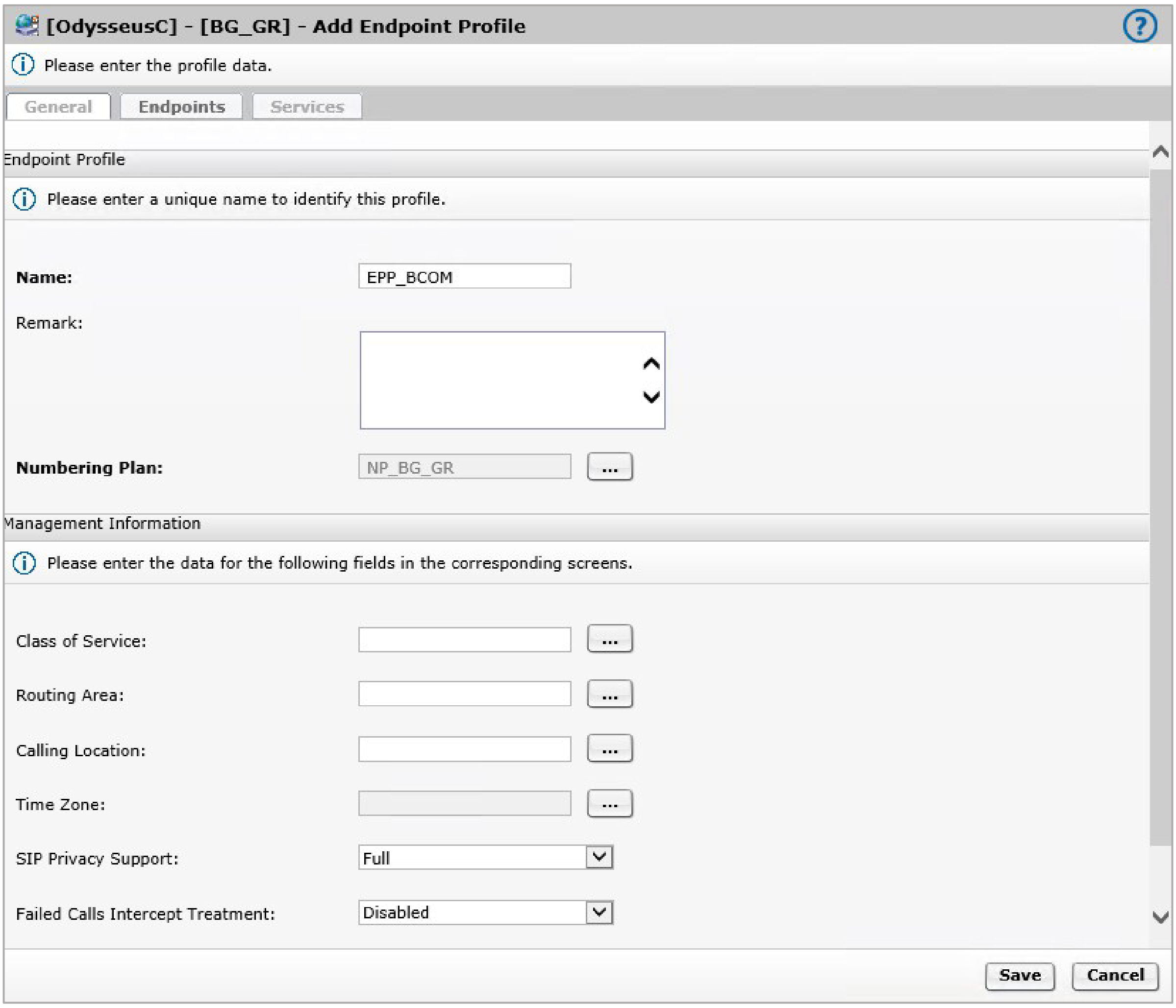
12. Click **OK** and then click **Save**.

13. Repeat the same procedure to create the endpoints for the remaining Teams FQDNs:
    - EP_Teams_SP2 with port 50002
      (for sip2.pstnhub.microsoft.com)
    - EP_Teams_SP3 with port 50003
      (for sip3.pstnhub.microsoft.com)
    - EP_Teams_ALL with port 50004
      (for sip-all.pstnhub.microsoft.com)

2.1.3 SSP Endpoint Configuration

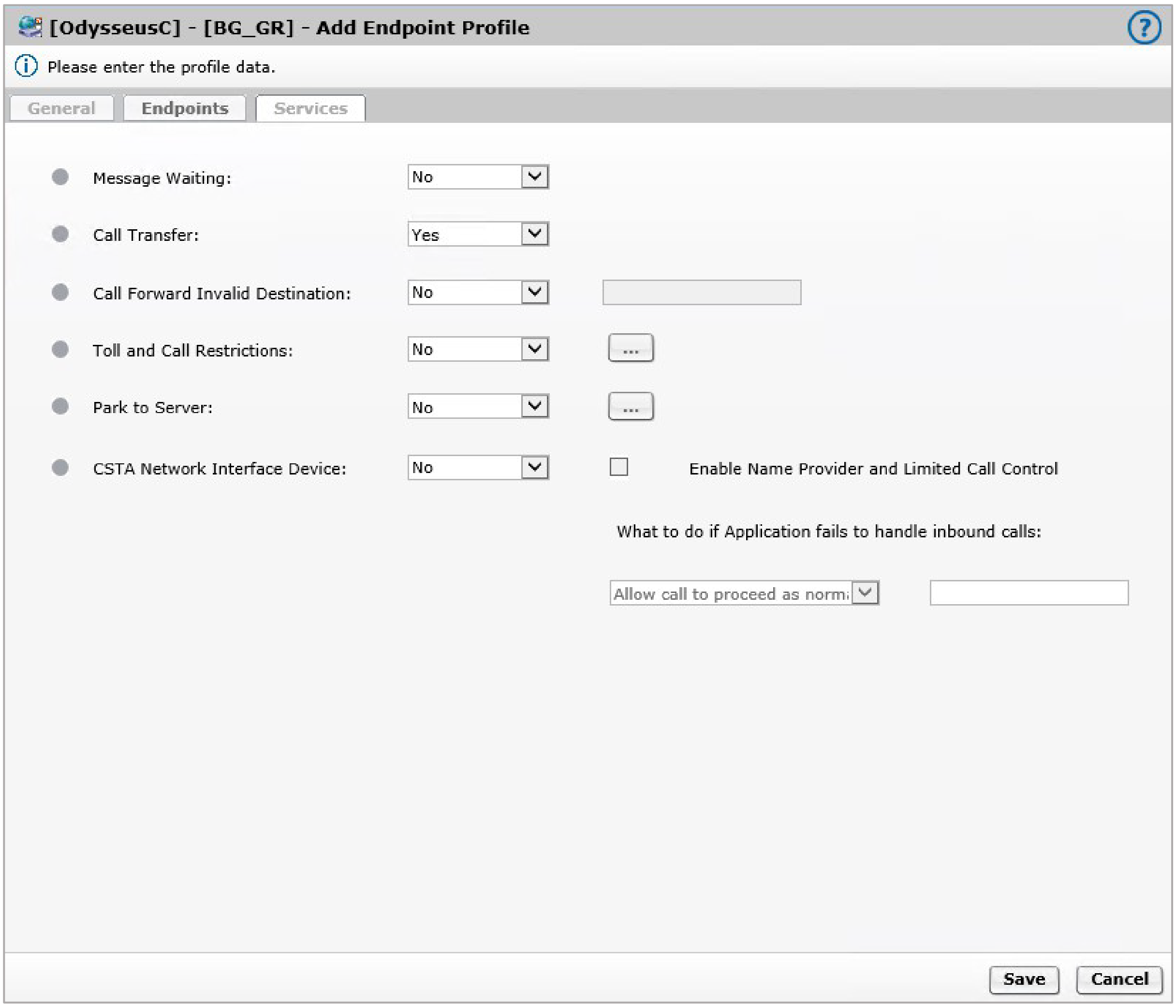
1. Go to OpenScape Common Management Platform > Configuration > OpenScape Voice > Business Group > Profiles > Endpoint to configure the Endpoint Profile.

2. Click **Add.**



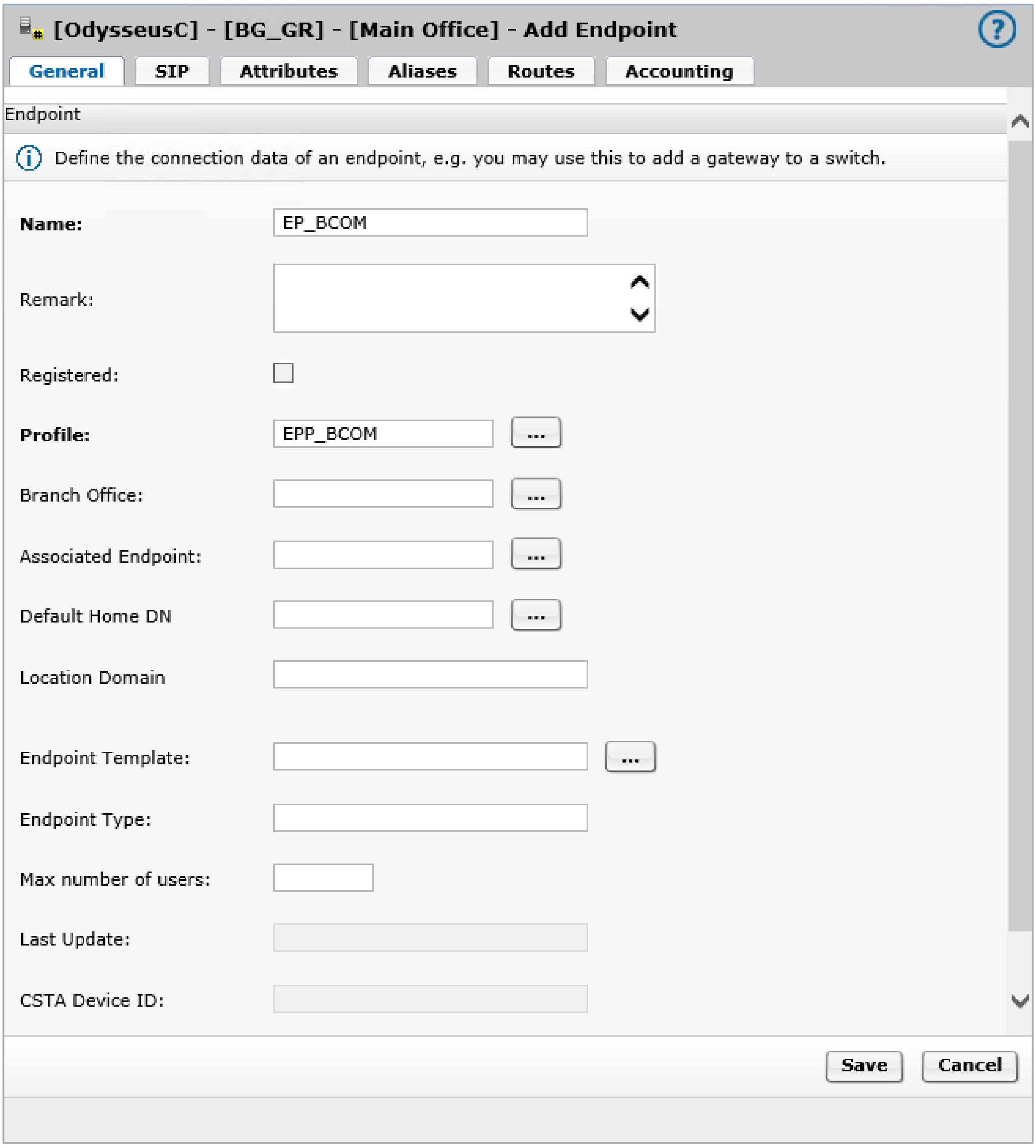
3. In the **General** tab, enter the following:

- **Name**: EPP_BCOM (a common-sense name)
- **SIP Privacy Support**: Full



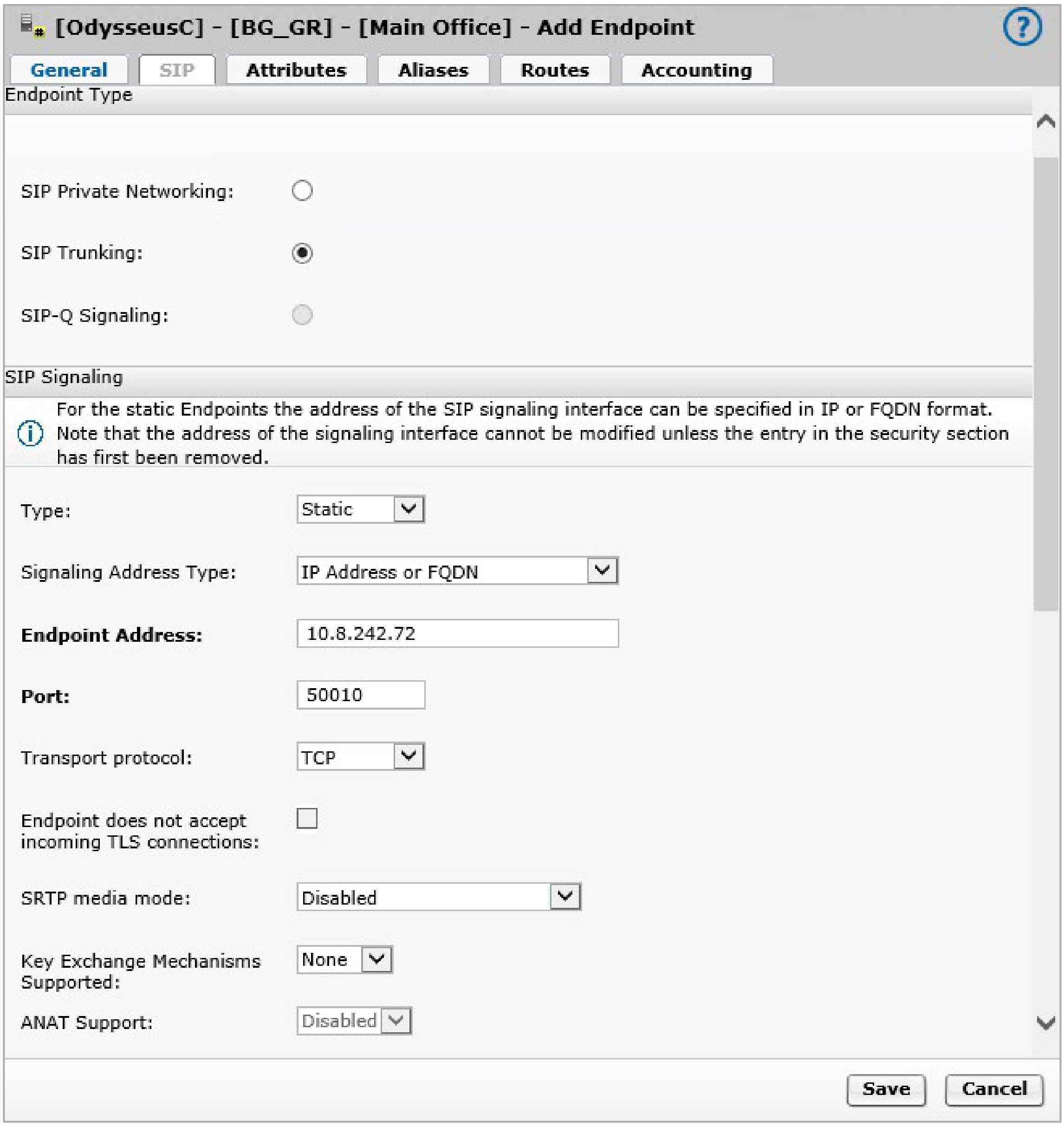
4. In the **Services** tab, from the **Call Transfer** drop-down menu, select **Yes**.
5. Click **Save**.

6. Go to **OpenScape Common Management Platform > Configuration > OpenScape Voice > Business Group > Members > Endpoints** to configure the Endpoint.
7. Click **Add**.

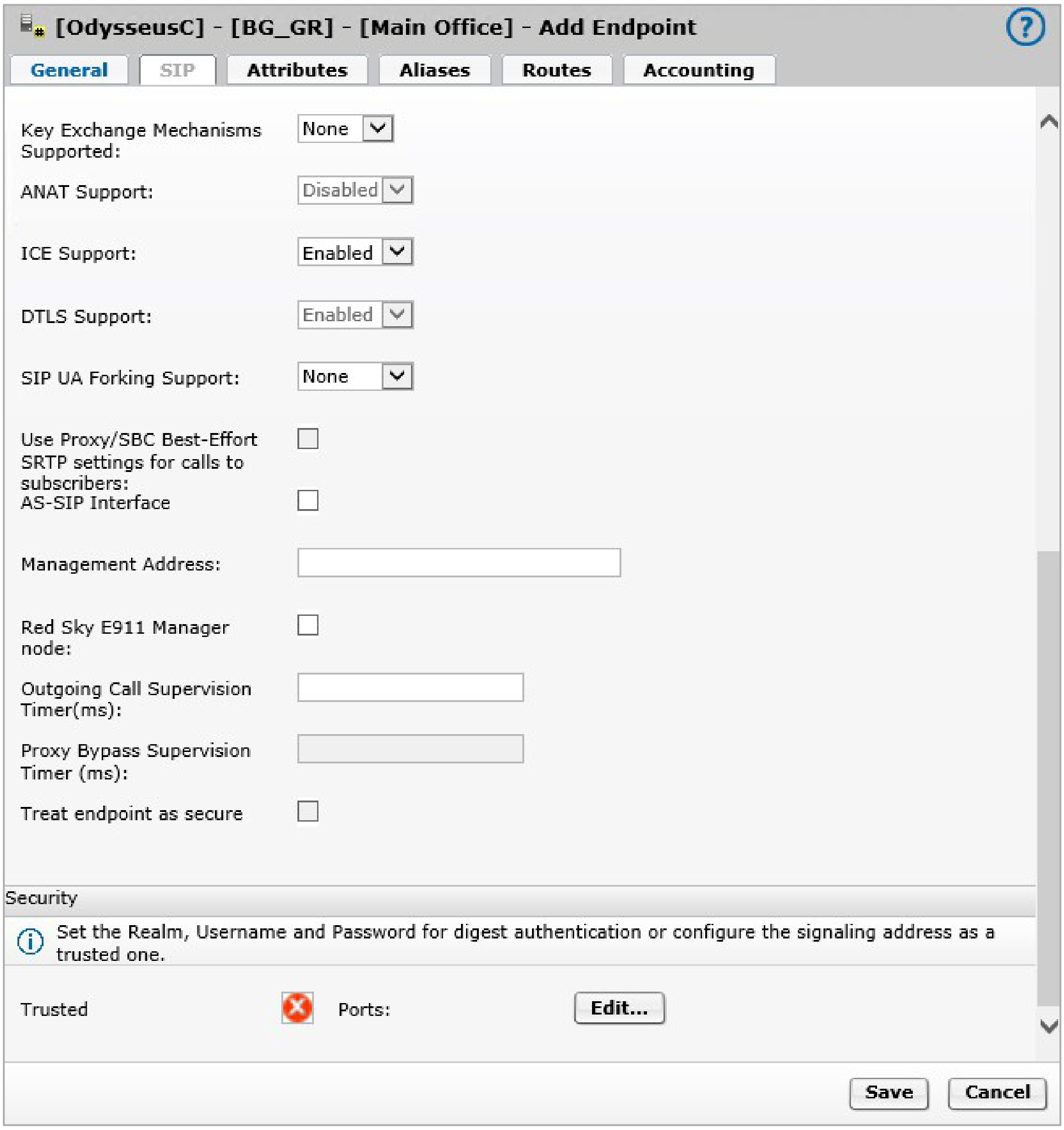


8. In the **General** tab, enter the following:

- **Name**: EP_BCOM (a common-sense name).
- **Profile**: EPP_BCOM (select previously created endpointprofile).



[OdysseusC] - [BG_GR] - [Main Office] - Add Endpoint
General
SIP
Attributes
Aliases
Routes
Accounting
Endpoint Type
SIP Private Networking:
SIP Trunking:
SIP-Q Signaling:
SIP Signaling
For the static Endpoints the address of the SIP signaling interface can be specified in IP or FQDN format. Note that the address of the signaling interface cannot be modified unless the entry in the security section has first been removed.
Type:
Static
Signaling Address Type:
IP Address or FQDN
Endpoint Address:
10.8.242.72
Port:
50010
Transport protocol:
TCP
Endpoint does not accept incoming TLS connections:
SRTP media mode:
Disabled
Key Exchange Mechanisms Supported:
None
ANAT Support:
Disabled
Save
Cancel



9. Select the **SIP** tab and configure the following:

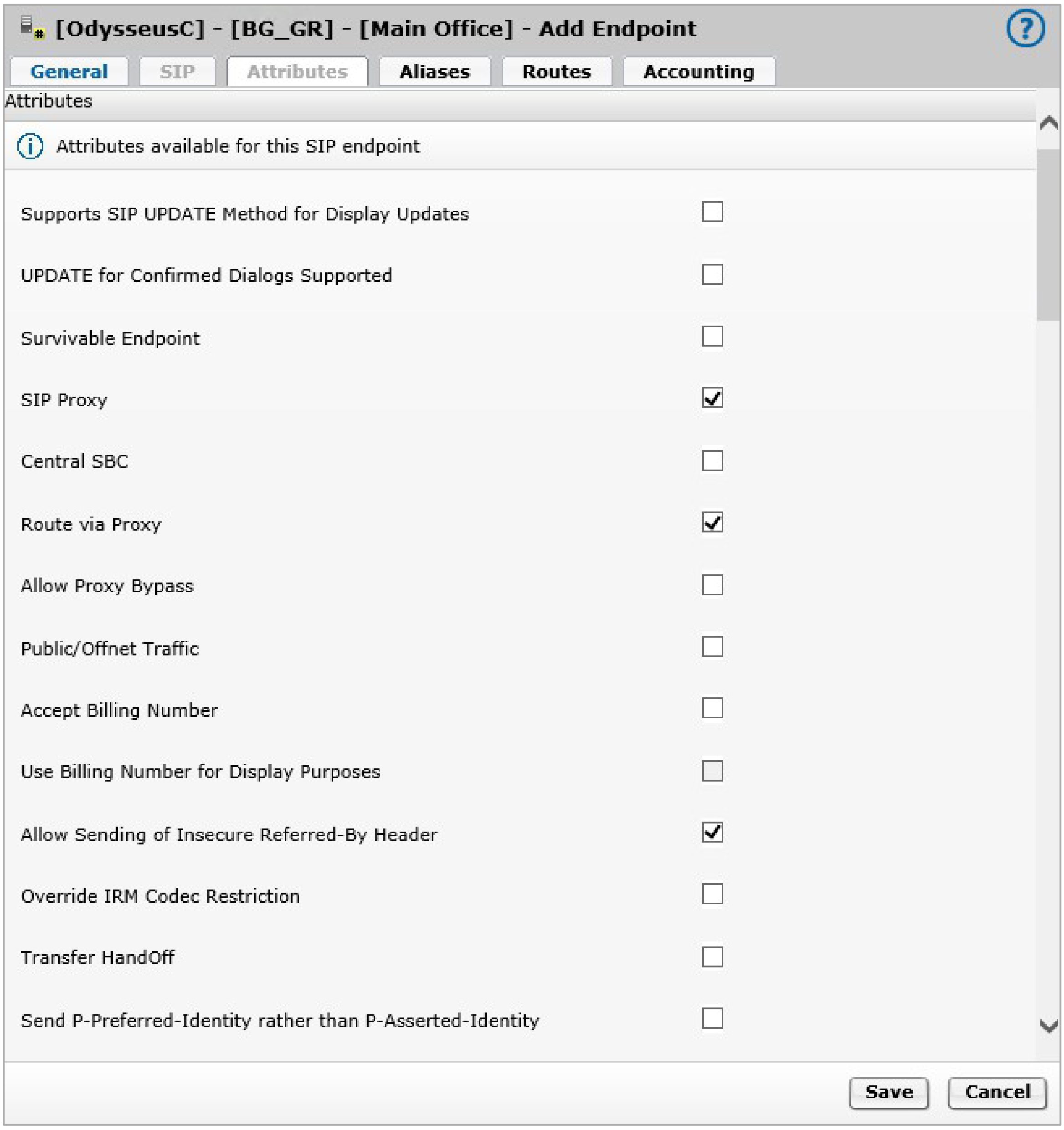
- **SIP Trunking**: Activated
- **Type**: Static
- **Signaling Address Type**: IP Address or FQDN
- **Endpoint Address**: 10.8.242.72 (SBC LAN interface)
- **Port:** 50010
- (will be configured in SBC for sip.pstnhub.microsoft.com trunk)
- **Transport protocol**: TCP (UDP, TLS or MTLS are also possible)
- **SRTP media mode:** Disabled
- **ICE Support:** Enabled (default value for static endpoints)
- **SIP UA Forking Support**: None
- (if activated, SIP UA fully complies with RFC3261 SDP offer/answer rules)
- **Trusted** Ports: All (click on **[Edit]** and **[Add]** all Ports)

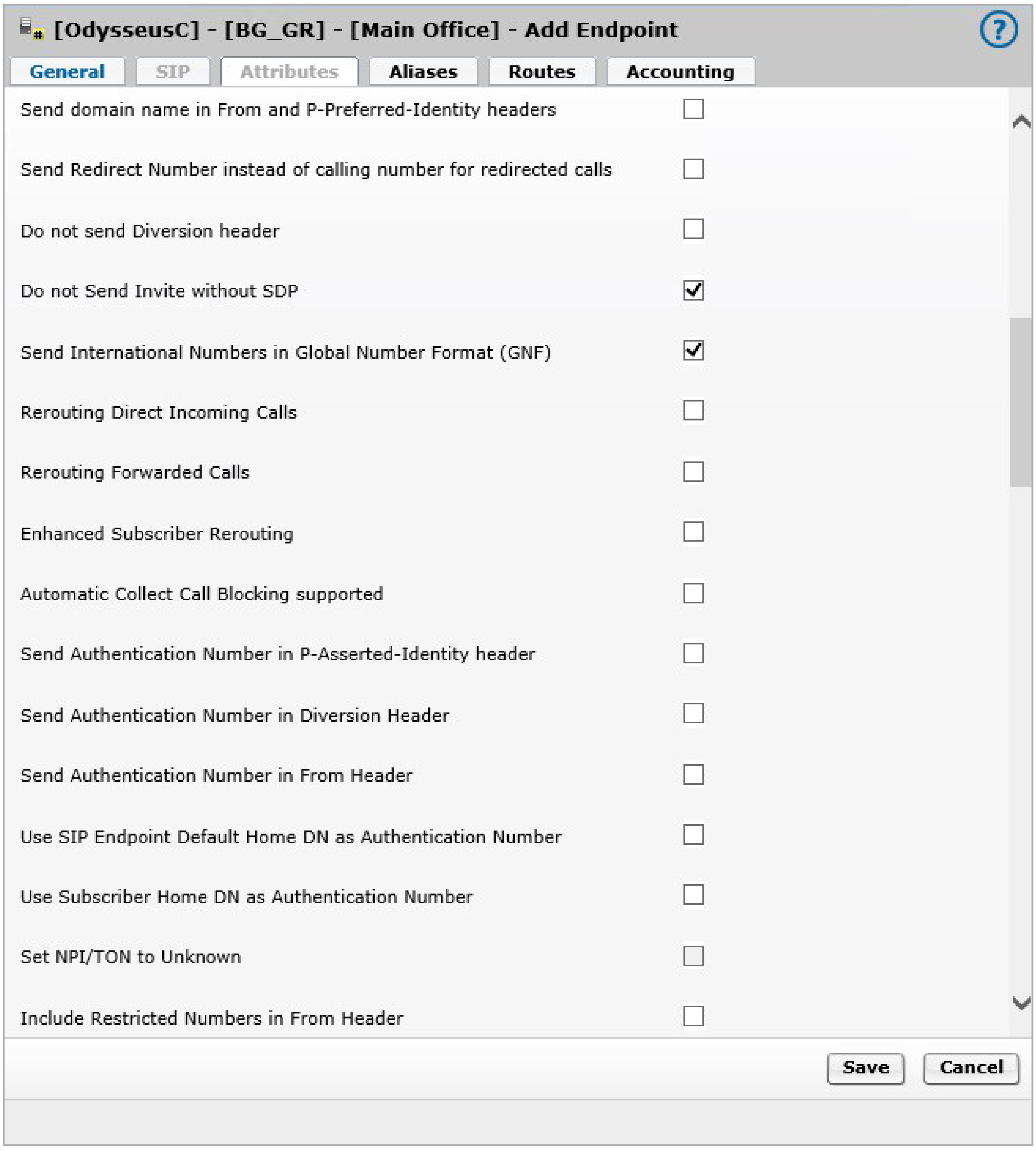
10. Select the **Attributes** tab and configure the following:

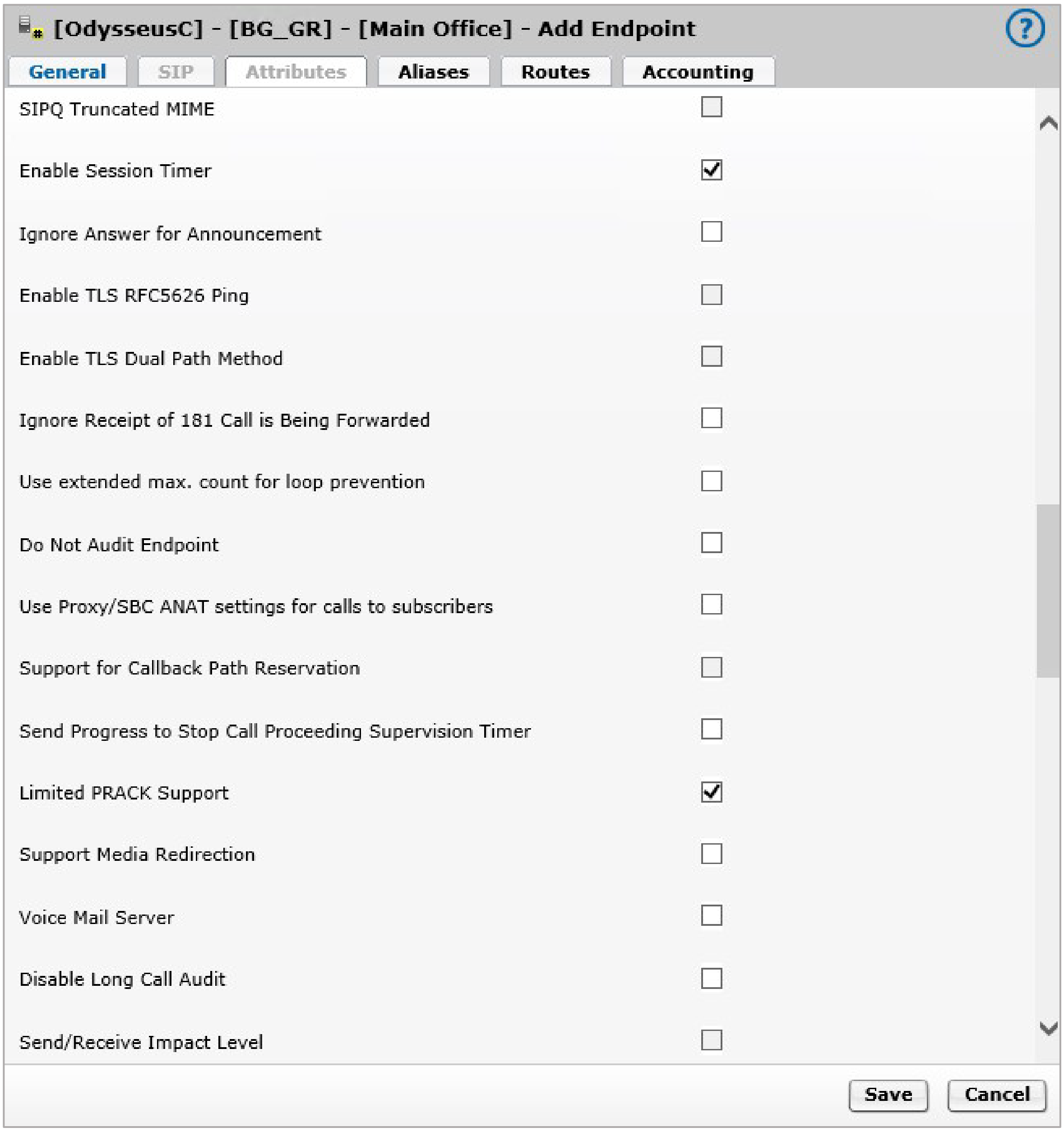
- **SIP Proxy**: Activated
- **Route via Proxy:** Activated
  (route the calls via proxy)
- **Allow Sending of Insecure Referred-By Header**: Activated
  (this attribute makes sure, that calls get charged to the right call account)
- **Do not Send Invite without SDP:** Activated
  (SIP re-INVITE requests that do not include SDP will not be sent during redirection procedures; OSV will reuse the SDP previously received from the endpoint to send as an SDP offer to the new partner endpoint; when the SDP answer is received the new SDP will be sent in a re-INVITE and the 200 OK answer will be consumed by OS Voice)
- **Send International Numbers in GNF**: Activated
  (the OS Voice adds a '+' in front of all numbers which have NPI = PUBLIC / NOA = INTERNATIONAL and to do this, both the Translation and the Display Modification tables MUST be provisioned to send numbers with NPI = PUBLIC / NOA = INTERNATIONAL to this endpoint)
- **Enable Session Timer**: Activated
- **Limited PRACK Support**: Activated

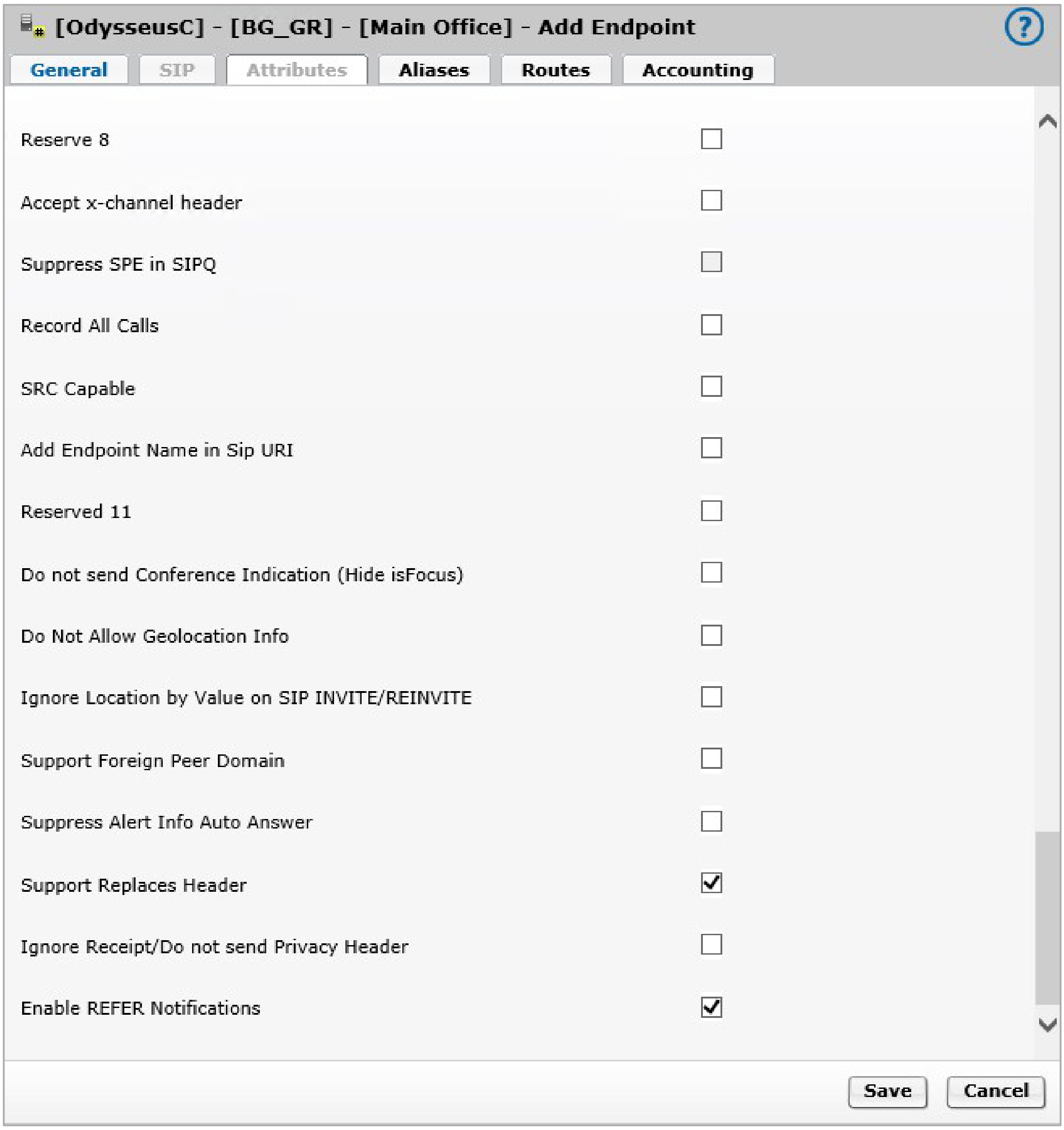
(the PRACK-Lite feature provides a limited form of RFC3262 PRACK within Voice, supporting PRACK on a half-call basis and only for SIP network-network interfaces)

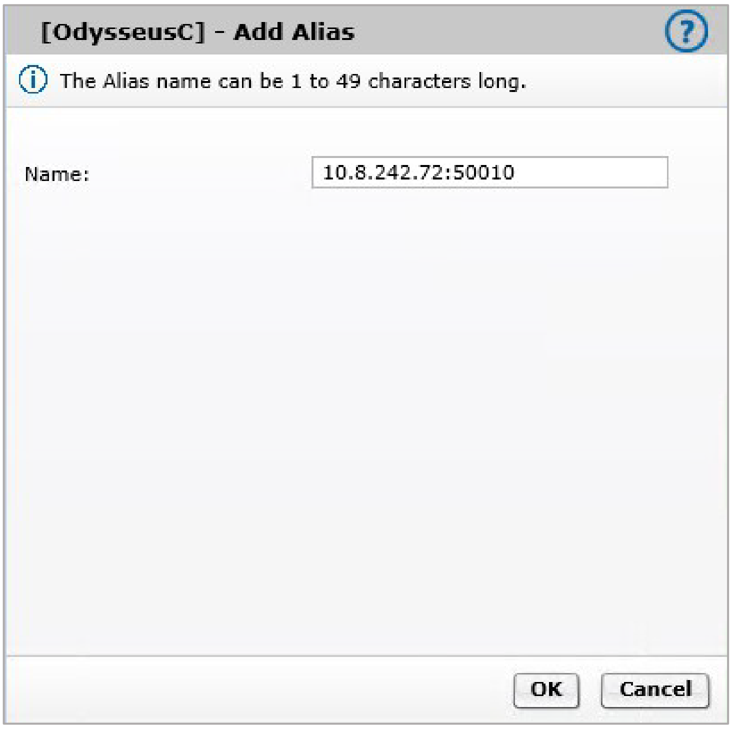
- **Support Replaces Header:** Activated
- **Ignore Receipt/Do not send Privacy Header**: Deactivated
- **Enable REFER Notifications**: Activated


[OdysseusC] - [BG_GR] - [Main Office] - Add Endpoint
General
SIP
Attributes
Aliases
Routes
Accounting
Attributes
Attributes available for this SIP endpoint
Supports SIP UPDATE Method for Display Updates
UPDATE for Confirmed Dialogs Supported
Survivable Endpoint
SIP Proxy
Central SBC
Route via Proxy
Allow Proxy Bypass
Public/Offnet Traffic
Accept Billing Number
Use Billing Number for Display Purposes
Allow Sending of Insecure Referred-By Header
Override IRM Codec Restriction
Transfer HandOff
Send P-Preferred-Identity rather than P-Asserted-Identity
Save
Cancel


[OdysseusC] - [BG_GR] - [Main Office] - Add Endpoint
General
SIP
Attributes
Aliases
Routes
Accounting
Send domain name in From and P-Preferred-Identity headers
Send Redirect Number instead of calling number for redirected calls
Do not send Diversion header
Do not Send Invite without SDP
Send International Numbers in Global Number Format (GNF)
Rerouting Direct Incoming Calls
Rerouting Forwarded Calls
Enhanced Subscriber Rerouting
Automatic Collect Call Blocking supported
Send Authentication Number in P-Asserted-Identity header
Send Authentication Number in Diversion Header
Send Authentication Number in From Header
Use SIP Endpoint Default Home DN as Authentication Number
Use Subscriber Home DN as Authentication Number
Set NPI/TON to Unknown
Include Restricted Numbers in From Header
Save
Cancel


[OdysseusC] - [BG_GR] - [Main Office] - Add Endpoint
General
SIP
Attributes
Aliases
Routes
Accounting
SIPQ Truncated MIME
Enable Session Timer
Ignore Answer for Announcement
Enable TLS RFC5626 Ping
Enable TLS Dual Path Method
Ignore Receipt of 181 Call is Being Forwarded
Use extended max. count for loop prevention
Do Not Audit Endpoint
Use Proxy/SBC ANAT settings for calls to subscribers
Support for Callback Path Reservation
Send Progress to Stop Call Proceeding Supervision Timer
Limited PRACK Support
Support Media Redirection
Voice Mail Server
Disable Long Call Audit
Send/Receive Impact Level
Save
Cancel


[OdysseusC] - [BG_GR] - [Main Office] - Add Endpoint
General
SIP
Attributes
Aliases
Routes
Accounting
Reserve 8
Accept x-channel header
Suppress SPE in SIPQ
Record All Calls
SRC Capable
Add Endpoint Name in Sip URI
Reserved 11
Do not send Conference Indication (Hide isFocus)
Do Not Allow Geolocation Info
Ignore Location by Value on SIP INVITE/REINVITE
Support Foreign Peer Domain
Suppress Alert Info Auto Answer
Support Replaces Header
Ignore Receipt/Do not send Privacy Header
Enable REFER Notifications
Save
Cancel



11. Select the **Aliases** tab and click **Add**.
    - In the **Name** field, enter 10.8.242.72:50010
      (the SBC LAN interface for incoming SIP traffic restricted for port 50010, that corresponds to BCOM ITSP related traffic)

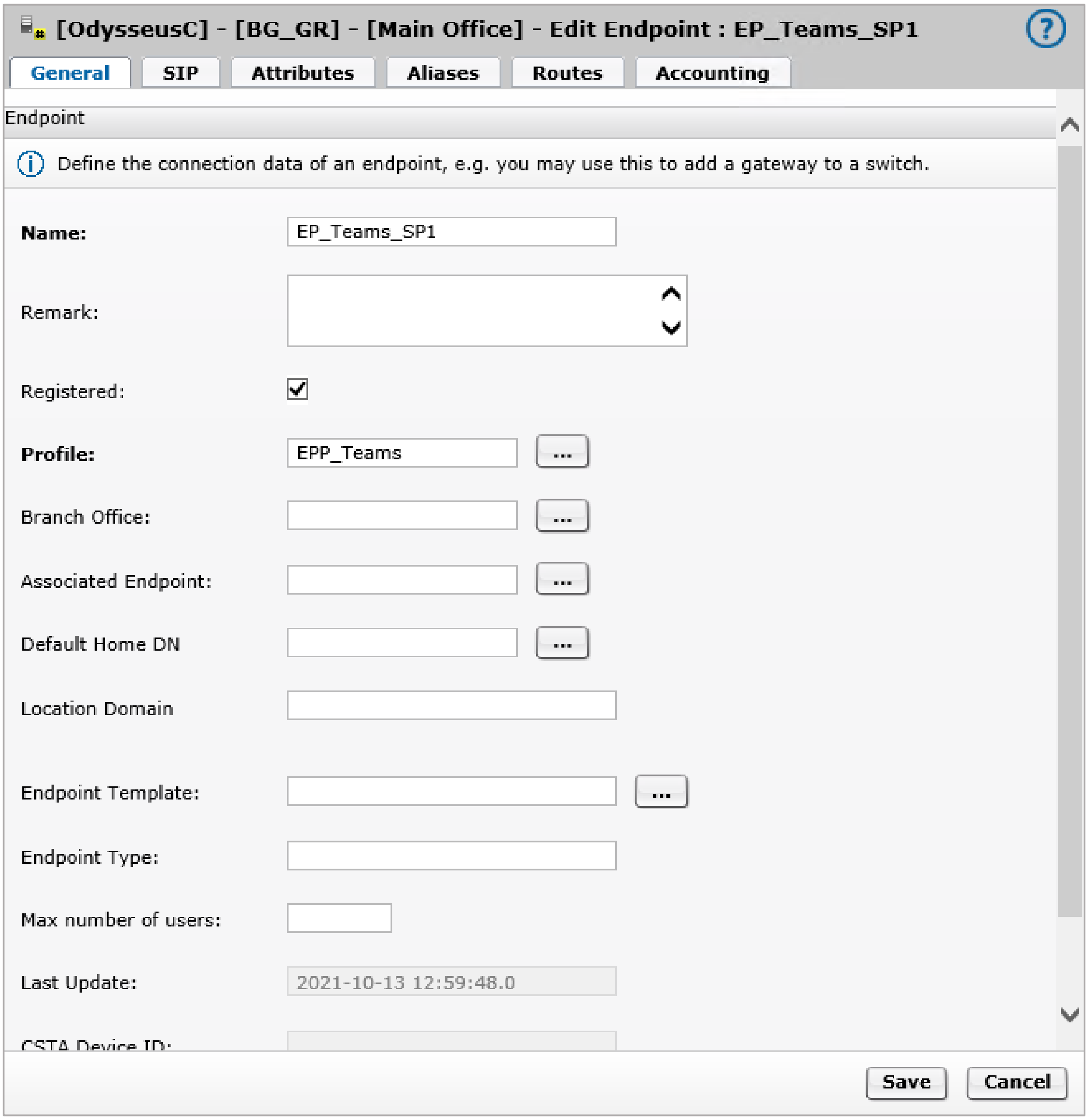
12. Click **OK** and then click **Save**.

2.2. Enabling the Registration flag

1. Go to OpenScape Common Management Platform > Configuration > OpenScape Voice > Business Group > Members > Endpoints window. A list of all the configured endpoints in OS Voice is displayed.

The registration flag for the newly created endpoints needs to be activated.

2. Select an endpoint and click **Edit.** As an example:



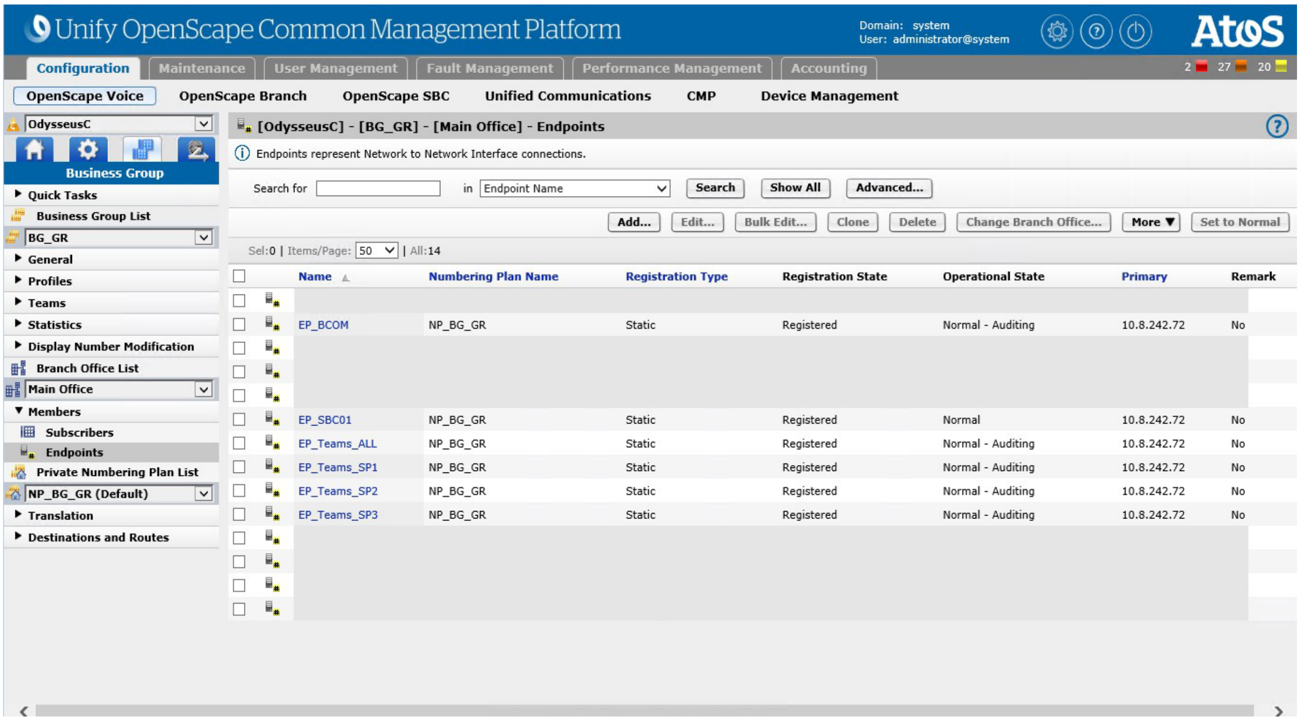
2. In the **General** tab, select the **Registered** checkbox.

   Note: The *Registered* checkbox can only be enabled if the SIP Type under the SIP tab is set to *Static*. This option specifies the registration state for the endpoint.

3. To activate the sending of **SIP OPTIONS** messages for the Teams and SSP endpoints, select the endpoint and click **More**.

4. Select the "Periodic Audit Enable" option from the drop-down menu.

The newly created endpoints in the CMP endpoint overview window are shown below as an example:



2.3 Configuring Destinations & Routes

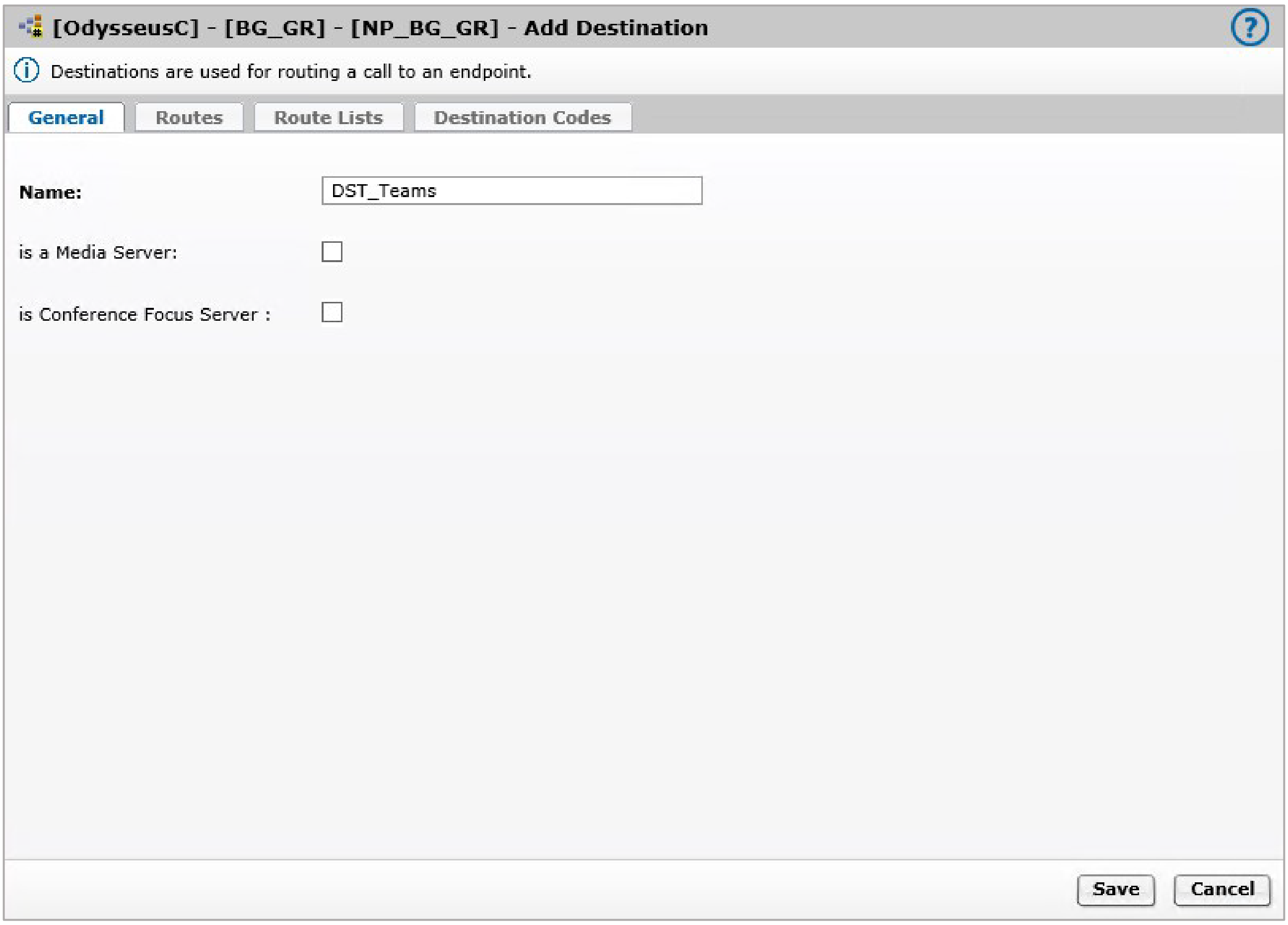
Destinations are logical targets for off-net or on-net routing. When a destination is created, the name of the destination is bound to the numbering plan where the destination is created. Destinations are used to route a call to an endpoint representing a gateway.

Each **Route** is a collection of groups or addresses that provide a path to a destination.

2.3.1 MS Teams Destination Configuration

1. Navigate to OpenScape Common Management Platform > Configuration > OpenScape Voice > Business Group > Destinations and Routes > Destinations.

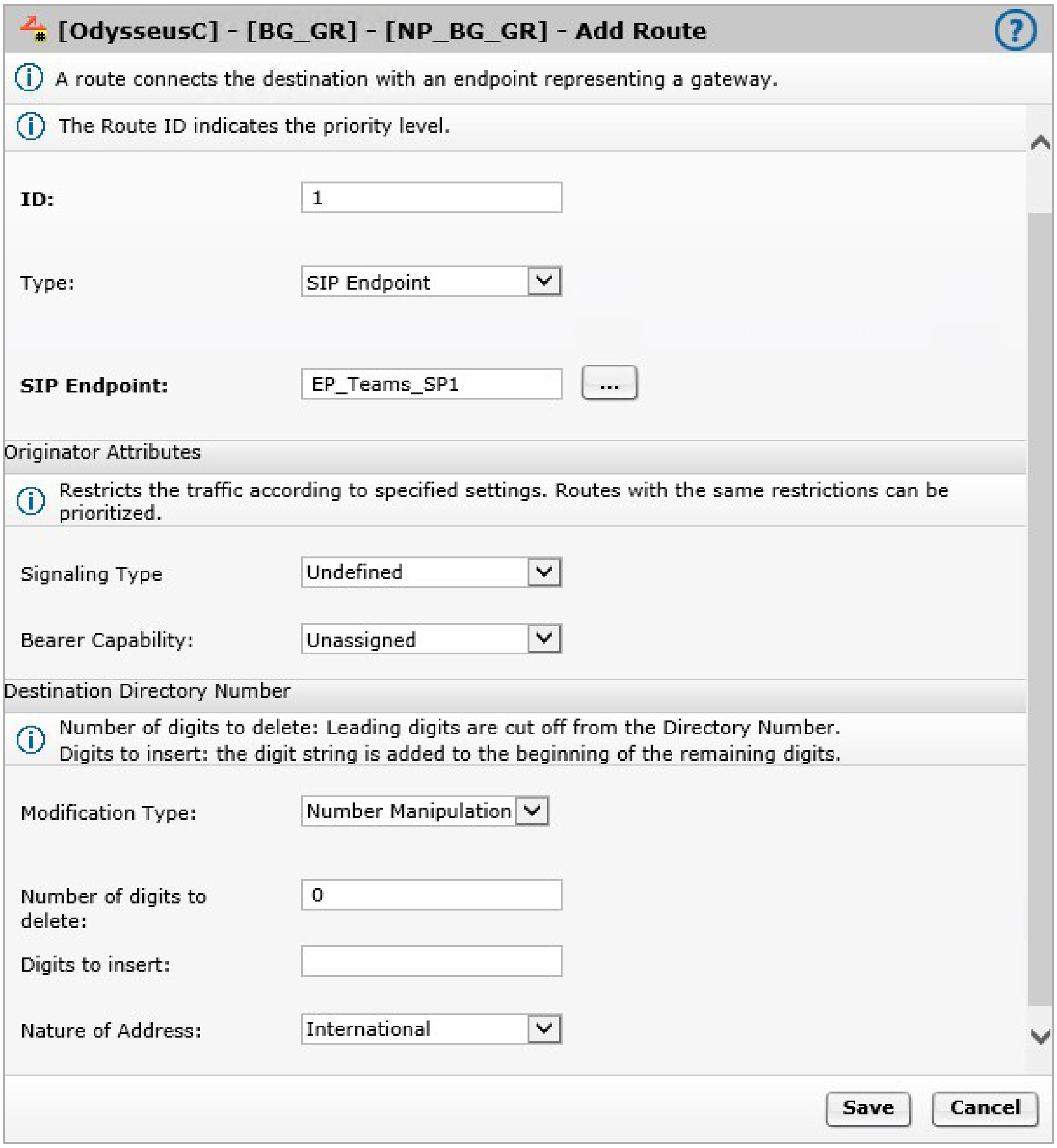
2. Click **Add.**



3. In the **General** tab enter the following:
   - **Name**: DST_Teams (a common-sense name).

4. Click **Save.**

5. Select the "DST_Teams" destination and click **Edit**.



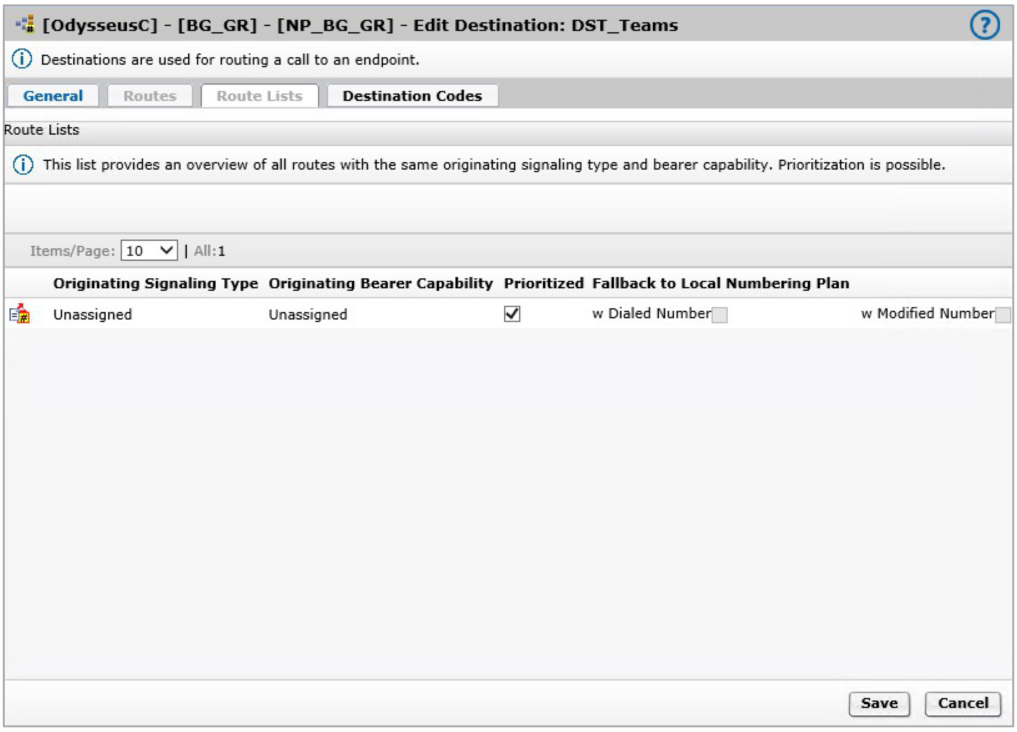
6. Configure the associated **"Route"**, by clicking in **"Routes"** tab and entering the following:

- **ID**: 1
- (the priority of the route; if there are multiple routes to a destination and route prioritization is selected, the route with the lowest numbered route ID has the highest priority, and will be selected first)
- **SIP Endpoint**: EP_Teams_SP1
- **Modification Type**: Number Manipulation
- **Nature of Address**: International

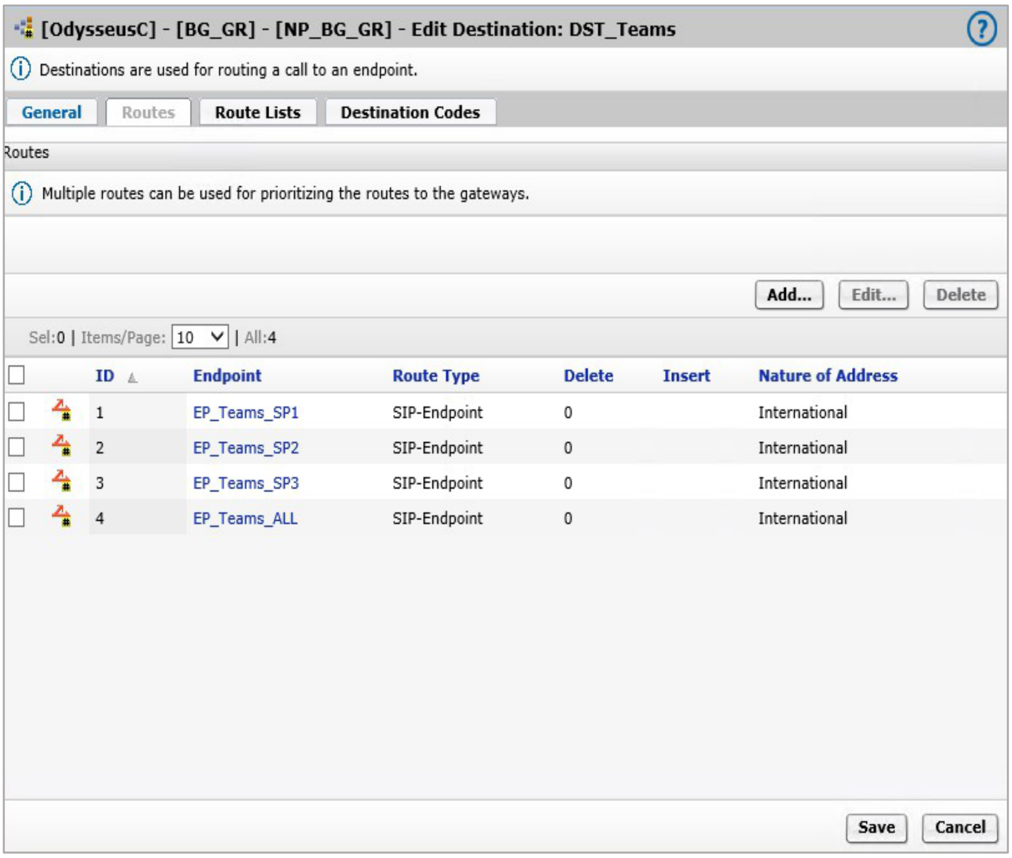
7. Repeat the same procedure for the remaining Teams endpoints, but different ID should be assigned per endpoint:
   - "EP_Teams_SP2 ➔ ID = 2"
   - "EP_Teams_SP3 ➔ ID = 3"
   - "EP_Teams_ALL ➔ ID = 4"

8. To populate the **"SIP Endpoint"** box with e.g., the "EP_Teams_SP1" endpoint:
   a. Click the corresponding button, then select **"Main Office"** in the pop-up window.
   b. Click **Next.**
   c. Select "EP_Teams_SP1" from the list.
   d. Click **OK**.

9. To enable the MS Teams route prioritization:
   a. Go to the **"Route Lists"** tab
   b. Select the **"Prioritized"** checkbox.
   c. Click **Save**.

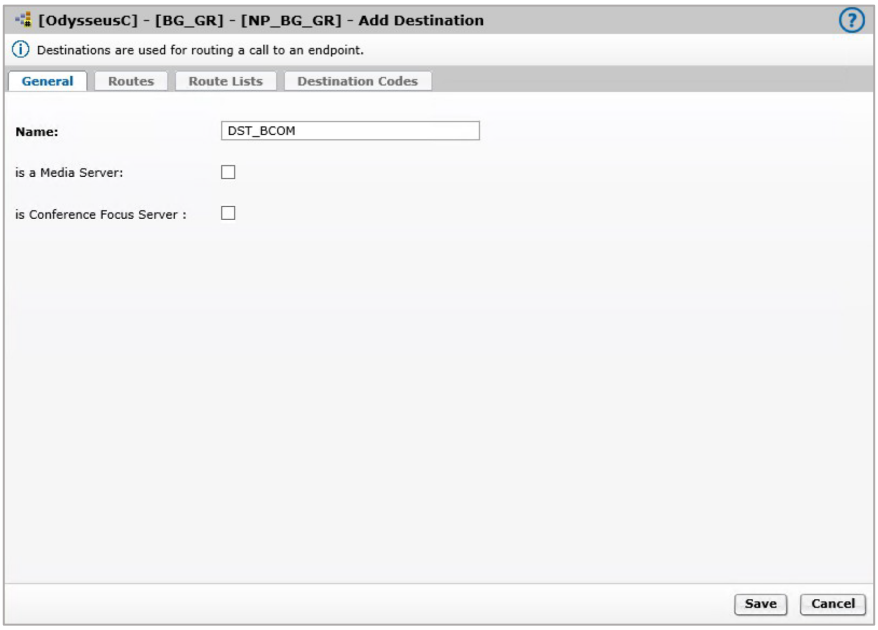


After saving the configuration, the **"Routes"** tab should appear as shown below:

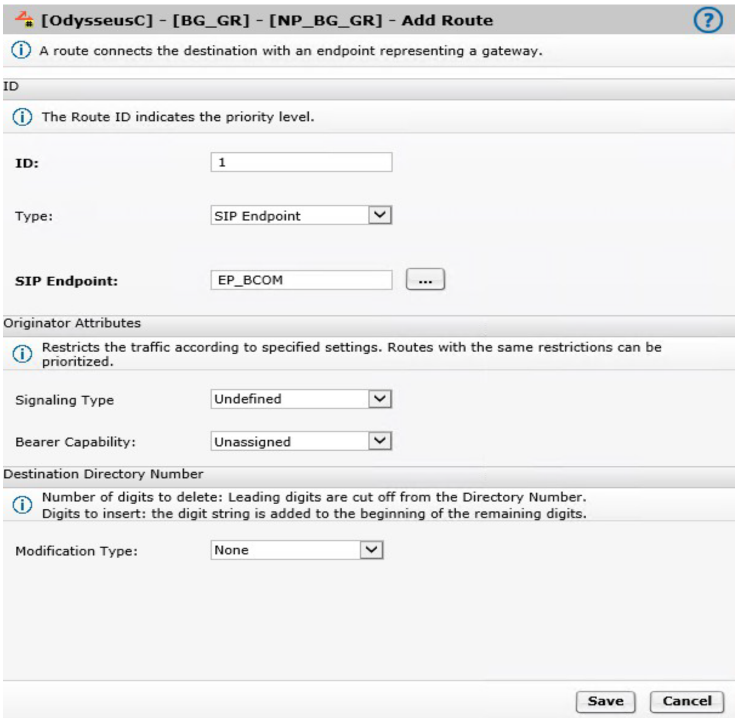


2.3.2 SSP Destination Configuration

1. Go to **OpenScape Common Management Platform** > **Configuration** > **OpenScape Voice** > **Business Group** > **Destinations and Routes** > **Destinations**.
2. Click **Add.**



3. In the **General** tab enter the following:
   - **Name**: DST_BCOM (a common-sense name).
4. Click **Save.**
5. Select the **"DST_BCOM"** destination and click **Edit**.

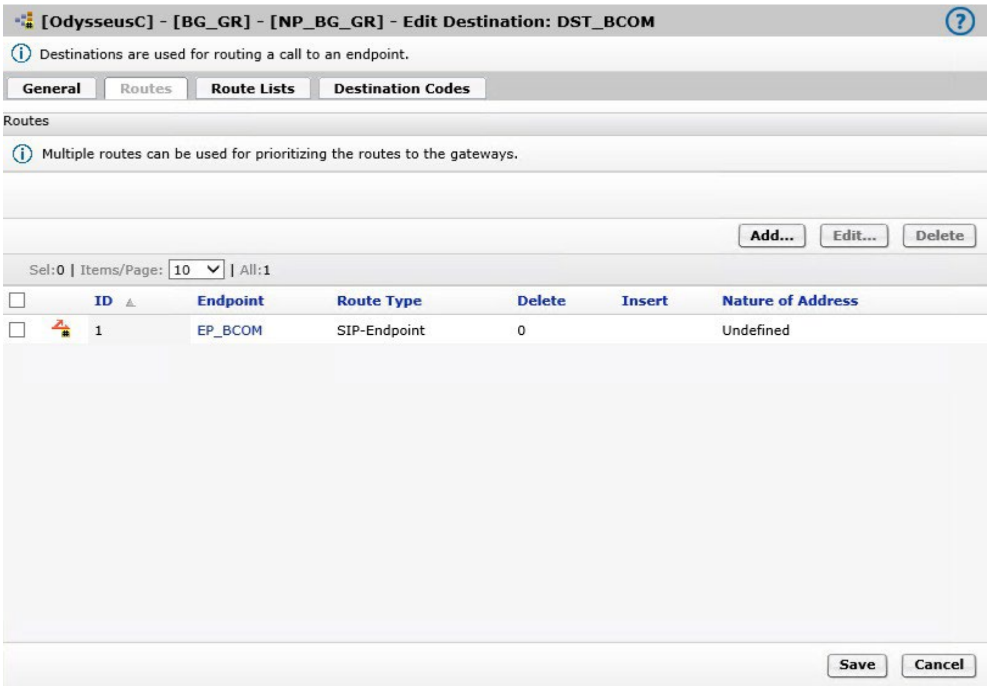


6. Configure the associated **Route**, by clicking in **Routes** tab and entering the following:

- **ID**: 1
- SIP Endpoint: EP_BCOM
- Modification Type: None

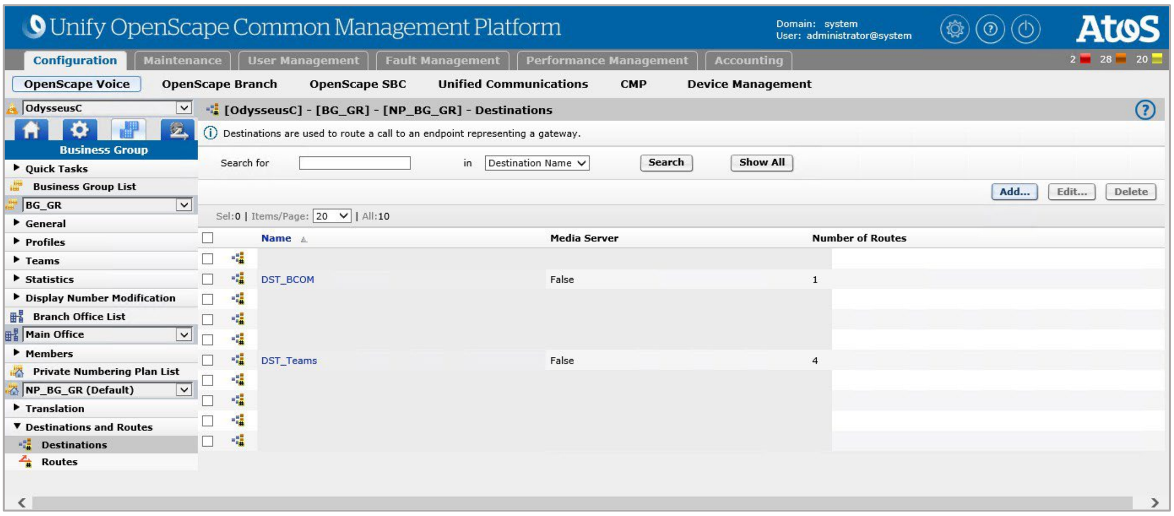
6. Click **Save**.

The **Route** tab should appear as shown below:



2.3.3 Viewing Destinations

Go to OpenScape Common Management Platform > Configuration > OpenScape Voice **> Business Group > Destinations and Routes > Destinations** window.
The MS Teams and SSP (BCOM) destinations are displayed:



2.4 Configuring Translation

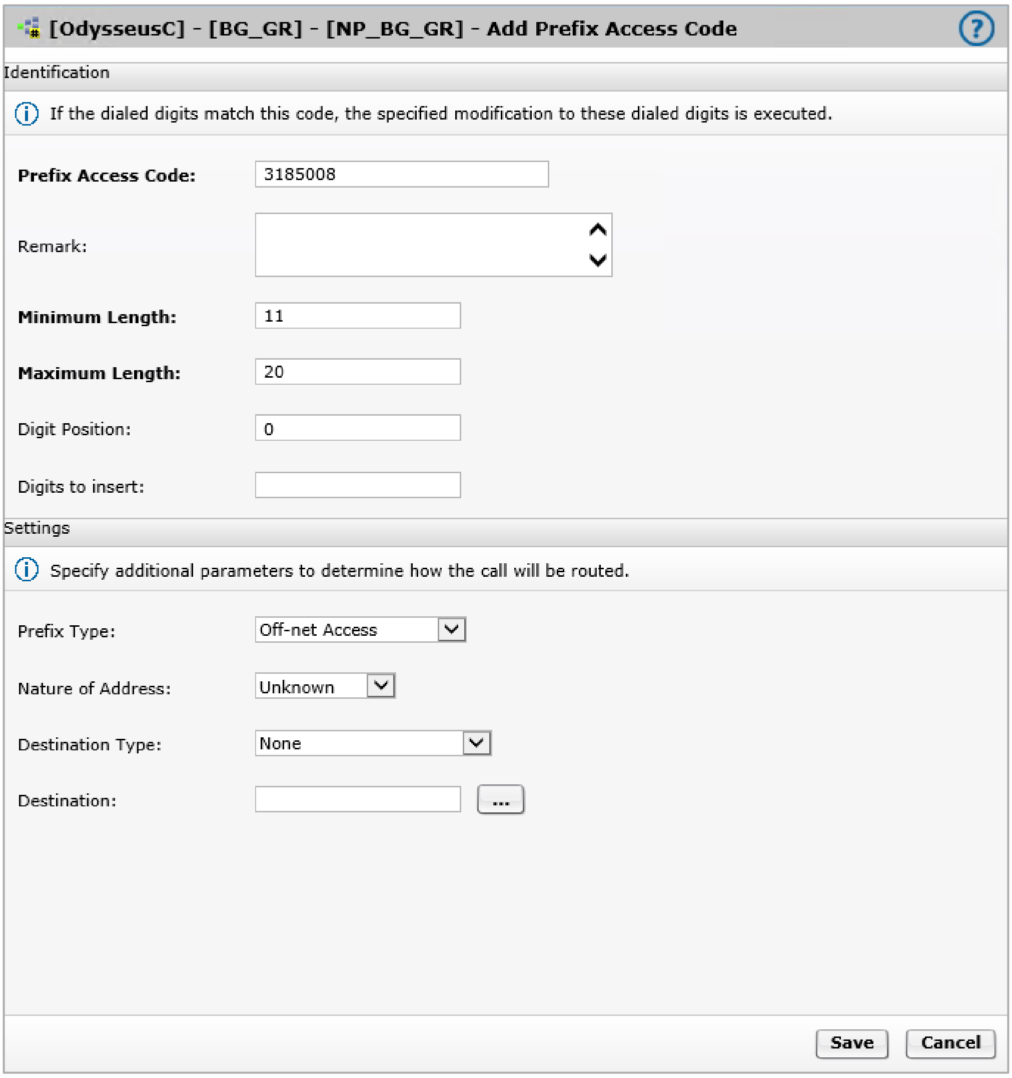
With **Translation,** the administrator configures where outgoing calls from OpenScape Voice subscribers are routed based on the dialed digits.

A call can only be routed if the dialed digits match a **PAC (Prefix Access Code)**.

The **Destination Code** feature provides destination codes for basic telephone service. A destination code is used for a call if the dialed (or PAC-modified) digits and the address match.

2.4.1 MS Teams Numbers Routing Configuration

1. Navigate to OpenScape Common Management Platform > Configuration > OpenScape Voice > Business Group > Translation > Prefix Access Codes.



2. Click **Add** and configure the following:

- **Prefix Access Code**: 3185008
(the starting digits of Teams subscriber numbers)
- **Minimum Length**: 11
(minimum expected length of Teams numbers)
- **Maximum Length**: 20
(maximum expected length of Teams numbers)
- **Digit Position**: 0

(don’t remove any digits from dialed number before sending them to destination)

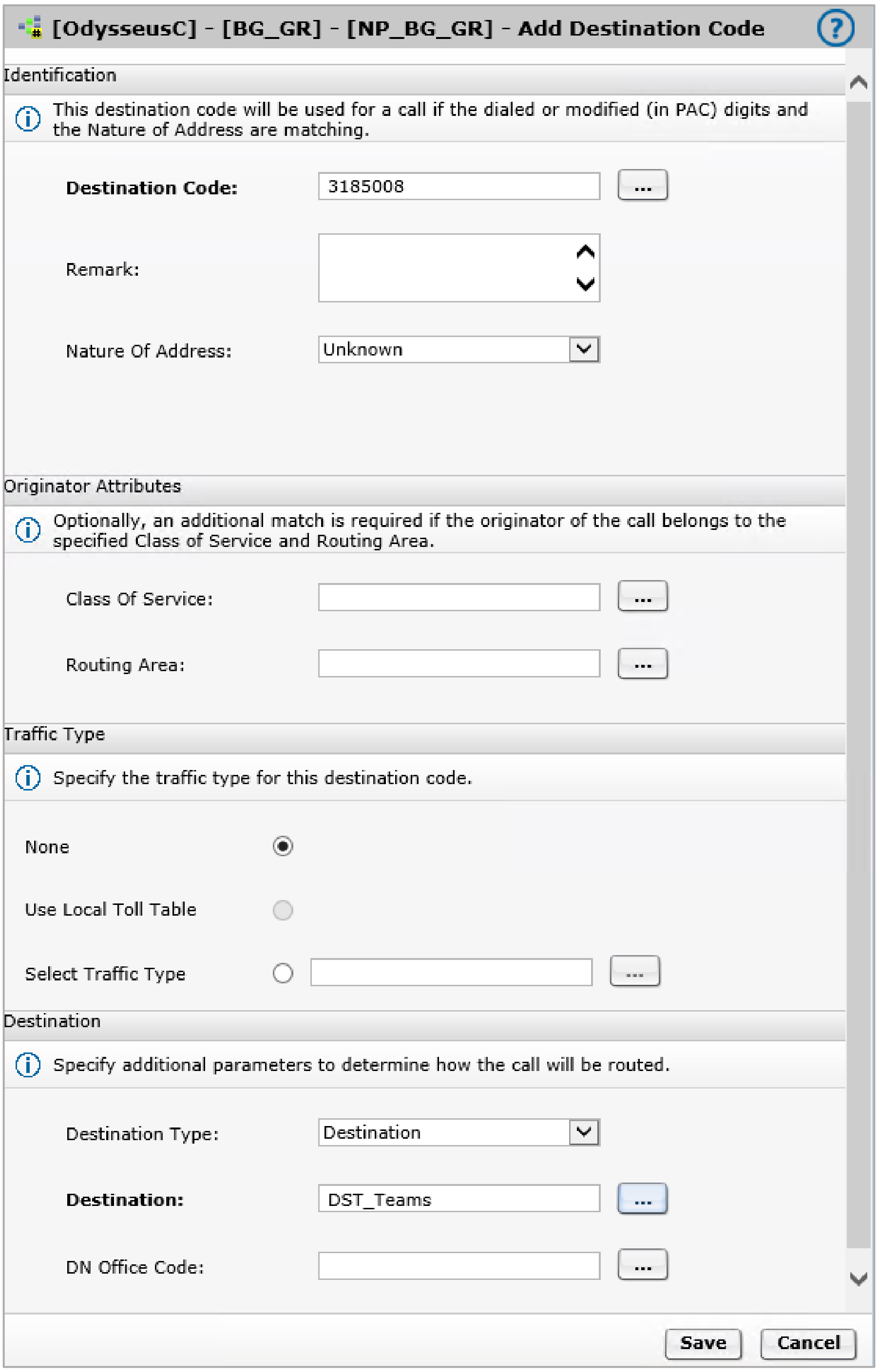
- **Prefix Type:** Off-net Access
  (a prefix access code to permit access to remote destinations)

  - **Nature of Address**: Unknown
  - **Destination Type**: None

The resulting digits will be processed in the user’s numbering plan’s destination codes table.

3. Click **Save**.

4. Go to **OpenScape Common Management Platform > Configuration > OpenScape Voice > Business Group > Translation > Destination Codes**.



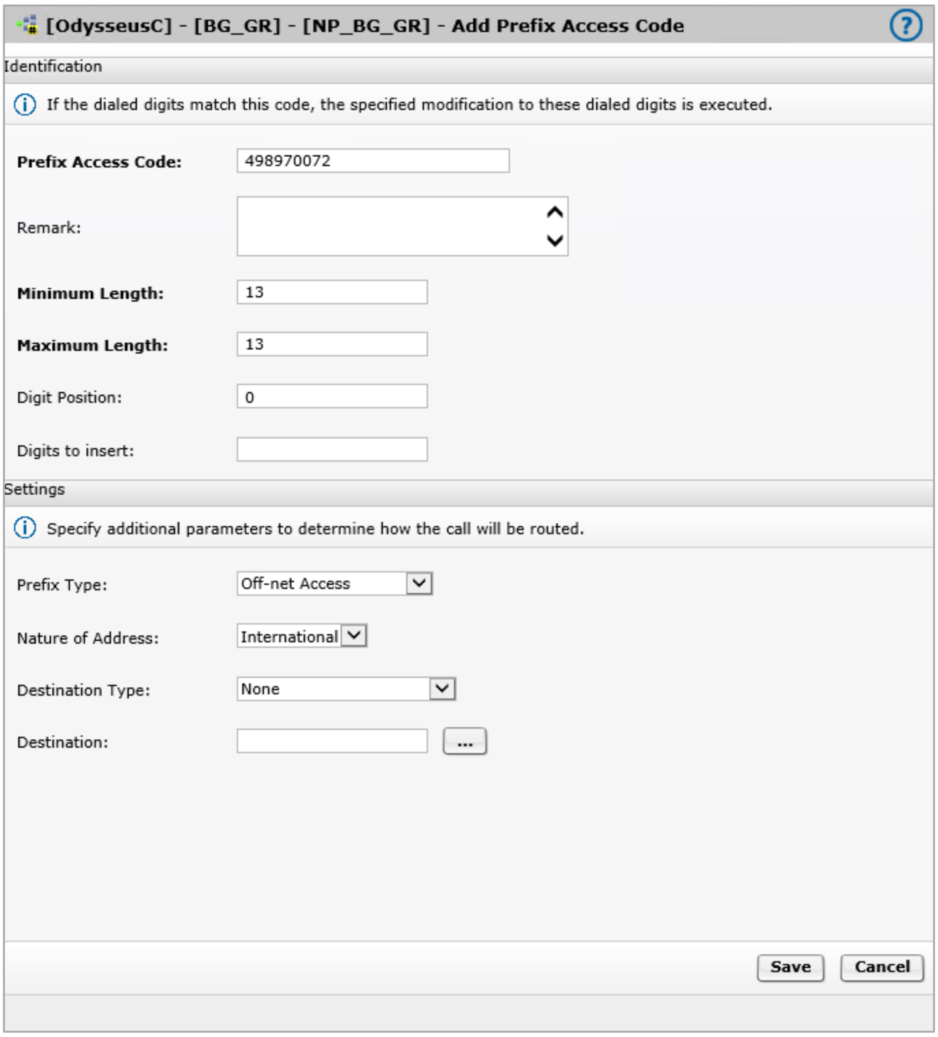
5. Click **Add** and enter the following:

- **Destination Code**: 3185008 (select the previously created PAC).
- **Nature of Address**: Unknown
- **Destination Type**: Destination
- **Destination**: DST_Teams

6. Click **Save**.

2.4.2 PSTN Numbers Routing Configuration

1. Go to **OpenScape Common Management Platform > Configuration > OpenScape Voice > Business Group > Translation > Prefix Access Codes**.



2. Click **Add.**

   The **Add Prefix Access Code** window opens.

3. Configure the following settings:

   - **Prefix Access Code**: 498970072

     (the starting digits of PSTN subscriber numbers).
   - **Minimum Length**: 13 (minimum expected length of PSTN numbers).
   - **Maximum Length:** 13 (maximum expected length of PSTN numbers).

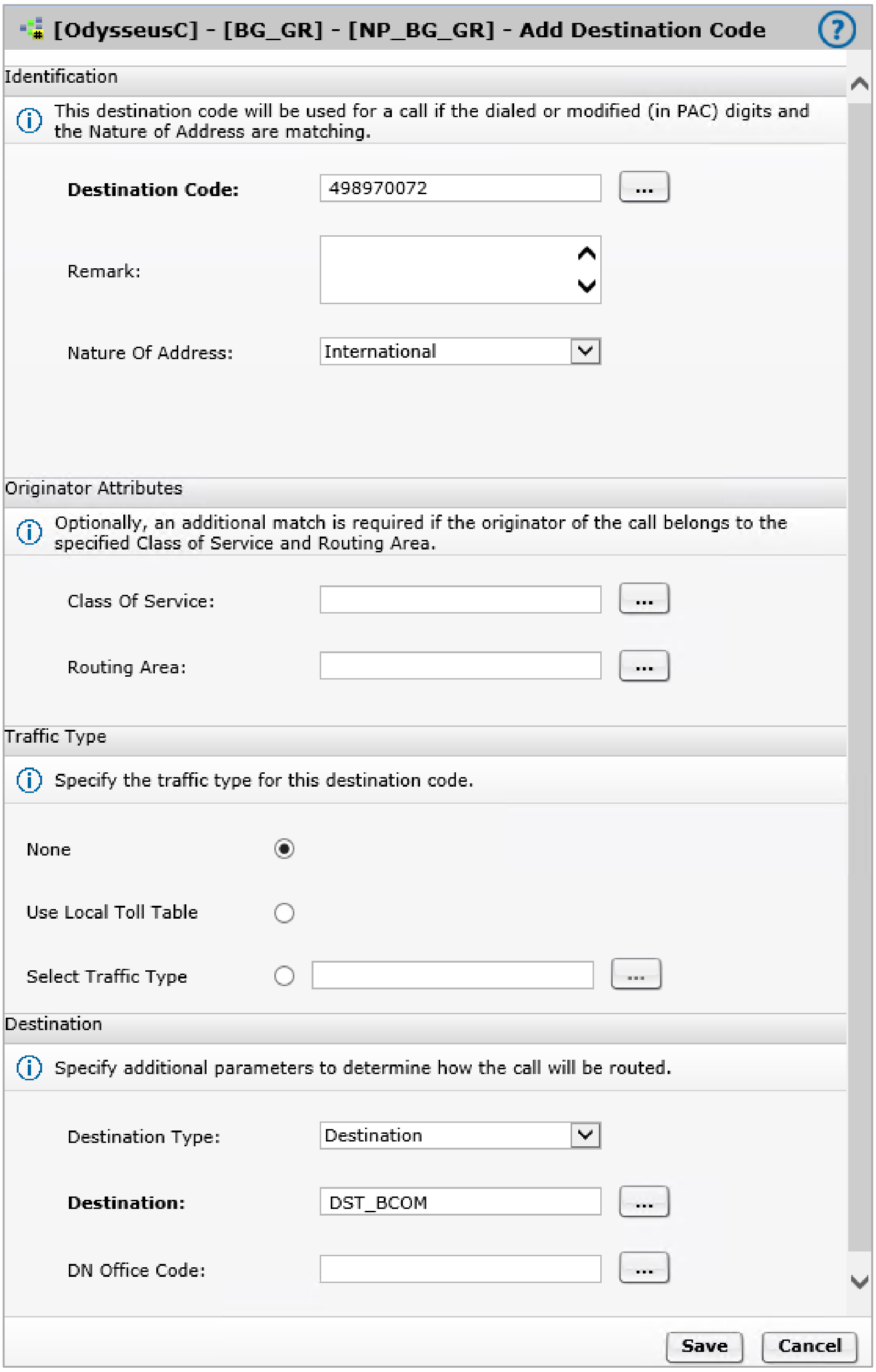
- **Digit Position:** 0

(don’t remove any digits from dialed number before sending them to destination).

- **Prefix Type:** Off-net Access
  (a prefix access code to permit access to remote destinations).
- **Nature of Address**: International.
- **Destination Type:** None
  (the resulting digits will be processed in the user’s numbering plan’s destination codes table).

4. Click **Save**.

5. Go to **OpenScape Common Management Platform > Configuration > OpenScape Voice > Business Group > Translation > Destination Codes**.



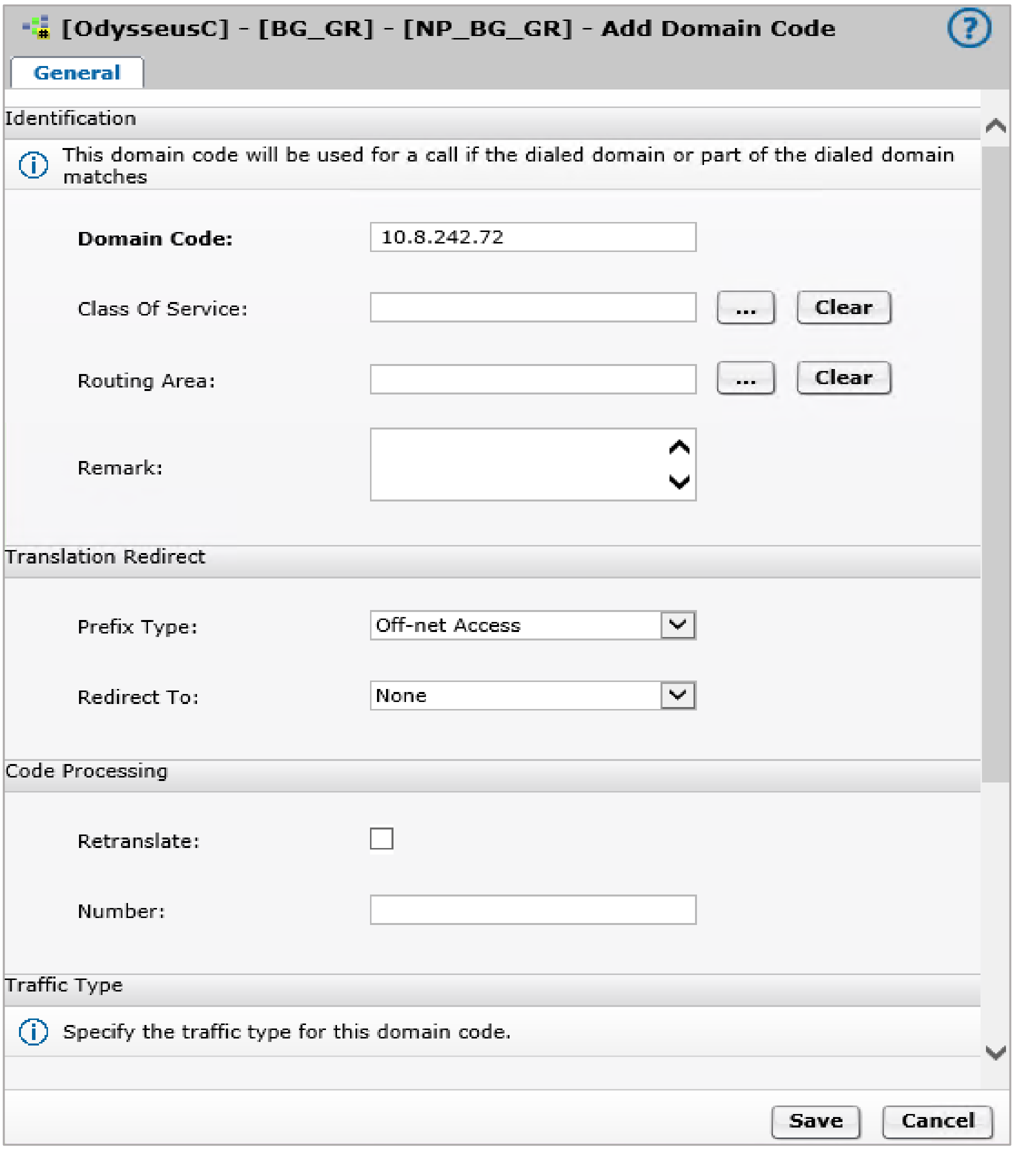
6. Click **Add** and enter the following:

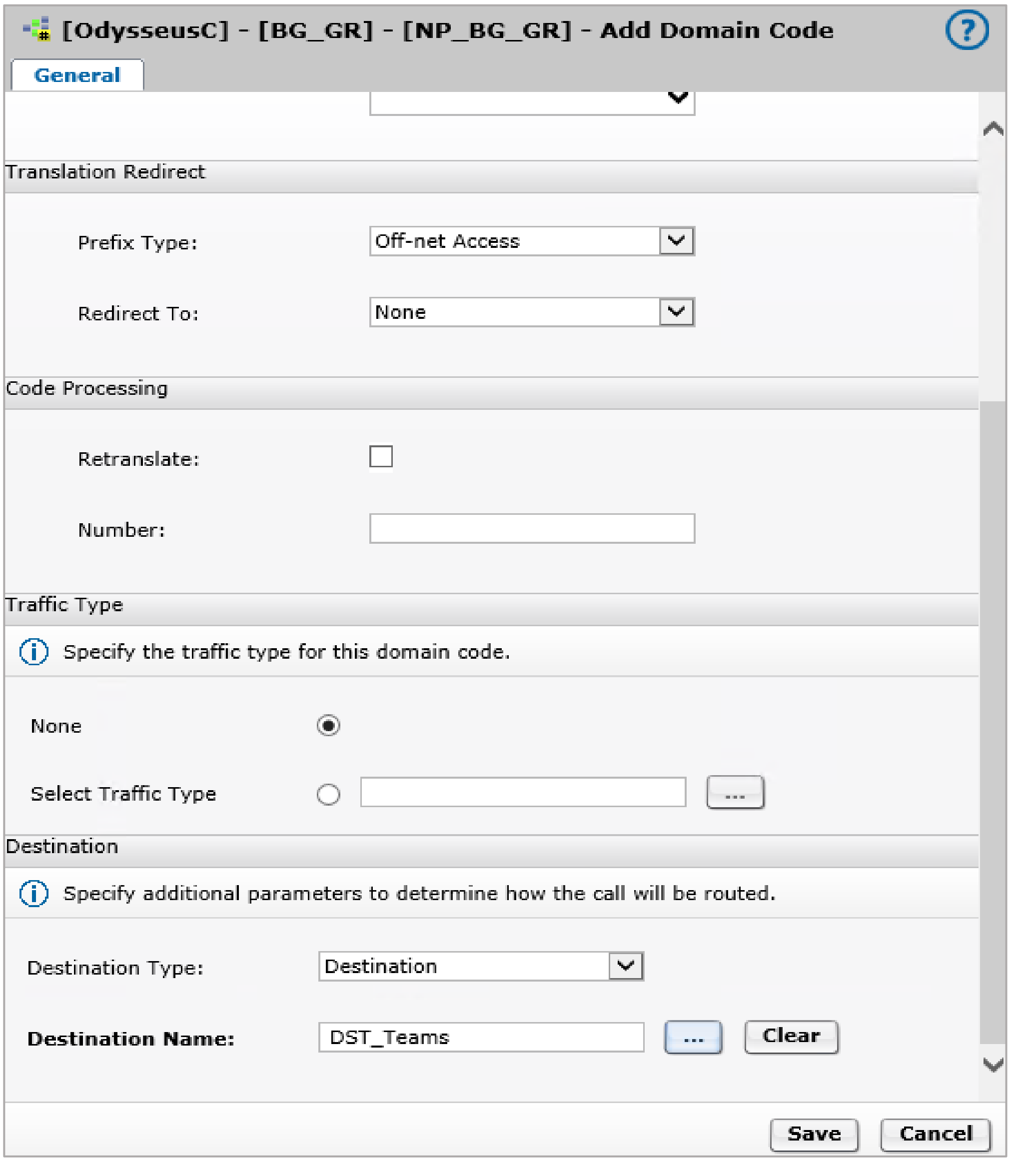
- **Destination Code**: 498970072 (select the previously created PAC).
- Nature of Address: International
- Destination Type: Destination
- Destination: DST_BCOM

6. Click **Save**.

2.4.3 Domain Codes Configuration

1. Go to **OpenScape Common Management Platform > Configuration > OpenScape Voice > Business Group > Translation > Domain Codes**.





2. Click **Add** and configure the following:

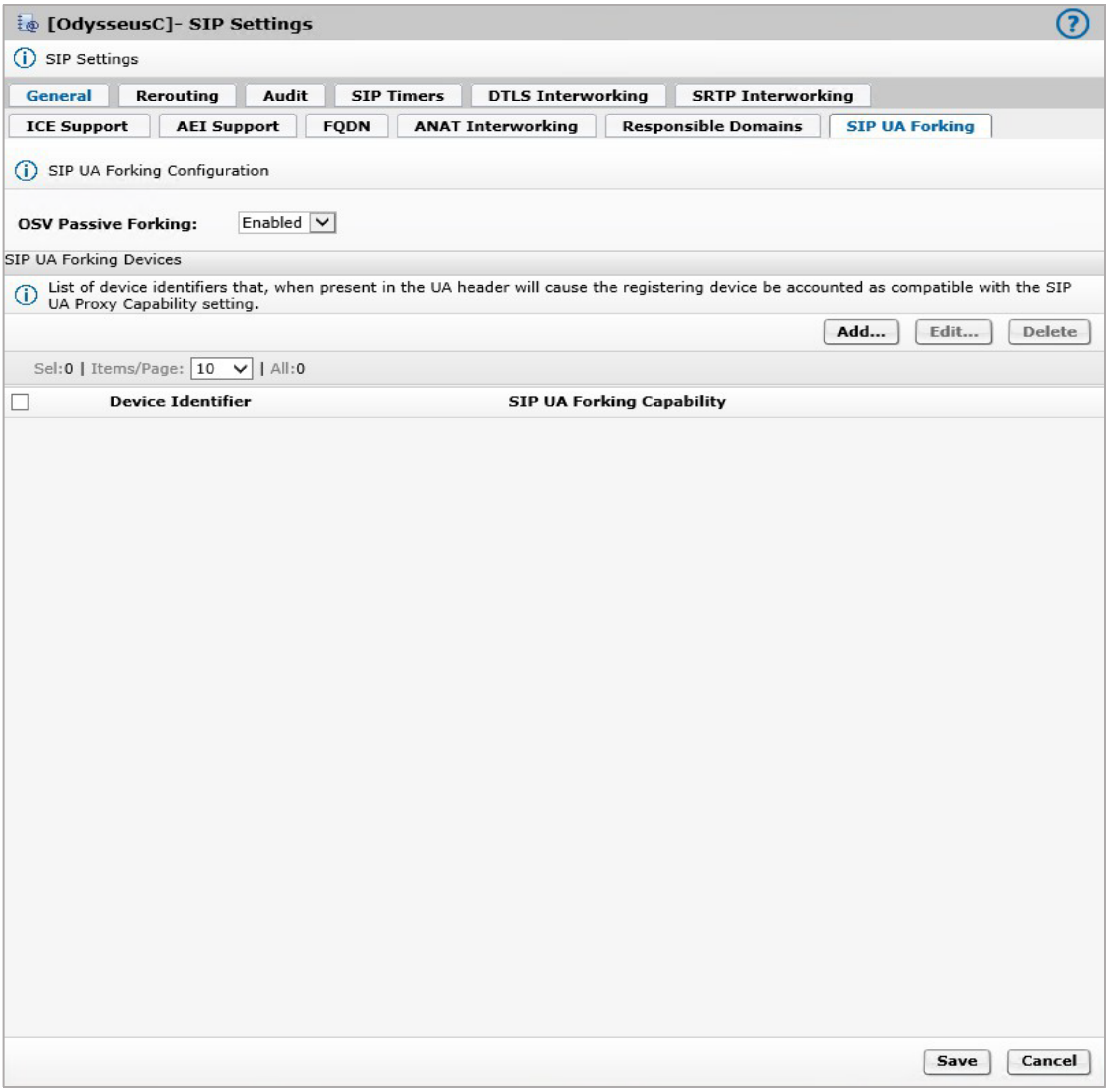
   - **Domain Code**: 10.8.242.72 (SBC core IP)
   - **Prefix Type**: Off-net Access
   - **Redirect To**: None
   - **Destination Name**: DST_Teams

3. Click **Save**.

2.5 Activating SIP UA Forking

OSV Passive Forking (UAC) provides an interworking function which essentially merges multiple Microsoft Teams downstream early dialogs into a single upstream SIP dialog. This functionality shields upstream SIP clients (SIP UAC) establishing sessions with the MS Teams network from being exposed to the full RFC 3261/RFC3264 forking SIP Proxy server behavior of the MS Teams Phone System. The SIP UA Forking tab is used to enable the feature and list all devices configured with their respective SIP forking capability.

1. To activate the OSV Passive Forking feature, go to **OpenScape Common Management Platform > Configuration > OSV > Administration > Signaling Management > SIP**.



2. From the **OSV Passive Forking** drop-down menu, select **Enabled**.

3. Click **Save**.

2.6 Configuring Display Number Modification (optional)

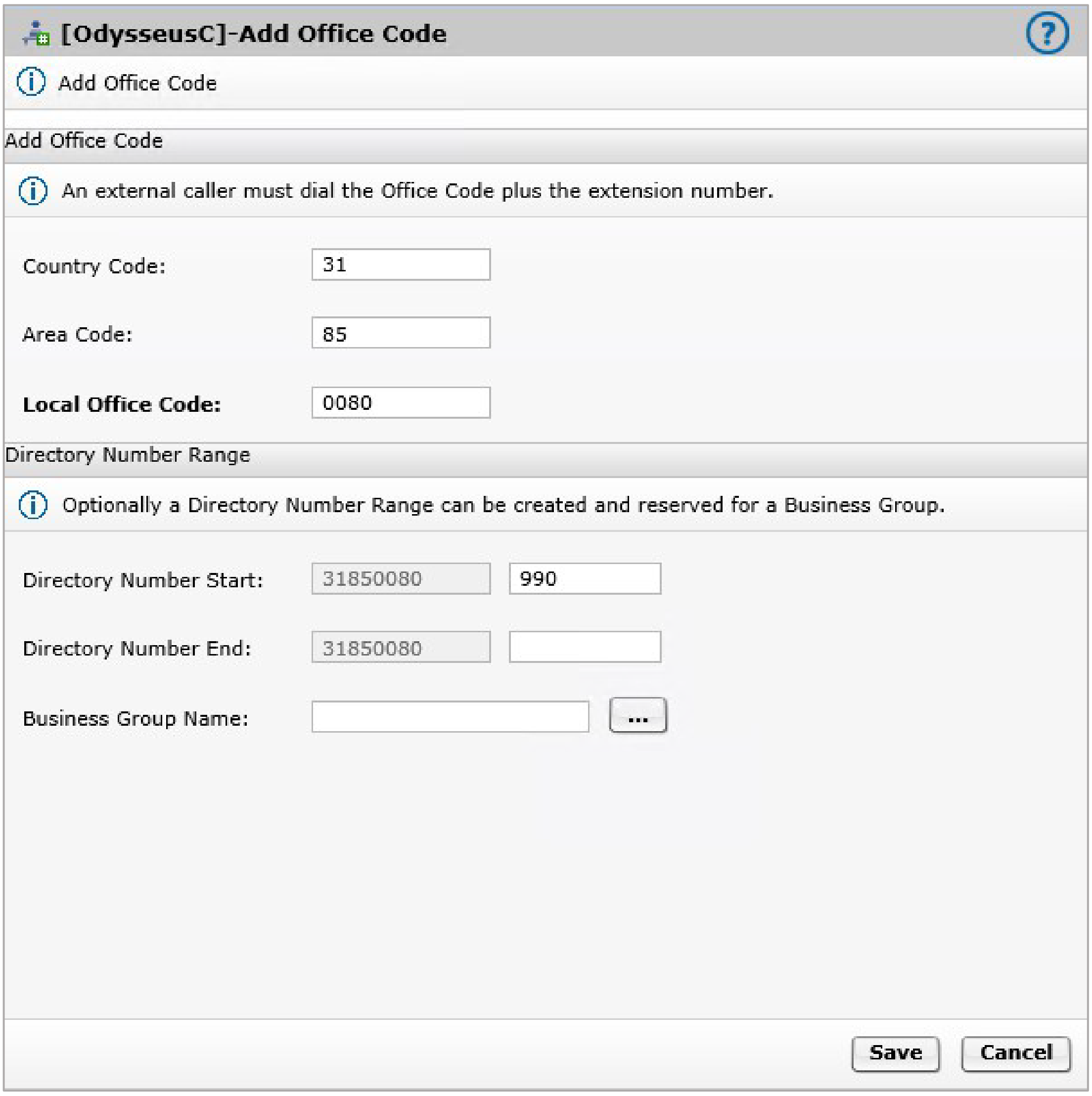
If there is a requirement to manipulate the **FROM** header of an INVITE message (e.g., from the SBC to SSP), the header manipulation is performed in the OS Voice **Display Number Modification** configuration area.

In the following example, the FROM header will be manipulated to include the number in **national format** instead of the international format (E.164).

1. To add the Dutch office code to OS Voice, go to **OpenScape Common Management Platform > Configuration > OSV > Global Translation and Routing > Directory Numbers > Office Codes**.

2. Click **Add**.

   The **Add Office Code** window opens.

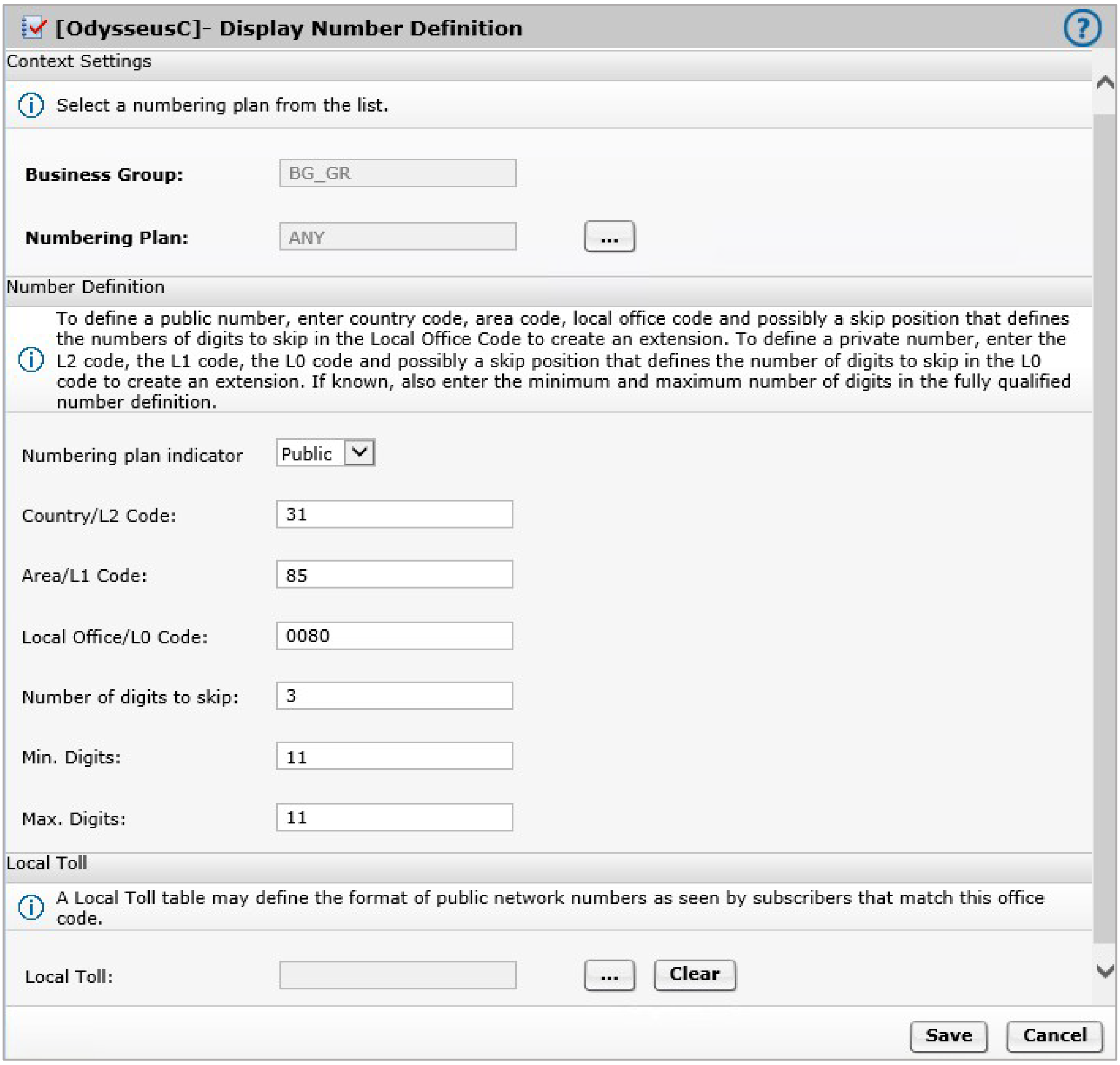


3. Configure the following:

   - Country Code: 31
   - Area Code: 85
   - Local Office Code: 0080
   - Directory Number Start: 990 (starting extension).

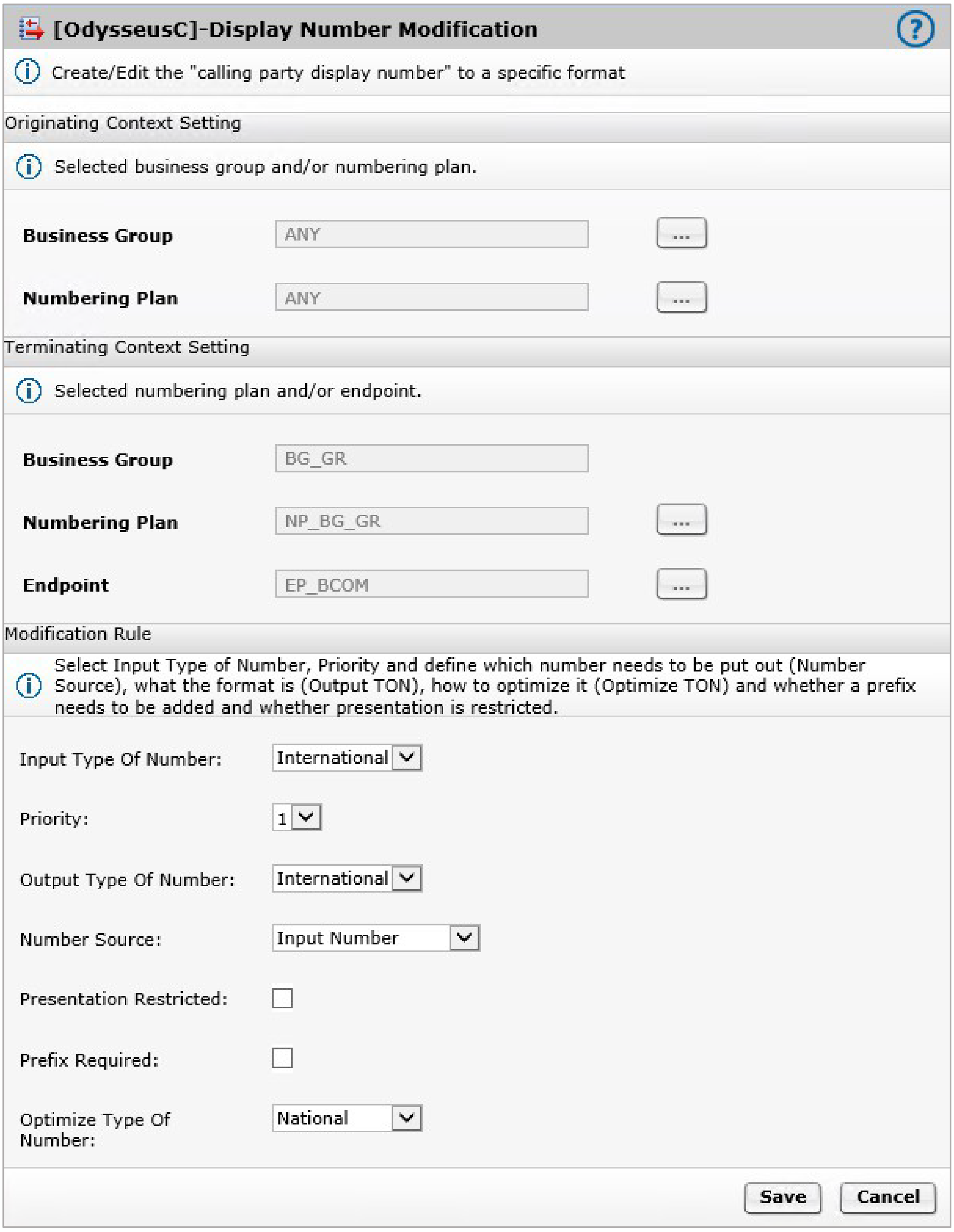
4. Click **Save**.

4. Go to **OpenScape Common Management Platform > Configuration > OSV > BG > Display Number Modification > Definitions**.
5. Click **Add**.



6. Add the Dutch office code by entering the following:
   - Numbering plan indicator: Public
   - Country/L2 Code: 32
   - Area/L1 Code: 85
   - Local Office/L0 Code: 0080
   - Number of digits to skip: 3
     (indicates the numbers of digits to skip in the Local Office Code (NPI = Public) or L0 Code (NPI = Private) to create an extension).
   - Min. Digits: 11
   - Max. Digits: 11
7. Click **Save**.

8. Go to **OpenScape Common Management Platform > Configuration > OSV > BG > Display Number Modification > Modifications**.
9. Click **Add**.

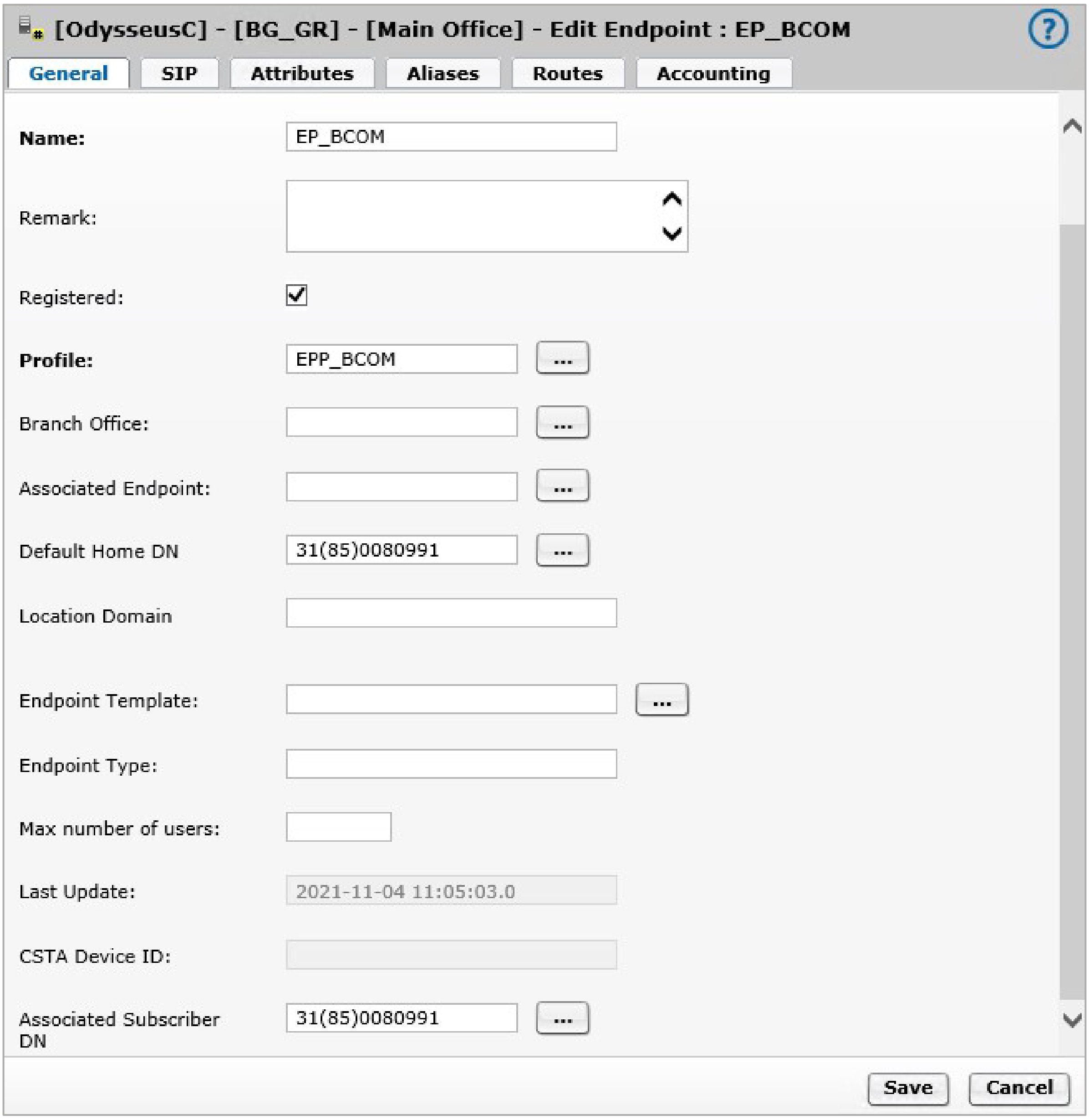


10. Add the Dutch office code by configuring the following:

- **Endpoint**: EP_BCOM
- Input Type Of Number: International
- **Priority:** 1 (highest priority)
- Output Type Of Number: International
- **Number Source:** Input Number
  (defines the input format of the "presenter number" when it comes into the OpenScape Voice)

- **Optimize Type Of number:** National
  (defines the conversion by call processing of an explicit *"calling number"* to a desired format before the number is transmitted to the destination)

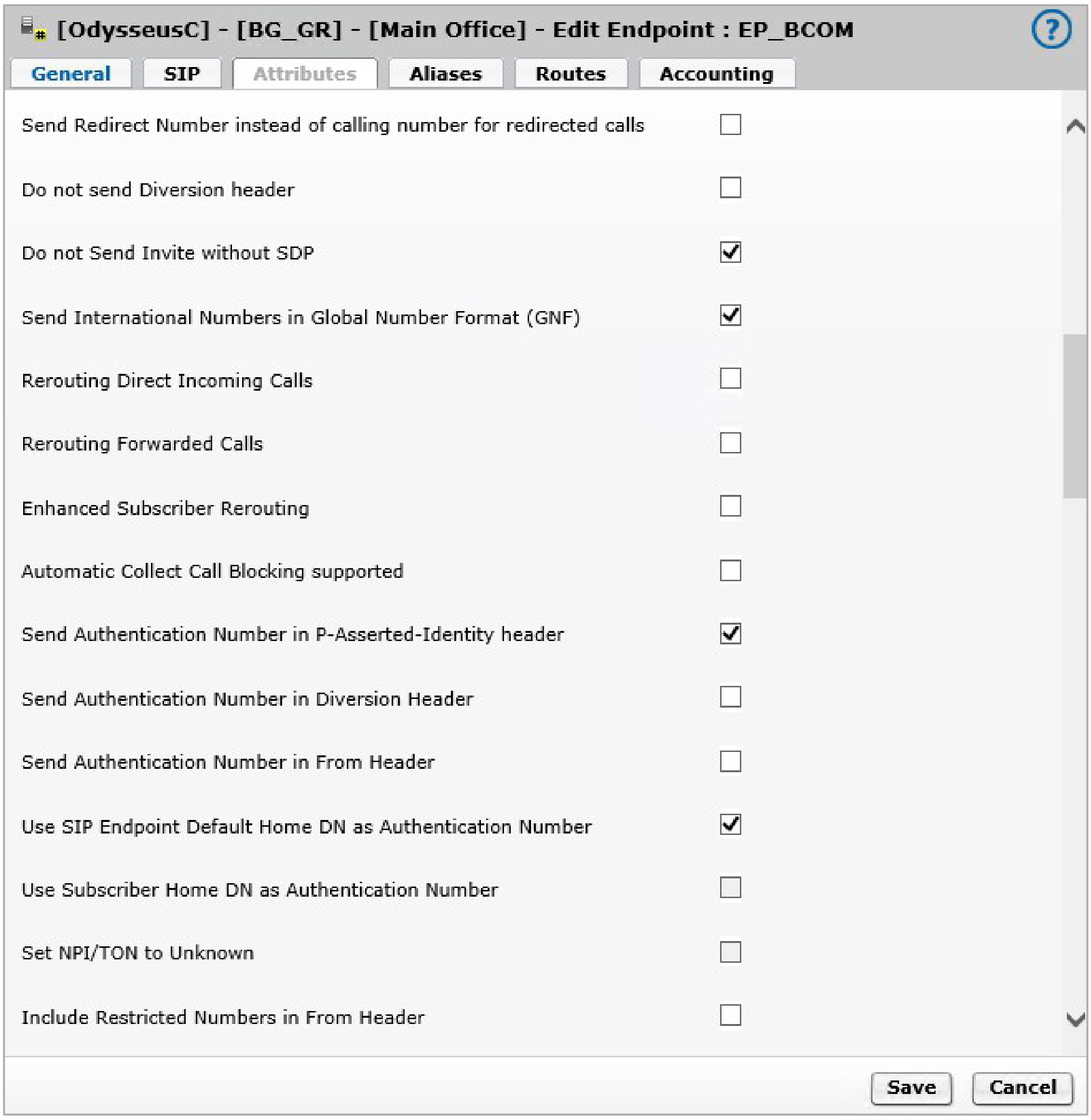
11. Click **Save**.
12. Go to **OpenScape Common Management Platform > Configuration > OSV > BG > Members > Endpoints** and select the "EP_BCOM" endpoint.
13. Click **Edit**.



14. Set the **"Default Home DN"** (set the location of the endpoint in Dutch) to "31850080990" (provided by the SSP).

Note: With current OS Voice implementation, the PAI header modification is not possible from the **Display Number Modification** configuration.

15. If the international number format is not desired in PAI header for calls from Teams to PSTN, you can configure the following in the **Attributes** tab:



When **"Send Authentication Number in P-Asserted-Identity header"** is "Activated", the PAI header contains the calling party's name and the dialable number.

When enabled, the display rules that are usually used for populating the PAI header are overridden, apart from the **Display Number Modification** rules.

When **"Use SIP Endpoint Default Home DN as Authentication Number"** is "Activated", the Default Home DN provisioned for the SIP endpoint is used to populate the authenticated number.

3 Configuring OpenScape SBC

This chapter outlines the configuration of OpenScape SBC for interworking with Teams Direct Routing.

The OpenScape SBC will be configured with the connection to OS Voice, SSP (BCOM) and Teams Phone System (remote) endpoints.

As an example:

Items	Example
SBC Core (LAN) IP	10.8.242.72
SBC Access (WAN) IP	195.97.14.76
SBC Public FQDN	sbc01.athdrlabs.xyz
OS Voice node 1 (SIP Signaling) IP	10.8.242.16 TCP 5060
OS Voice node 2 (SIP Signaling) IP	10.8.242.26 TCP 5060
Teams FQDN 1 SIP trunk	sip.pstnhub.microsoft.com TLS 5061 (LAN port for OS Voice trunk 50001)
Teams FQDN 2 SIP trunk	sip2.pstnhub.microsoft.com TLS 5061 (LAN port for OS Voice trunk 50002)
Teams FQDN 3 SIP trunk	sip3.pstnhub.microsoft.com TLS 5061 (LAN port for OS Voice trunk 50003)
Teams FQDN ALL SIP trunk	sip-all.pstnhub.microsoft.com **see note** TLS 5061 (LAN port for OS Voice trunk 50004)
SSP (BCOM) SIP trunk	Remote URL: sip.bcom.nl Default Home DN: 31850080990 (LAN port for OS Voice trunk 50010)

Routine or not MS Teams Direct Routing specific OS SBC configuration will be omitted. OpenScape SBC installation and administration documentation can be found in the [Unify customer documentation site](#).

Important:
Per Microsoft's announcement, support for the *"sip-all.pstnhub.microsoft.com"* FQDN will end in March 2022.

Although Microsoft recommends using the three FQDNs for Direct Routing connection points — **"sip.pstnhub.microsoft.com"**, **"sip2.pstnhub.microsoft.com"**, and **"sip3.pstnhub.microsoft.com"** — the **"sip-all.pstnhub.microsoft.com"** FQDN was originally used in Unify component configurations due to DNS resolution issues in some countries.
However, there have been reported cases where the **"sip-all.pstnhub.microsoft.com"** FQDN can cause incorrect certificate negotiation between OpenScape SBC and the Microsoft Teams tenant.

Therefore, do NOT configure the SIP trunk to point to "sip-all.pstnhub.microsoft.com" in Unify components unless explicitly recommended by Unify support.

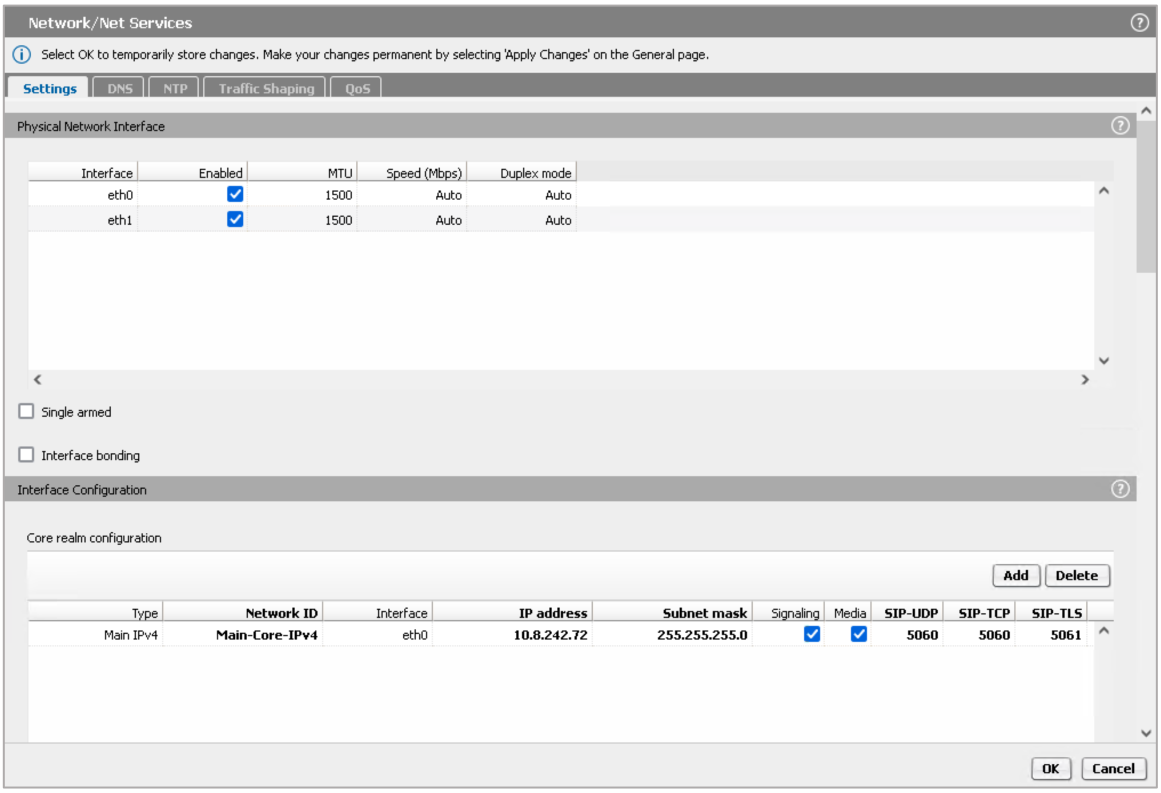
For completeness, this document still presents the configuration of the **"sip-all.pstnhub.microsoft.com"** SIP trunk.

3.3 Connecting to OpenScape Voice Server

3.3.1 Core Realm Interface Configuration

Use the TCP port number from subsection 2.1.1 for the connection of OS SBC's eth0 (core) interface to OS Voice.

1. Go to **OS SBC Management Portal > Network/Net Services > Settings**.



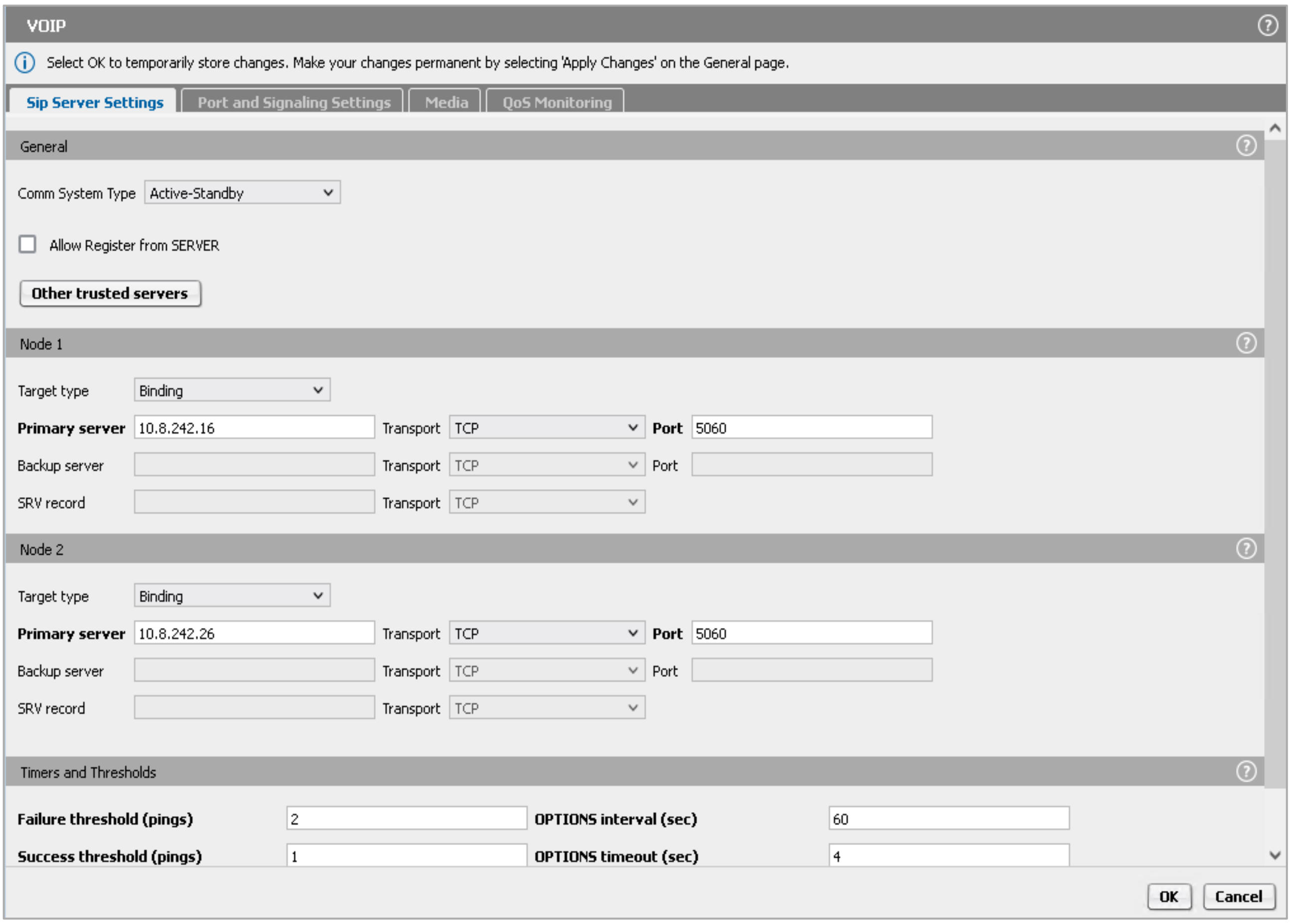
2. In the **Core realm configuration** area, ensure that for the **eth0** interface, the **"SIP-TCP"** value is "5060".

3. Click **OK**.

3. Click **Apply Changes** on OS SBC main page.

3.3.2 SIP Server Configuration

The SIP connectivity to OS Voice is configured in **OS SBC Management Portal** > **VOIP** window.



1. In the **Sip Server Settings** tab, enter the following:
   - **Comm System Type**: Active-Standby

     (OS Voice is a Duplex system i.e., two nodes in active-standby mode)
   - **Target Type**: Binding (for both **OS Voice Node1** and **Node2**)
   - **Primary Server**: 10.8.242.16
     (OSV SIPSM address over TCP for **OS Voice Node1**)
   - **Transport**: TCP (for both OS Voice Node 1 and Node 2)
   - **Port**: 5060 (listening port for both **OS Voice Node 1** and **Node 2**).
2. Click **OK**.
3. Click **Apply Changes** on OS SBC main page.

Note: The OS Voice SIP Signaling Manager addresses for UDP/TCP/TLS can be found in OS Voice node's **node.cfg** file located in the **/etc/hiq8000** folder (parameters: "sipsm1_vip" for **OS Voice Node1** and "sipsm2_vip" for **OS Voice Node 2**).

Alternatively, the OS Voice SIPSM IP addresses can be retrieved from CMP.

3.4 Configuring Certificates

Microsoft Phone System Direct Routing interface allows only TLS connections for SIP traffic from SBCs with a certificate signed by one of Microsoft's trusted Certification Authorities.

The certificate needs to have the SBC FQDN as the common name (CN) in the subject field. Certificates with a wildcard in the certificate Subject Alternate Name field conforming to RFC2818 are also supported.

For more information about the certificate and current Microsoft supported Certification Authorities, refer to Microsoft site:

https://docs.microsoft.com/en-us/microsoftteams/direct-routing-plan#public-trusted-certificate-for-the-sbc.

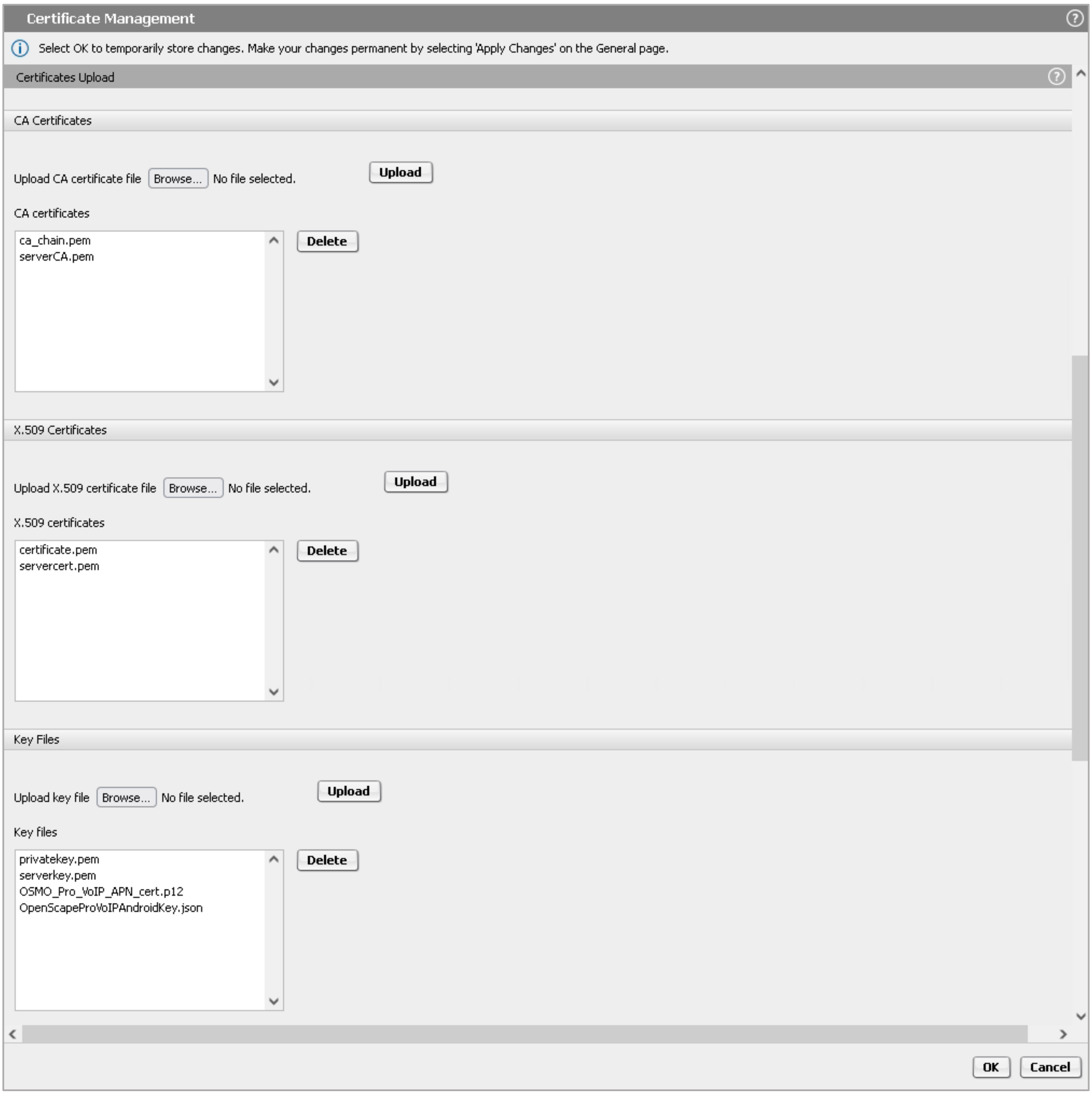
For the OpenScape SBC TLS interconnection to Microsoft Phone System, 3 files in *"pem"* format are required from the Certification Authority:

- CA certificate (e.g., "ca_chain.pem")
- Server certificate for OS SBC (e.g., "certificate.pem")
- OS SBC server certificate private key used for the CSR to CA (e.g., "privatekey.pem")

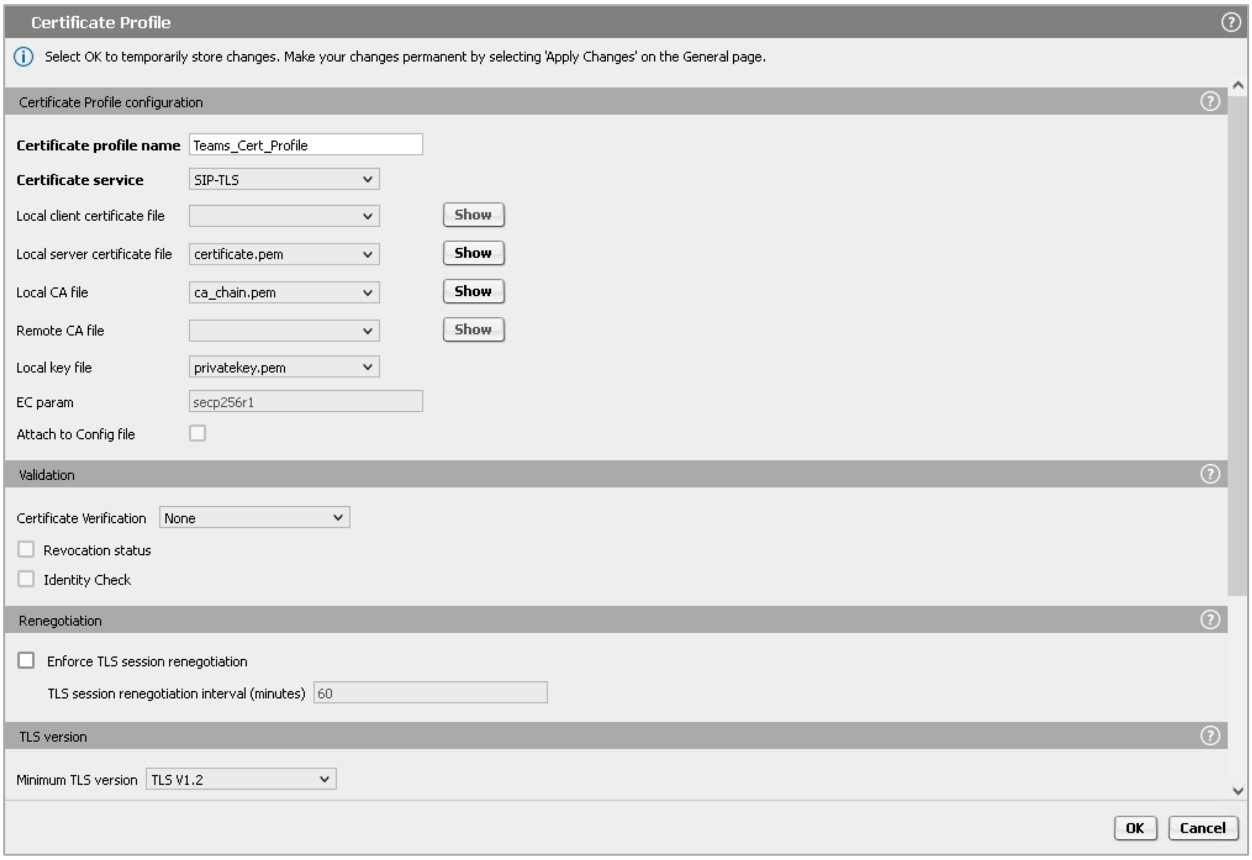
The files above must be uploaded to OS SBC and used for the TLS connection with the Microsoft Phone System interface.

1. Go to OS SBC **Management Portal > Security > General** and click on **Certificate Management**.

2. Upload "ca_chain.pem" to **CA Certificates**, "certificate.pem" to **"X.509"** and **"**privatekey.pem" to **"Key Files"** areas correspondingly, as shown in figure below:



3. Click **Add** to create the certificate profile.

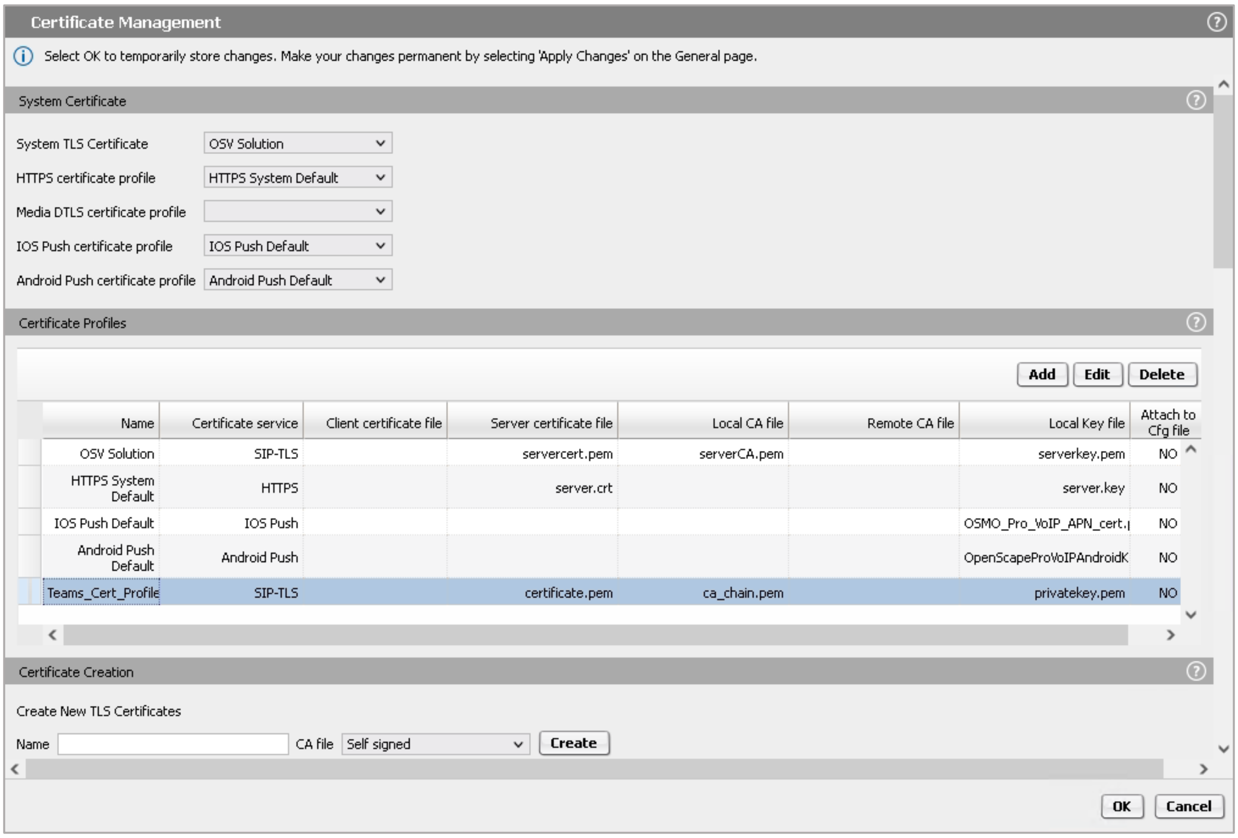


4. Enter the following:

- **Certificate profile name**: Teams_Cert_Profile (friendly name)
- Certificate service: SIP-TLS
- Local server certificate file: certificate.pem
- **Local CA file:** ca_chain.pem
- Local key file: privatekey.pem
- Minimum TLS version: TLS V1.2

5. Click **OK**.

Note: In case of MTLS is required, the **"Remote CA file"** should be selected (i.e., the Teams *"Baltimore"* CA certificate should have been uploaded to **"CA Certificates"** store) and a proper value should be selected for **"Certificate Verification"**.



5. Click **OK** in the **Certificate Management** window and then click **OK** in the **Security** window.

6. Click **Apply Changes** on OS SBC main page.

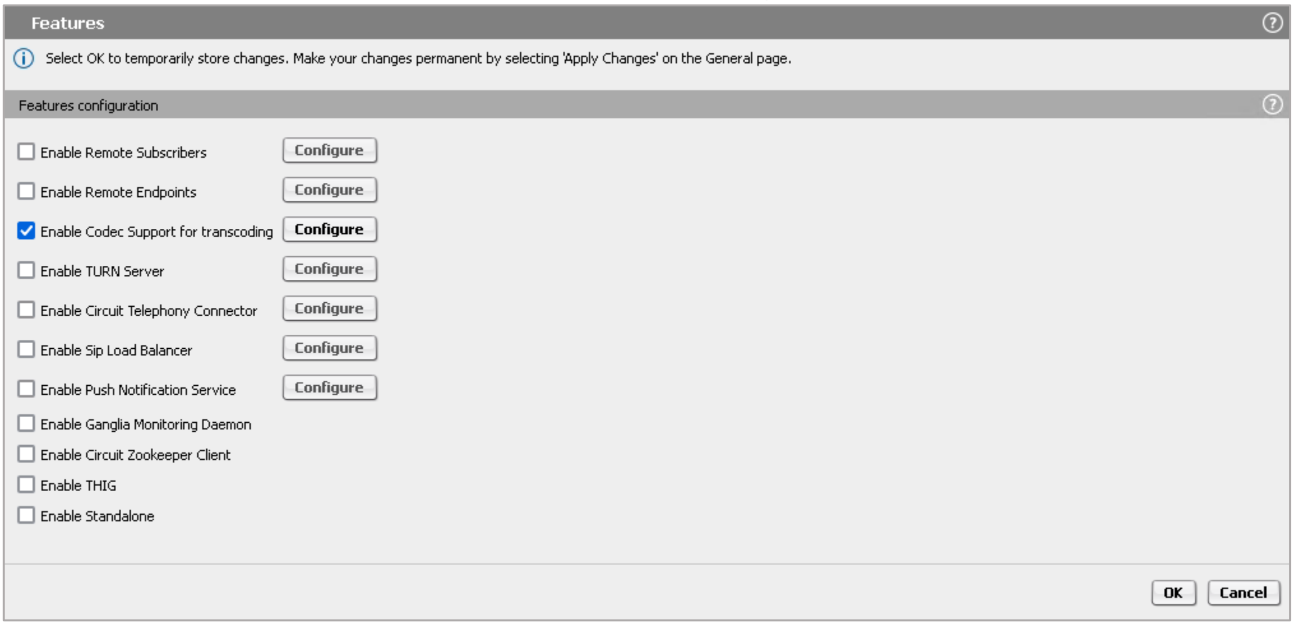
3.5 Configuring Media

With **Media Profiles** settings, various parameters regarding the SDP messages and audio (RTP) traffic may be configured for the OS SBC SIP endpoints to Teams Phone System, SSP (PSTN provider) and OS Voice.

3.5.1 Codec Manipulation Options

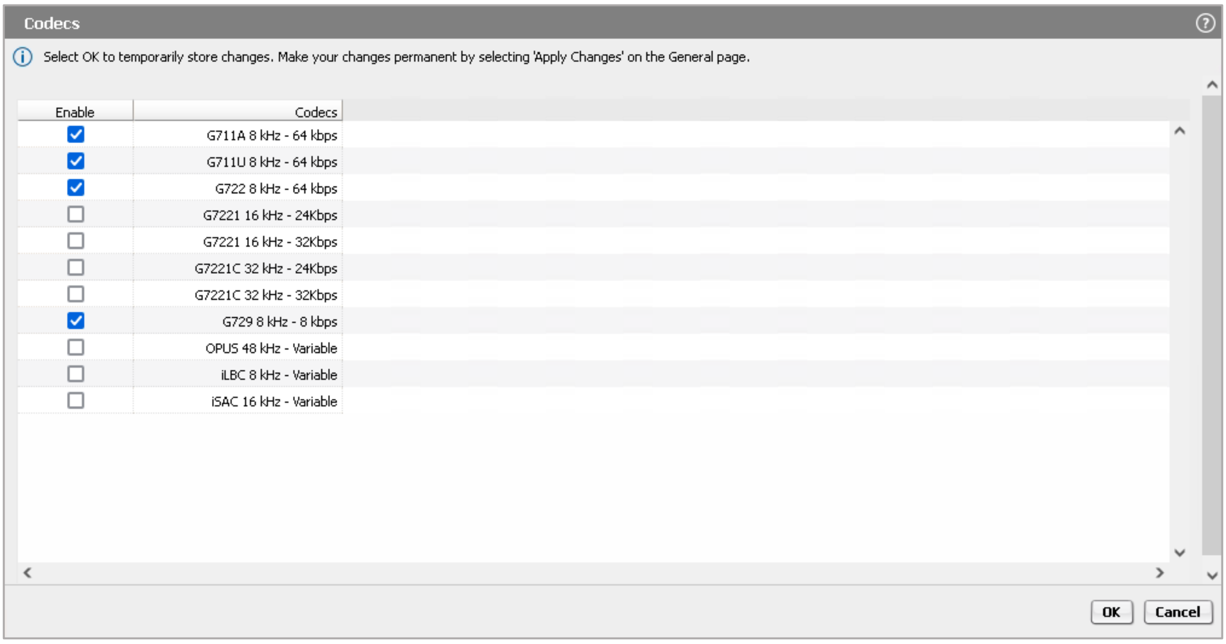
In case transcoding or certain codec prioritization for audio is required for the OS SBC – Teams Phone System & OS SBC – SSP media profiles for the corresponding SIP trunks, it is required to enable the codec configuration options first for the media profile setup.

1. Go to the **OS SBC Management Portal > Features** window.
2. Select the "**Enable Codec Support for transcoding**" checkbox.



3. Click **Configure**.

4. In the **Codecs** window, select the codecs to be available for the media profiles (for e.g., transcoding, prioritization), as shown in the example below:



5. Click **OK** on all the open windows.

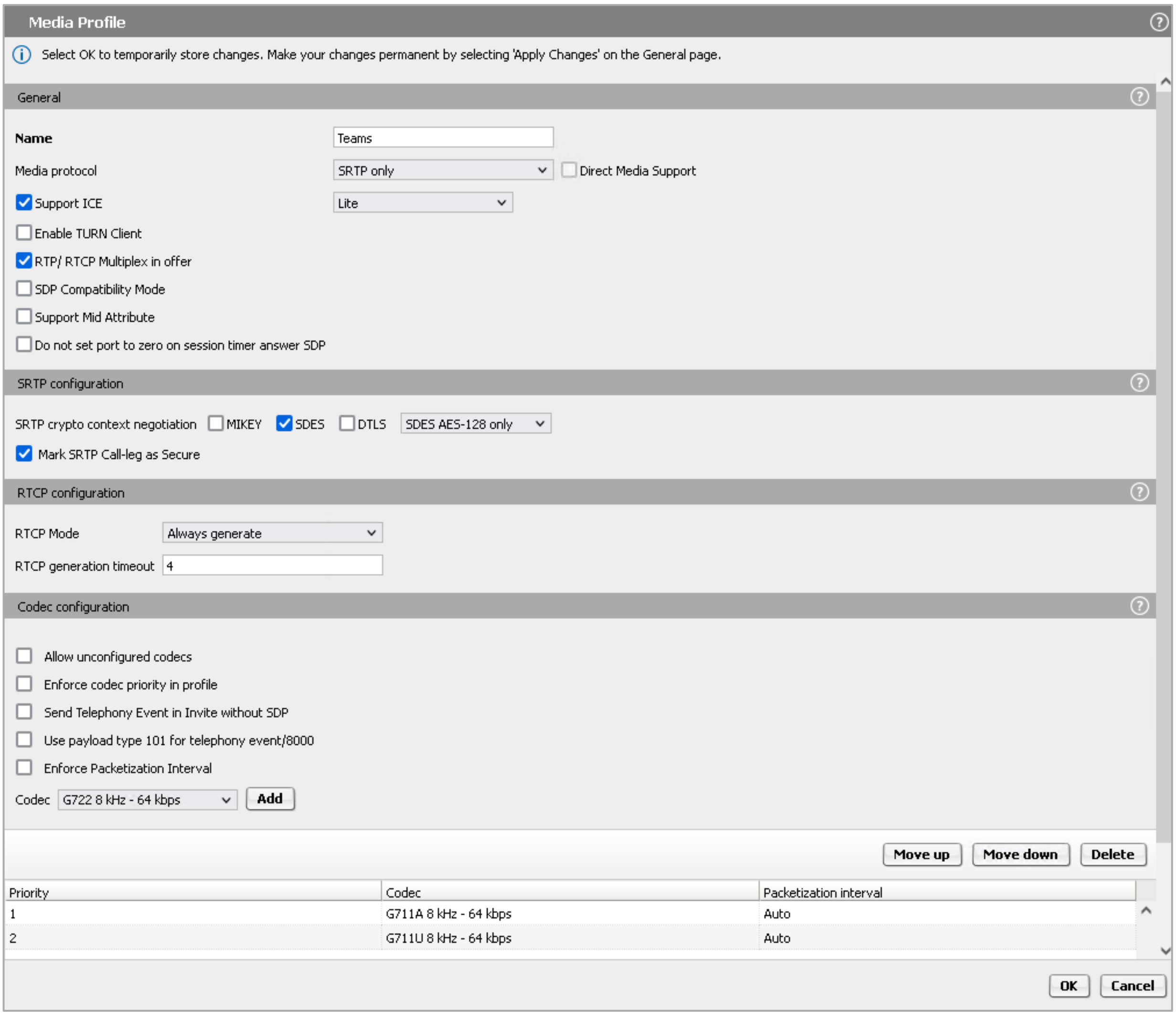
6. Click **Apply Changes**.

3.5.2 MS Teams Media Profile Configuration

The communication between Teams Phone System is secure with secure audio (SRTP).

In the example presented in current sub-section, it's supposed that the PSTN doesn't support G.711 and transcoding to G.711 is required for calls between PSTN subscribers and Teams clients on OS SBC – Teams Phone System SIP trunks.

1. Go to **OS SBC Management Portal > VOIP > Media**.



2. In the **Media Profiles** area click on **[Add]** to create the media profile for OS SBC - Teams connections by entering the following:

   - **Name:** Teams (friendly name)
   - **Media protocol**: SRTP only
   - **Support ICE**: Deactivated (for MB = OFF case) / Activated – Lite (for MB = ON case)
   - **RTP/RTCP Multiplex in offer**: Activated (adds rtcp-mux support to outgoing SDP)
   - **SRTP crypto context negotiation**: SDES
   - **Mark SRTP Call-leg as Secure**: Activated

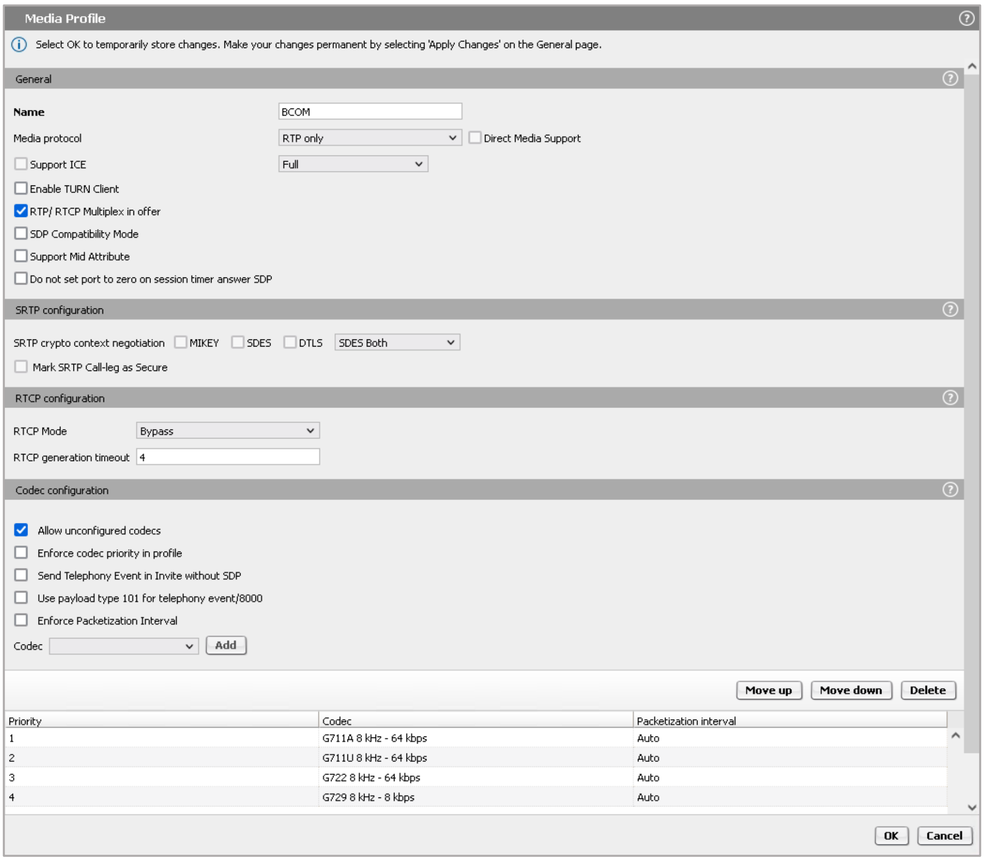
- **rtcpMode**: Bypass or Always Generate
  (with **Bypass** RTCP, messages from the SSP will be forwarded to Teams; with **Always Generate** RTCP messages will be generated from OS SBC to Teams, unless RTCP is received from SSP.)
- **Allow unconfigured codecs**: Deactivated

3. Click **Add** to add the desired codecs (with priority) for transcoding, e.g., G711A, G711U.
4. Click **OK** to return to the **Media** window.
5. Click **OK** in the **VOIP** window.
6. Click **Apply Changes.**

3.5.3 PSTN Service Provider Media Profile Configuration

In this sub-section, as an example, it is assumed that certain codecs need to be prioritized on the OS SBC – SSP (BCOM) SIP trunk for calls between Teams clients and PSTN subscribers.

1. Go to **OS SBC Management Portal > VOIP > Media**.



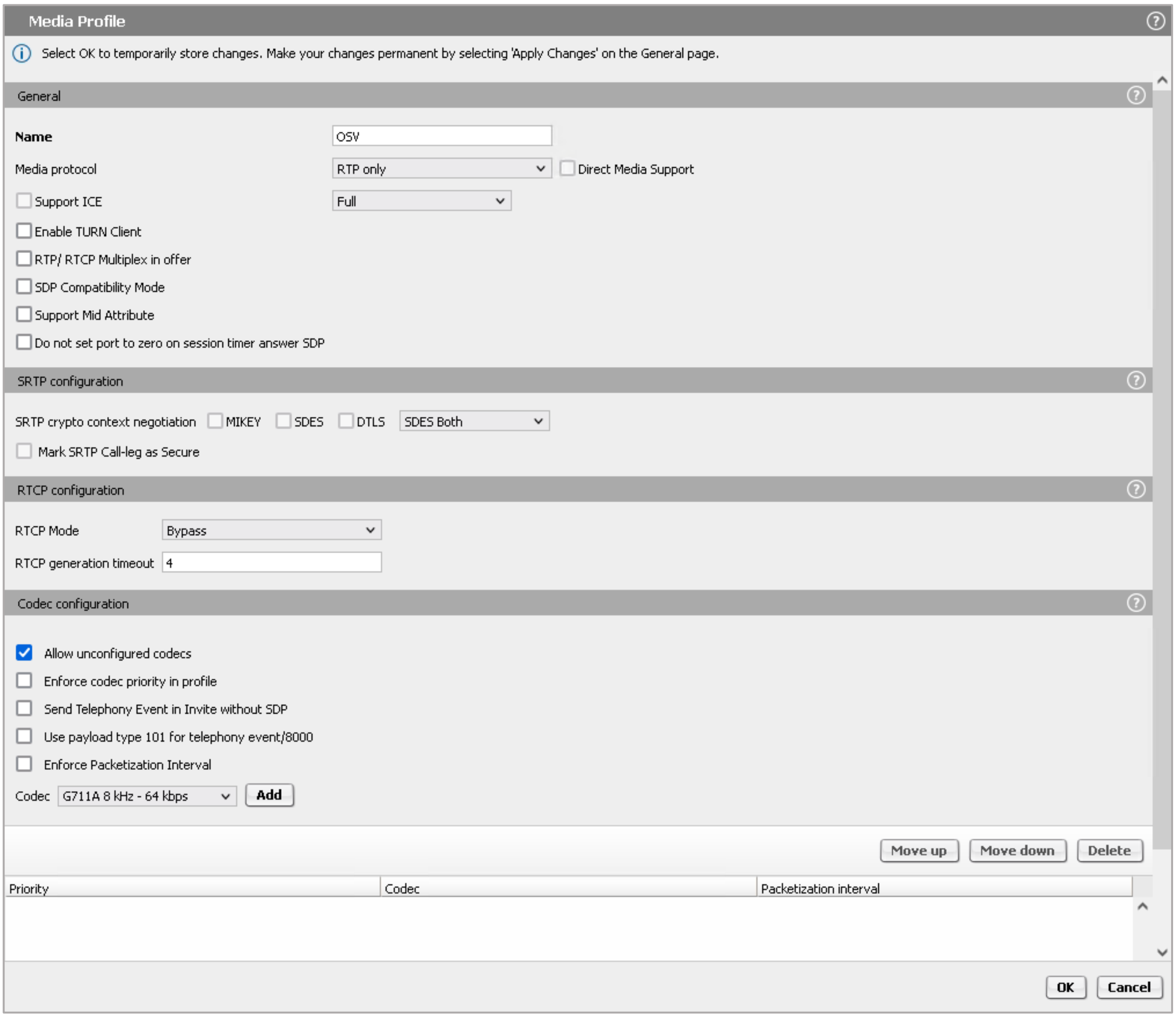
2. In the **Media Profiles** area, click **Add** to create the media profile for OS SBC - SSP trunk:

- **Name**: BCOM (friendly name)
- **Media protocol**: RTP only
- **RTP/RTCP Multiplex in offer**: Activated
- **rtcpMode**: Bypass or Always Generate
- **Allow unconfigured codecs**: Activated

3. If codec prioritization is required on the trunk over the other codecs (**"Enforce codec priority in profile"** is "Activated"), click **Add** to add the desired codecs to be prioritized (e.g., G711A, G711U, G722, G729).
4. Click **OK** to return to the **Media** window.
5. Click **OK** in the **VOIP** window.
6. Click **Apply Changes.**

3.5.4 OpenScape Voice Media Profile Configuration

1. Go to **OS SBC Management Portal > VOIP > Media**.



2. In the **Media Profiles** area click on **[Add]** to create the media profile for OS SBC – OS Voice connection by entering the following:

- **Name:** OSV (friendly name)
- **Media protocol**: RTP only
- **RTP/RTCP Multiplex in offer**: Deactivated
- **Allow unconfigured codecs**: Activated

3. Click **OK** to return to the **"Media"** window.

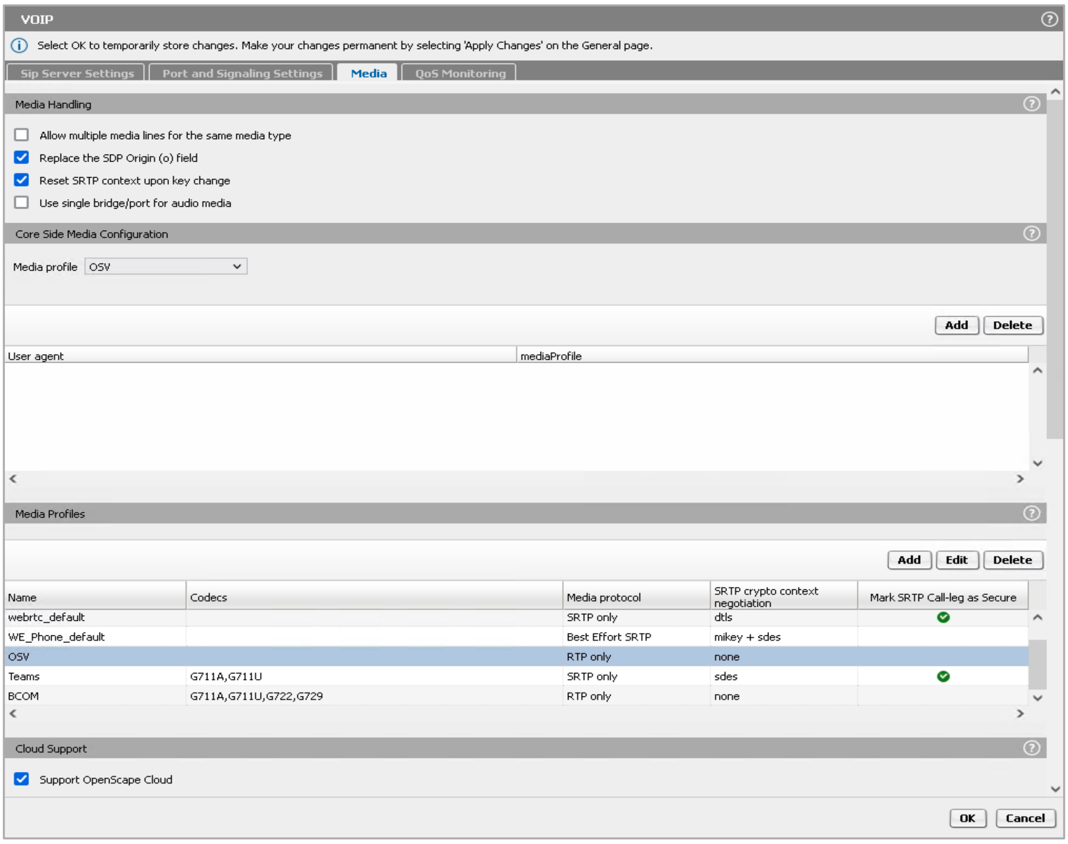
4. Click **OK** in the **"VOIP"** window.

5. Click **Apply Changes**.

3.5.5 General Media Settings

After creating the media profiles, configure the General media settings.

1. Go to the **OS SBC Management Portal** > **VOIP** > **Media** window.



2. In the **"Core Side Media Configuration"** area, select "OSV" from the **"Media profile"** drop-down list for the media profile used for the OS SBC – OS Voice SIP trunk.

3. Locate the **Cloud Support** area and select the **"Support OpenScape Cloud"** checkbox.

4. Click **OK**.

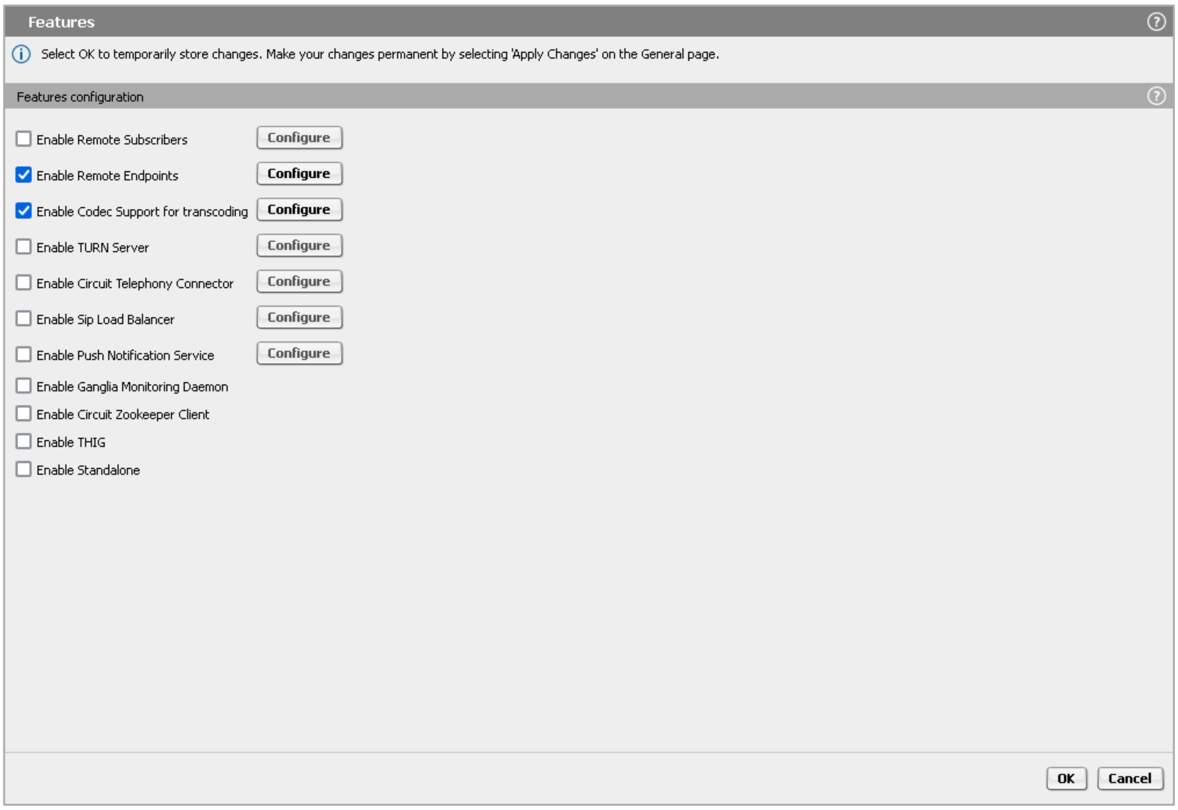
5. Click **Apply Changes**.

3.6 Configuring Remote Endpoints

You can set up the OS SBC with Teams Phone System and the PSTN (BCOM SSP) SIP trunks in the **Remote Endpoint** configuration area.

3.6.1 MS Teams Remote Endpoints Configuration

1. Go to the **OS SBC Management Portal > Features** window.
2. Select the **Enable Remote Endpoints** checkbox.



3. Click **Configure**.

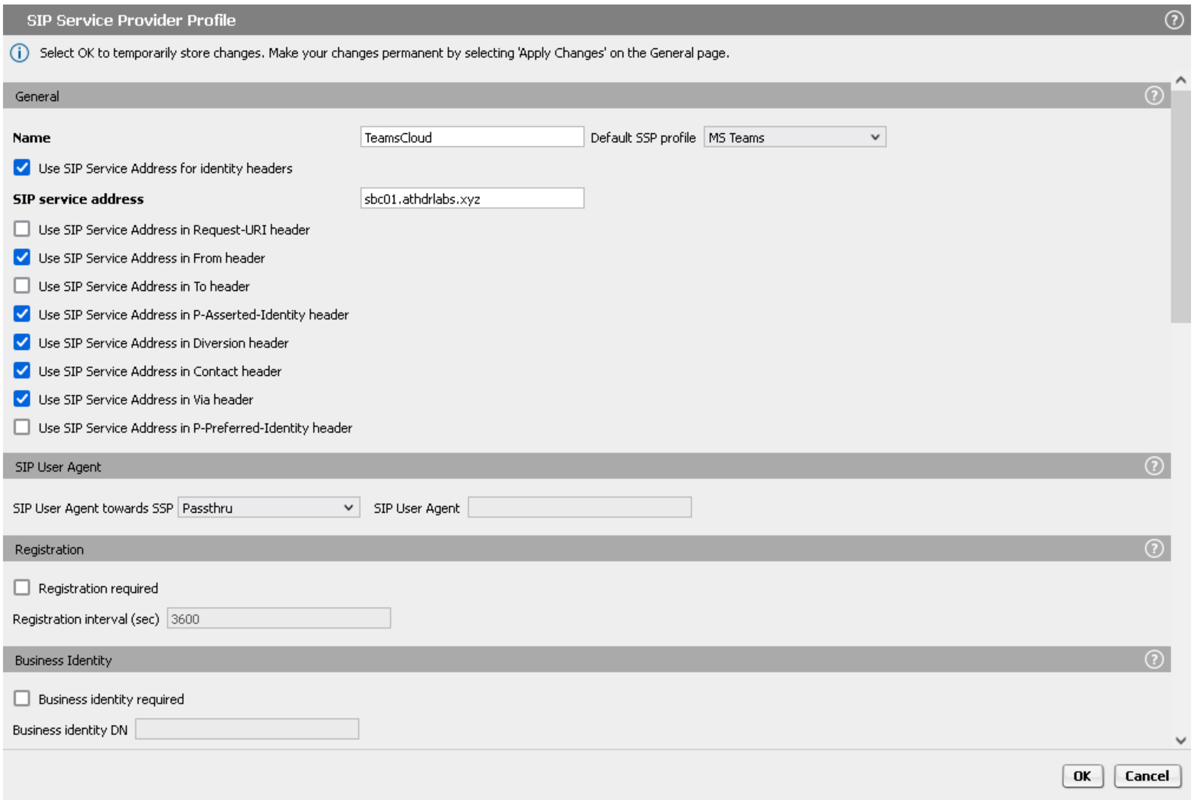
The **Remote Endpoints** window opens.

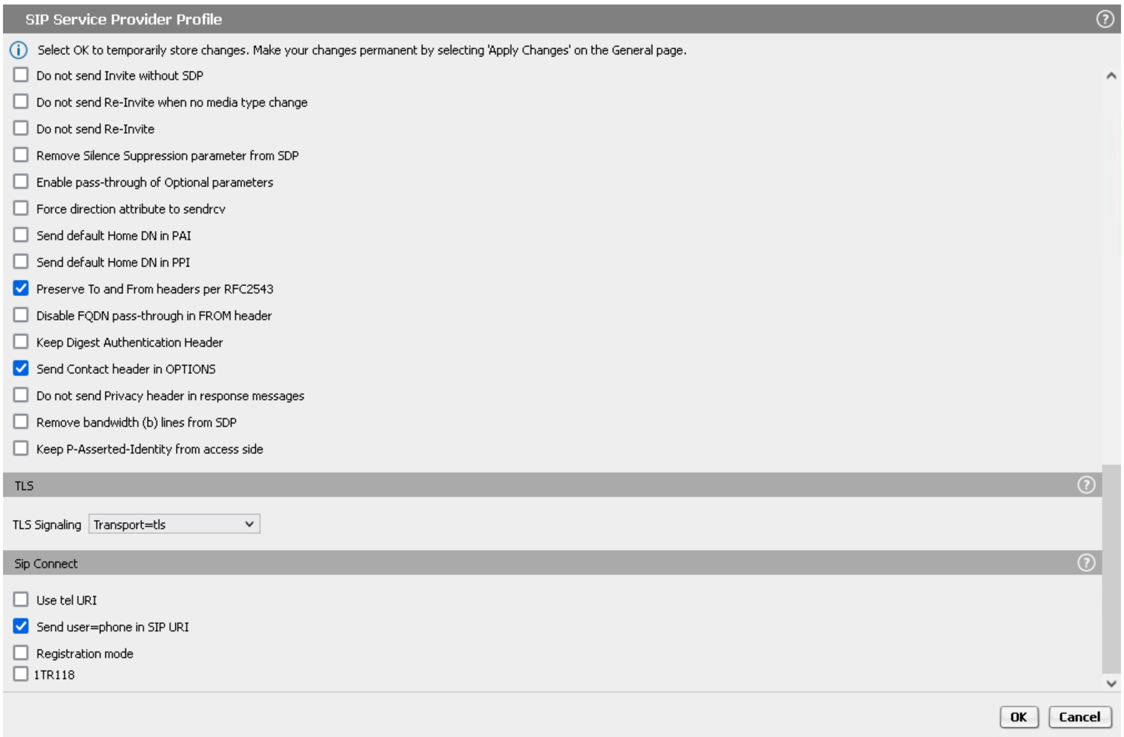
4. Click **Add** in the **"SIP Service Provider Profile"** area to add the endpoint profile for the OS SBC – Teams Phone System endpoint.

5. In the **SIP Service Provider** window, configure the following:

   - **Name**: TeamsCloud (friendly name)
   - **Default SSP Profile**: MS Teams
     (by selecting the MS Teams profile all the required flags are selected automatically)
   - **SIP service address**: sbc01.athdrlabs.xyz
     (SBC public FQDN)

   - Ensure that the checkboxes shown in the following images are selected:


SIP Service Provider Profile
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
General
Name
TeamsCloud
Default SSP profile
MS Teams
Use SIP Service Address for identity headers
SIP service address
sbc01.athdrlabs.xyz
Use SIP Service Address in Request-URI header
Use SIP Service Address in From header
Use SIP Service Address in To header
Use SIP Service Address in P-Asserted-Identity header
Use SIP Service Address in Diversion header
Use SIP Service Address in Contact header
Use SIP Service Address in Via header
Use SIP Service Address in P-Preferred-Identity header
SIP User Agent
SIP User Agent towards SSP
Passthru
SIP User Agent
Registration
Registration required
Registration interval (sec)
3600
Business Identity
Business identity required
Business identity DN
OK
Cancel

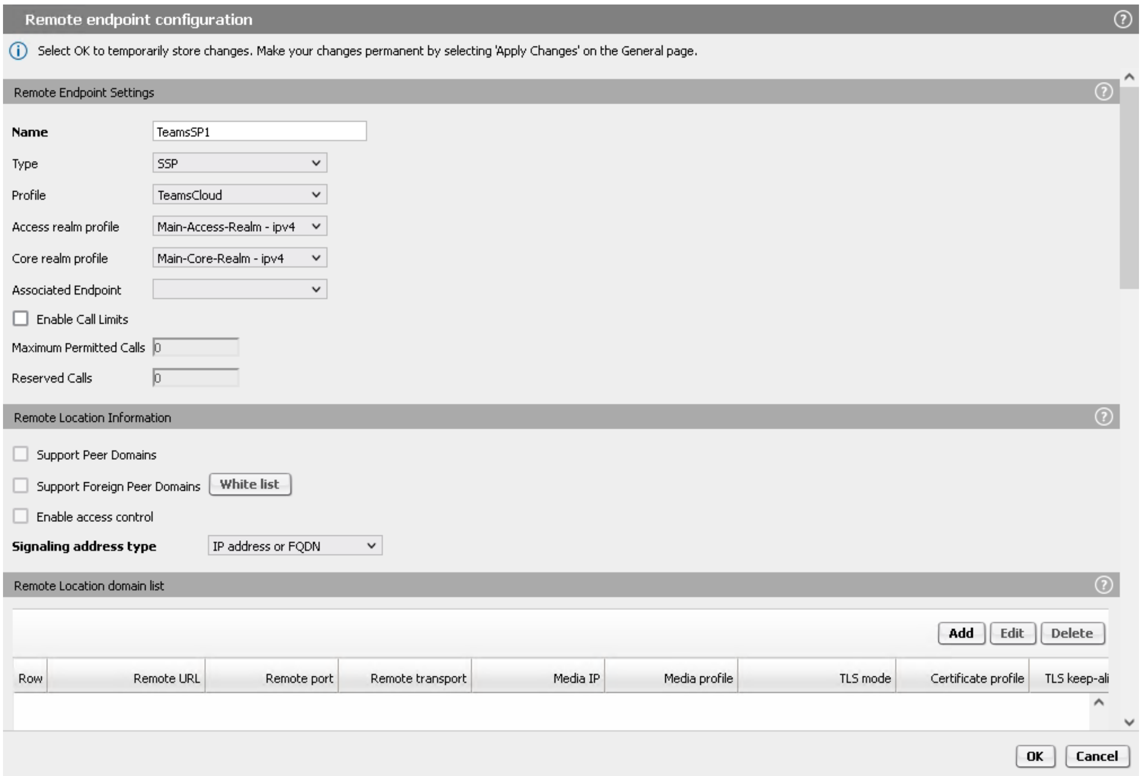

SIP Service Provider Profile
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
Do not send Invite without SDP
Do not send Re-Invite when no media type change
Do not send Re-Invite
Remove Silence Suppression parameter from SDP
Enable pass-through of Optional parameters
Force direction attribute to sendrcv
Send default Home DN in PAI
Send default Home DN in PPI
Preserve To and From headers per RFC2543
Disable FQDN pass-through in FROM header
Keep Digest Authentication Header
Send Contact header in OPTIONS
Do not send Privacy header in response messages
Remove bandwidth (b) lines from SDP
Keep P-Asserted-Identity from access side
TLS
TLS Signaling
Transport=tls
Sip Connect
Use tel URI
Send user=phone in SIP URI
Registration mode
1TR118
OK
Cancel

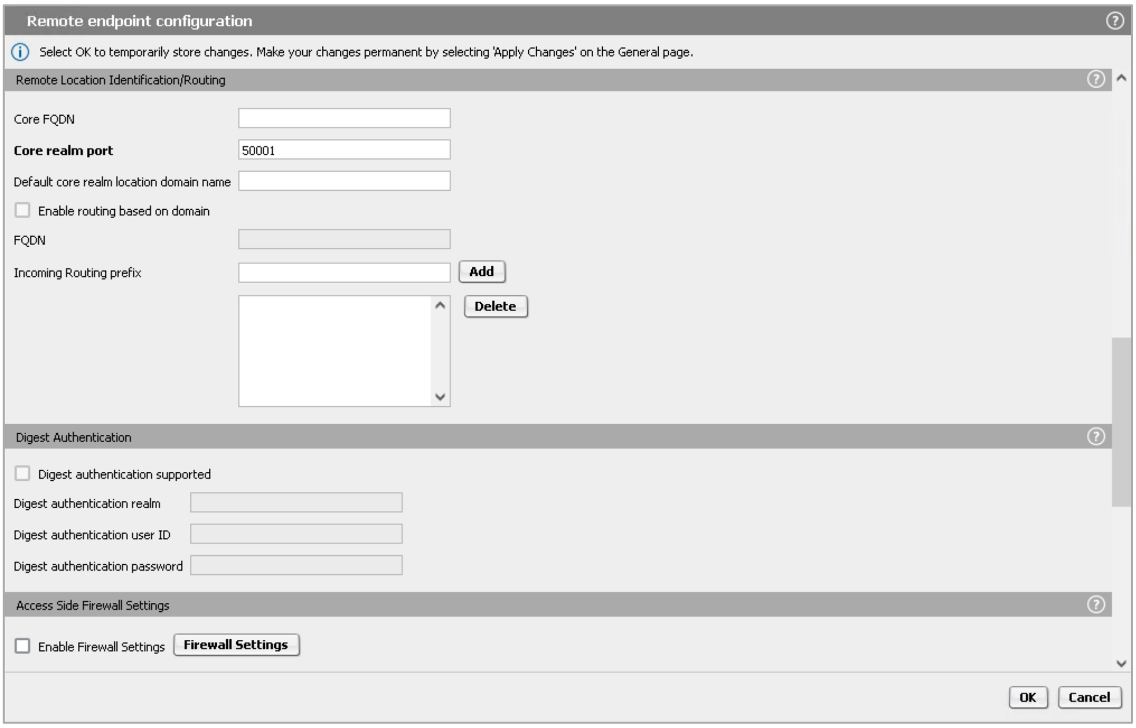
6. Click **OK** to return to the **Remote Endpoints** window.

7. In the **Remote endpoint configuration** area, click **Add**.

8. In the **Remote endpoint configuration** window, configure the following:

- **Name:** TeamsSP1 (friendly name)
- **Type**: SSP
- **Profile**: TeamsCloud
- **Signaling address type**: IP address or FQDN
- **Core realm port:** 50001 (refer to sub-section 2.1.2)

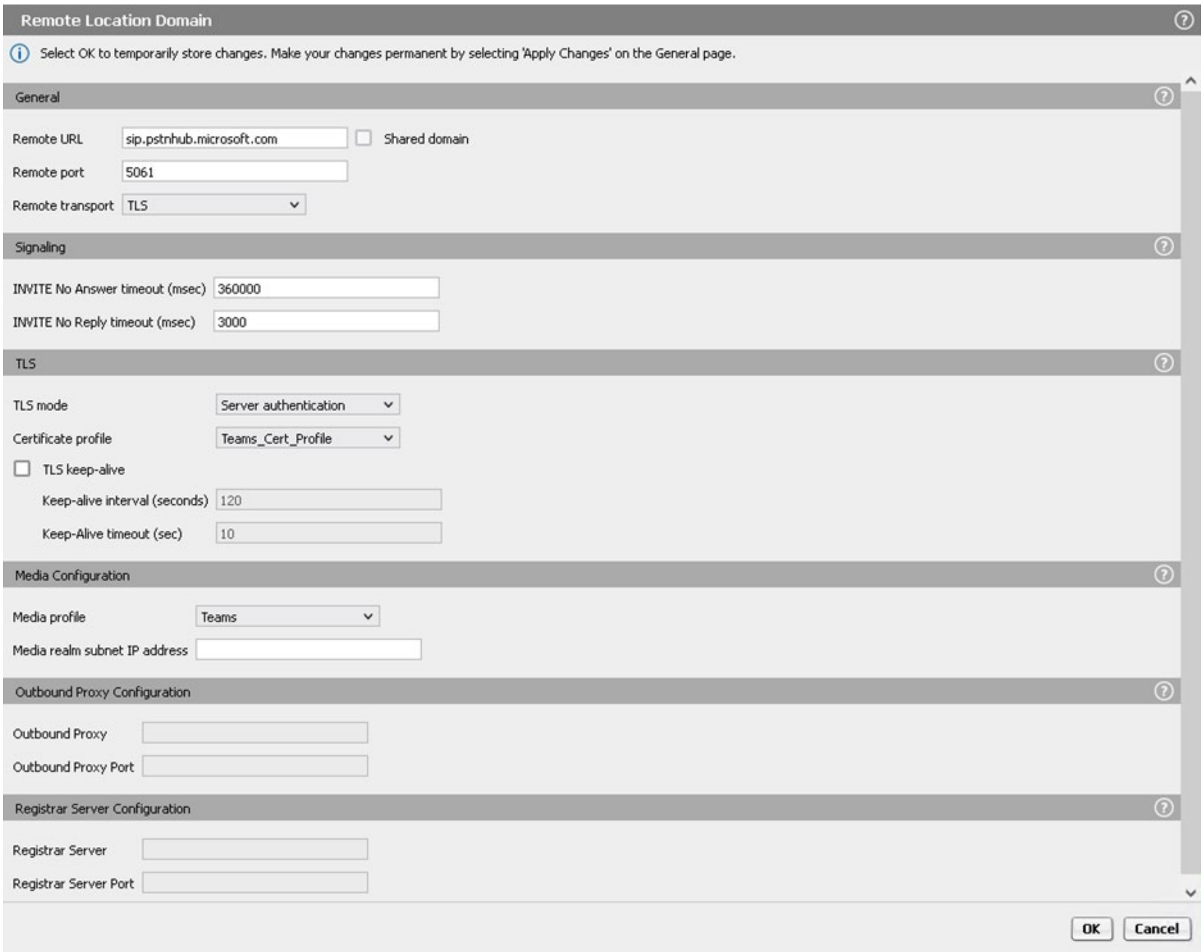




7. Click on **[Add]** in the **Remote Location domain list** area.

9. In the **Remote Location Domain** window, enter the following:

- **Remote URL:** sip.pstnhub.microsoft.com (Teams FQDN 1)
- **Remote port**: 5061
- **Remote transport**: TLS
- **TLS mode:** Server authentication
  (or Mutual authentication in case MTLS is required – refer to section 3.2)
- **Certificate profile:** Teams_Cert_Profile (refer to section 3.2)
- **Media profile:** Teams (refer to sub-section 3.3.2)

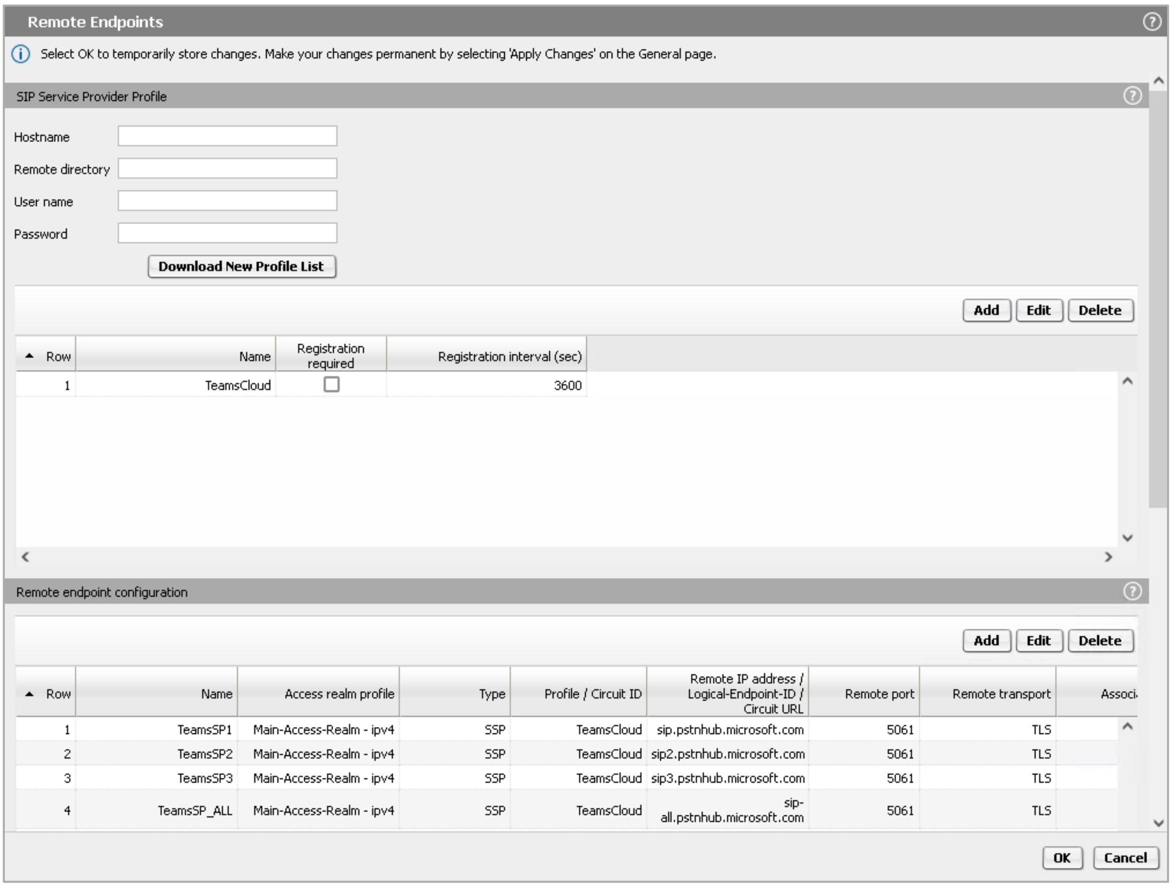


10. Click **OK** to return to the **Remote endpoint configuration** window.

11. Repeat the procedure **Remote endpoint configuration** for the rest MS Teams FQDNs (i.e., sip2.pstnhub.microsoft.com, sip3.pstnhub.microsoft.com and sip-all.pstnhub.microsoft.com - refer to chapter 3).

12. Click **OK** to return to the **Remote Endpoints** window.

The **"Remote Endpoints"** window should look like the figure below:



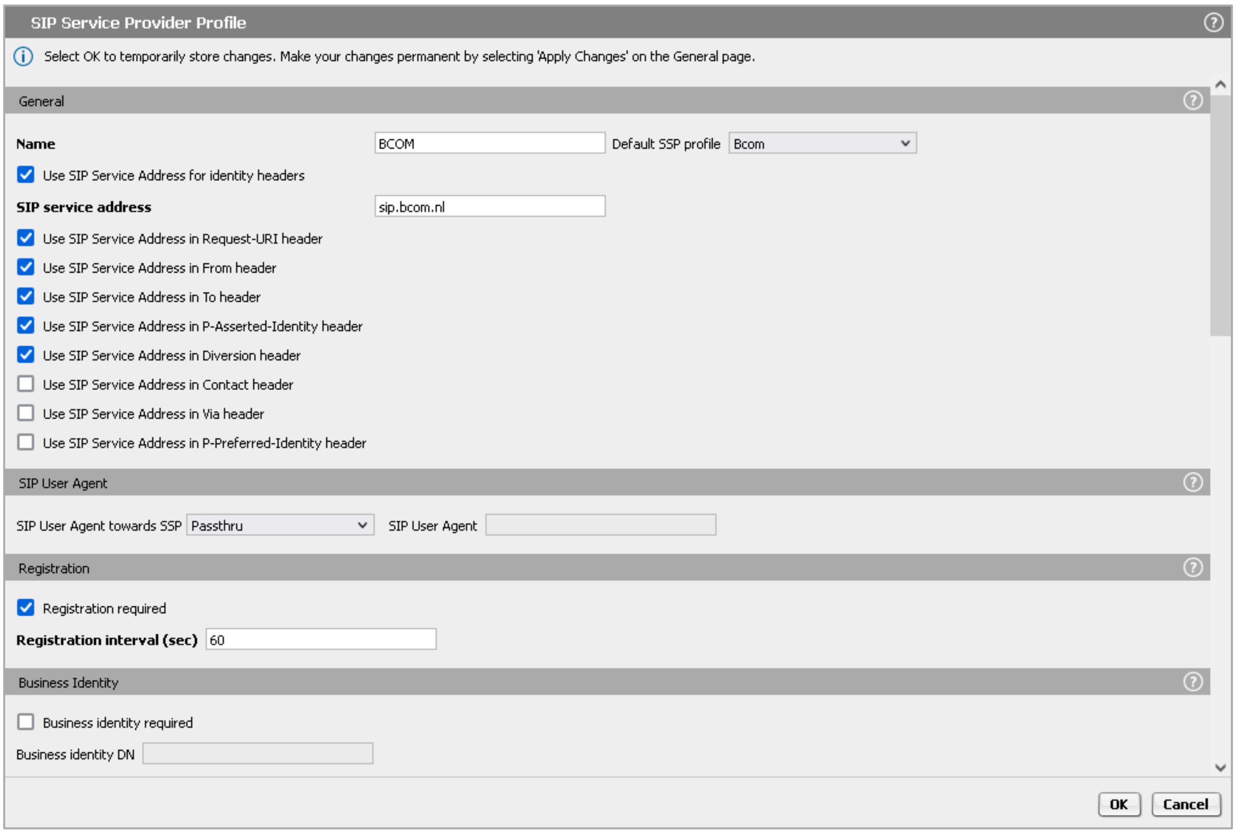
13. Click **OK** on all open windows.
14. Click **Apply Changes** on OS SBC main page.

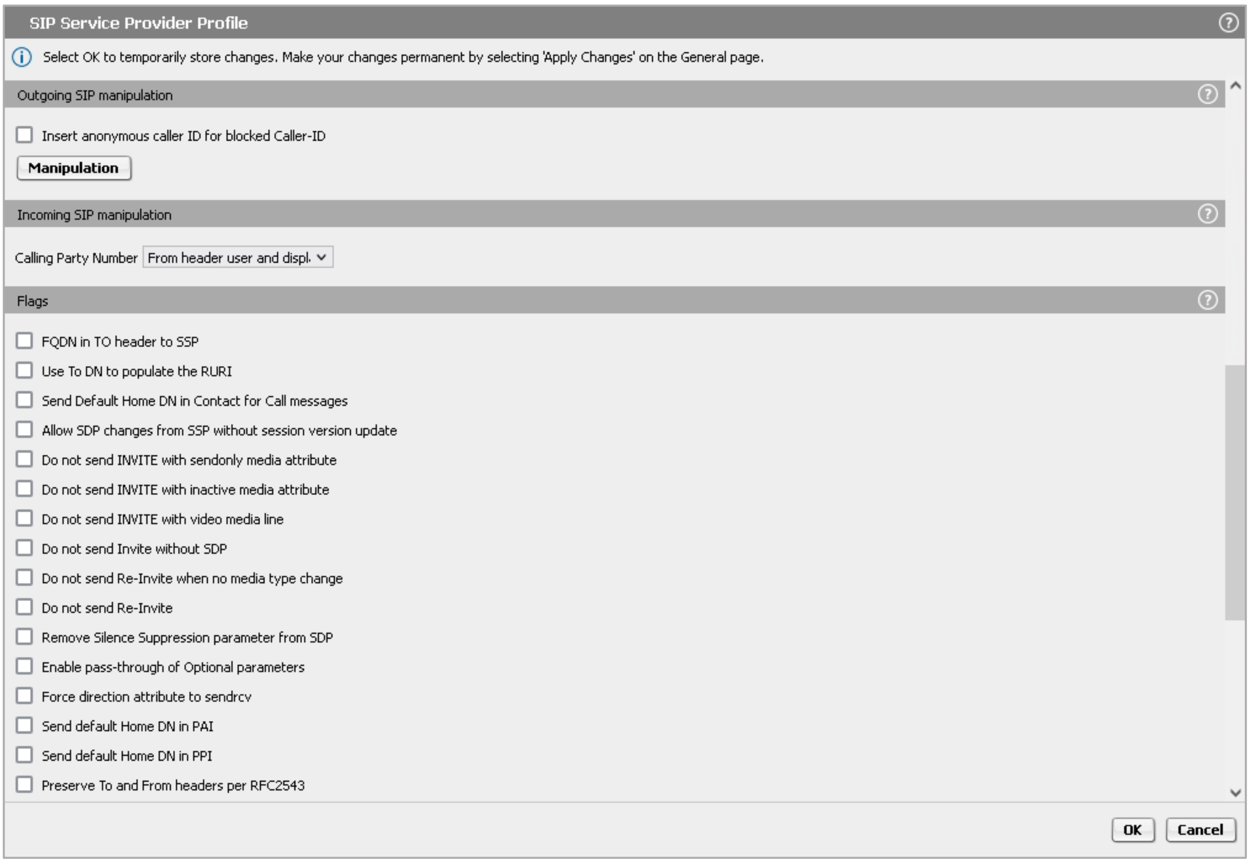
3.6.2 PSTN Remote Endpoint Configuration

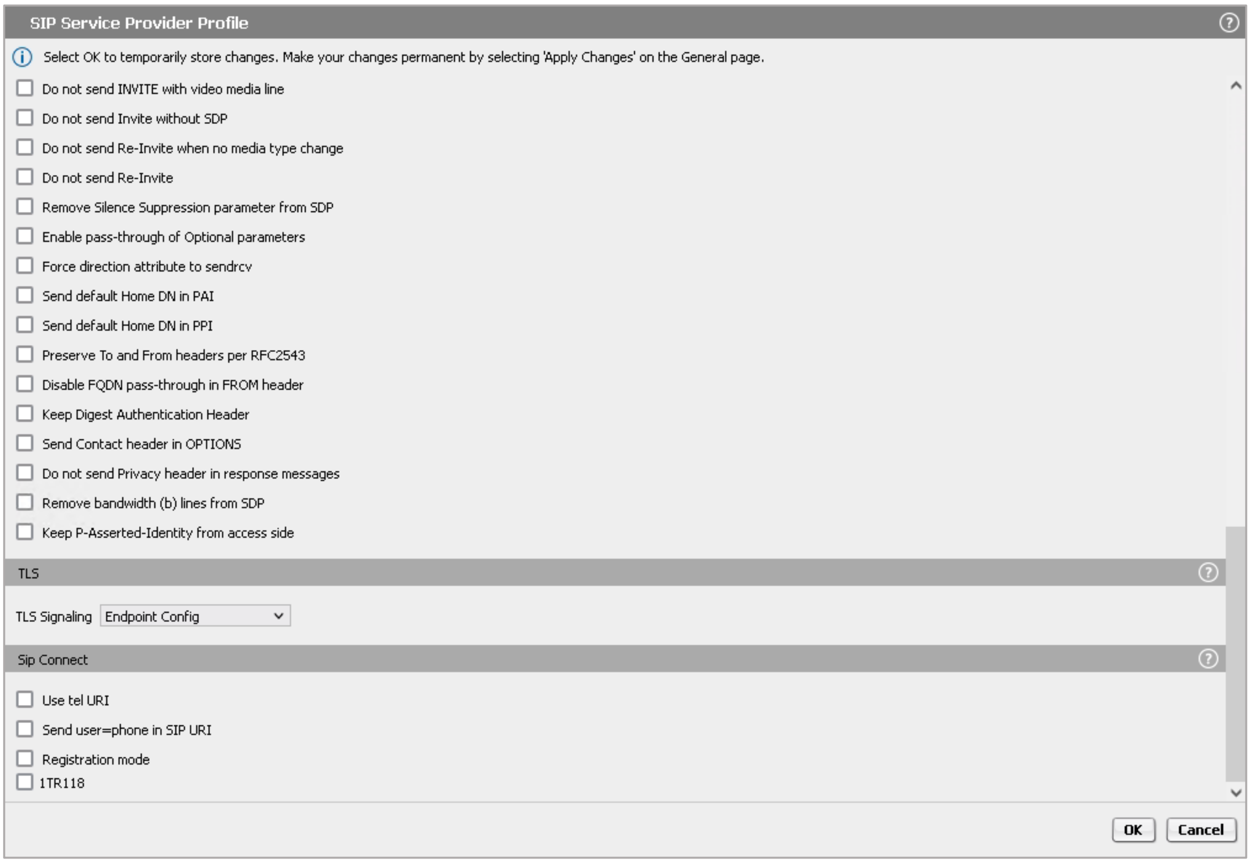
1. Go to **OS SBC Management Portal > Features > Enable Remote Endpoints**.
2. Click **Configure**.

The **Remote Endpoints** window opens.

3. Click **Add** in the **SIP Service Provider Profile** area to add the endpoint profile for the OS SBC – SSP (BCOM) endpoint.


SIP Service Provider Profile
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
General
Name
BCOM
Default SSP profile
Bcom
Use SIP Service Address for identity headers
SIP service address
sip.bcom.nl
Use SIP Service Address in Request-URI header
Use SIP Service Address in From header
Use SIP Service Address in To header
Use SIP Service Address in P-Asserted-Identity header
Use SIP Service Address in Diversion header
Use SIP Service Address in Contact header
Use SIP Service Address in Via header
Use SIP Service Address in P-Preferred-Identity header
SIP User Agent
SIP User Agent towards SSP
Passthru
SIP User Agent
Registration
Registration required
Registration interval (sec)
60
Business Identity
Business identity required
Business identity DN
OK
Cancel


SIP Service Provider Profile
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
Outgoing SIP manipulation
Insert anonymous caller ID for blocked Caller-ID
Manipulation
Incoming SIP manipulation
Calling Party Number
From header user and displ.
Flags
FQDN in TO header to SSP
Use To DN to populate the RURI
Send Default Home DN in Contact for Call messages
Allow SDP changes from SSP without session version update
Do not send INVITE with sendonly media attribute
Do not send INVITE with inactive media attribute
Do not send INVITE with video media line
Do not send Invite without SDP
Do not send Re-Invite when no media type change
Do not send Re-Invite
Remove Silence Suppression parameter from SDP
Enable pass-through of Optional parameters
Force direction attribute to sendrcv
Send default Home DN in PAI
Send default Home DN in PPI
Preserve To and From headers per RFC2543
OK
Cancel



4. In the **SIP Service Provider** window, configure the following:

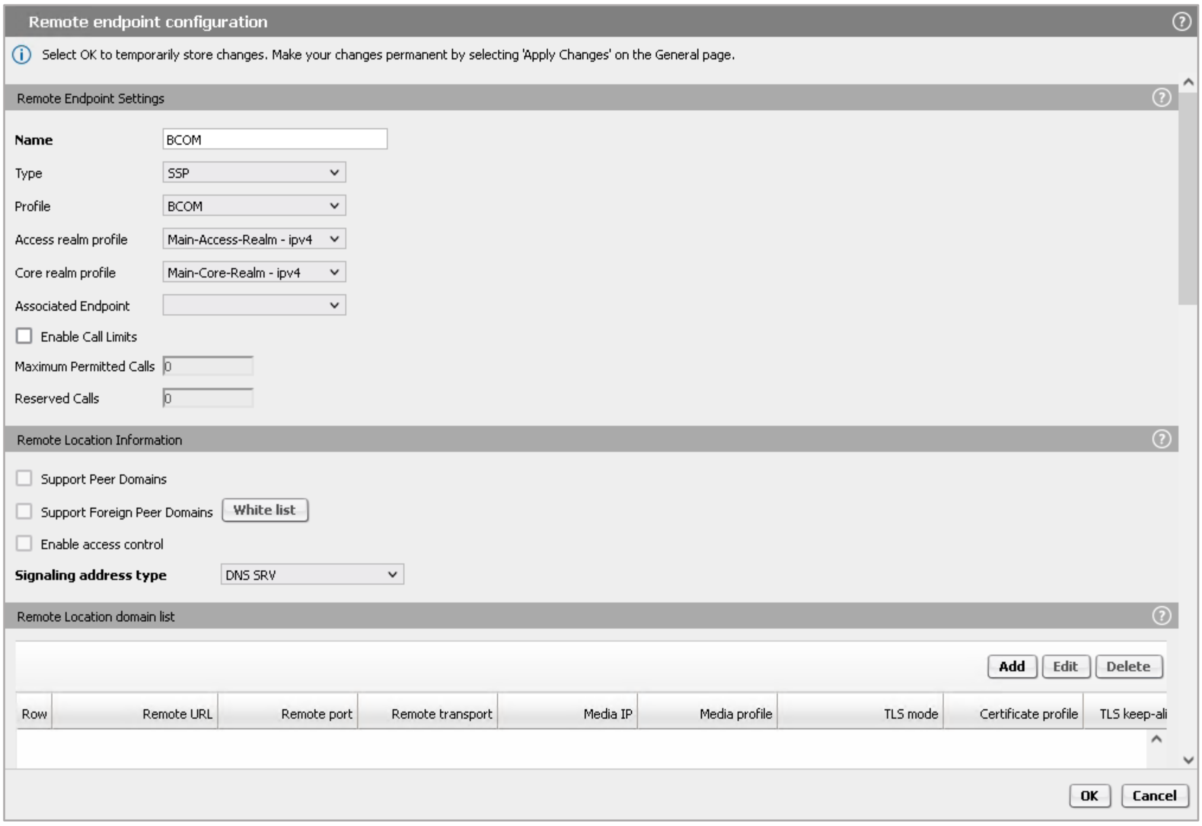
- **Name**: BCOM
  (friendly name)
- **Default SSP Profile**: BCOM

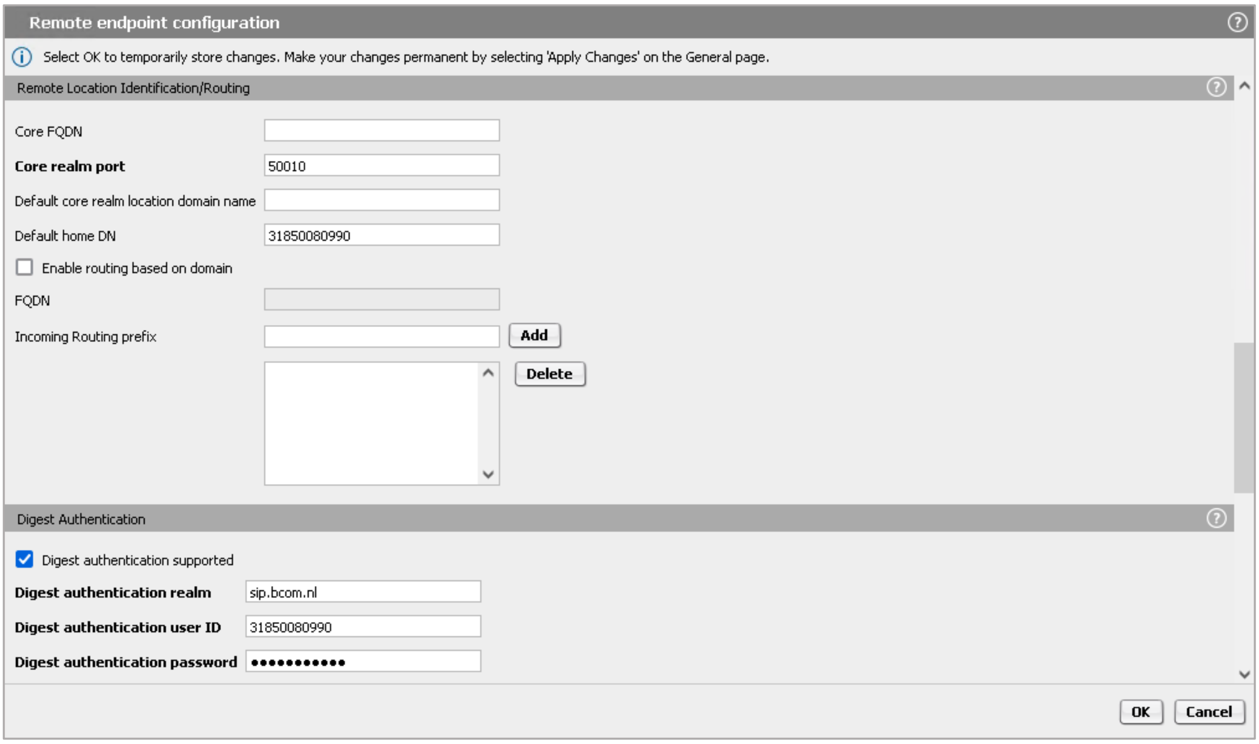
Note: Selecting the *"Bcom"* profile automatically enables all required flags. If the provider is not available in the drop-down list, leave this field blank and manually configure the required flags for the SSP in use.

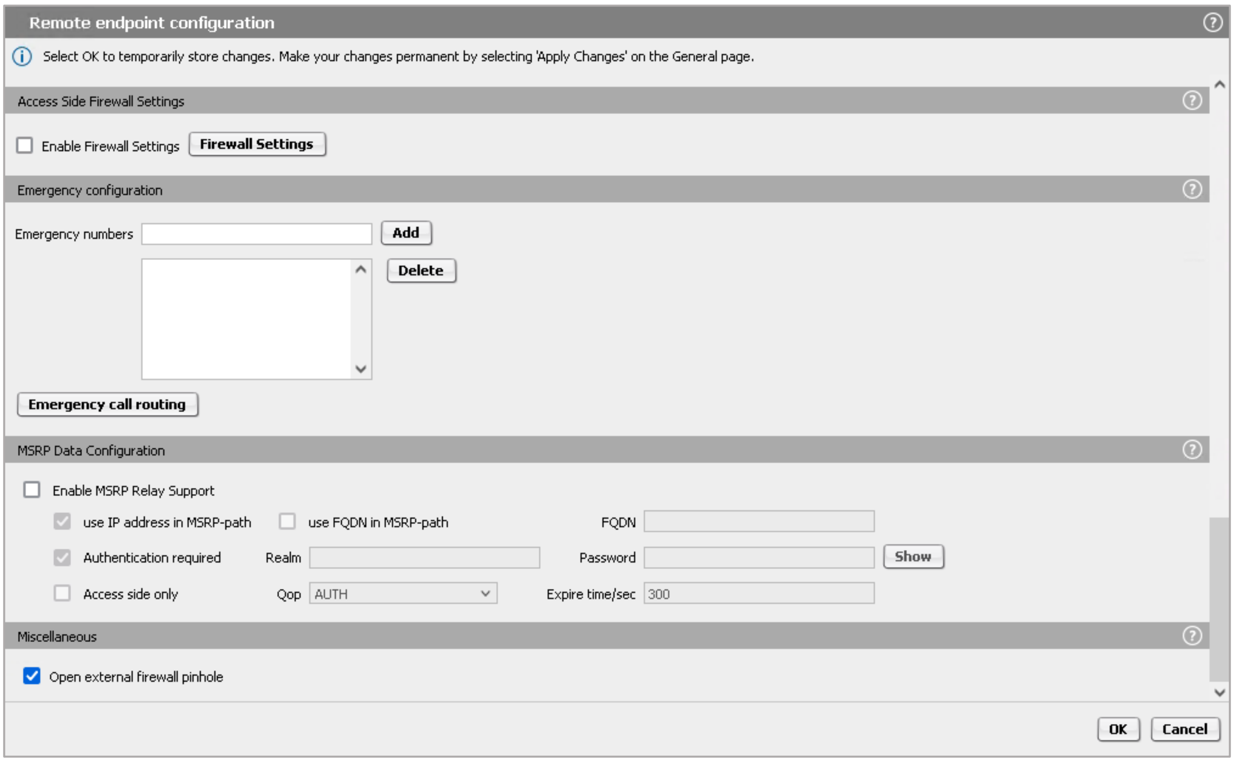
- **Use SIP Service Address for identity headers**: BCOM
- **Use SIP Service Address in Request-URI header**: Activated
- **Use SIP Service Address in From header**: Activated
- **Use SIP Service Address in To header**: Activated
- **Use SIP Service Address in P-Asserted-Identity header**: Activated
- **Use SIP Service Address in Diversion header**: Activated

5. Click **OK** to return to the **Remote Endpoints** window.

6. In the **Remote endpoint configuration** area, click **Add**.


Remote endpoint configuration
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
Remote Endpoint Settings
Name
BCOM
Type
SSP
Profile
BCOM
Access realm profile
Main-Access-Realm - ipv4
Core realm profile
Main-Core-Realm - ipv4
Associated Endpoint
Enable Call Limits
Maximum Permitted Calls
0
Reserved Calls
0
Remote Location Information
Support Peer Domains
Support Foreign Peer Domains
White list
Enable access control
Signaling address type
DNS SRV
Remote Location domain list
Add
Edit
Delete
Row
Remote URL
Remote port
Remote transport
Media IP
Media profile
TLS mode
Certificate profile
TLS keep-ali
OK
Cancel

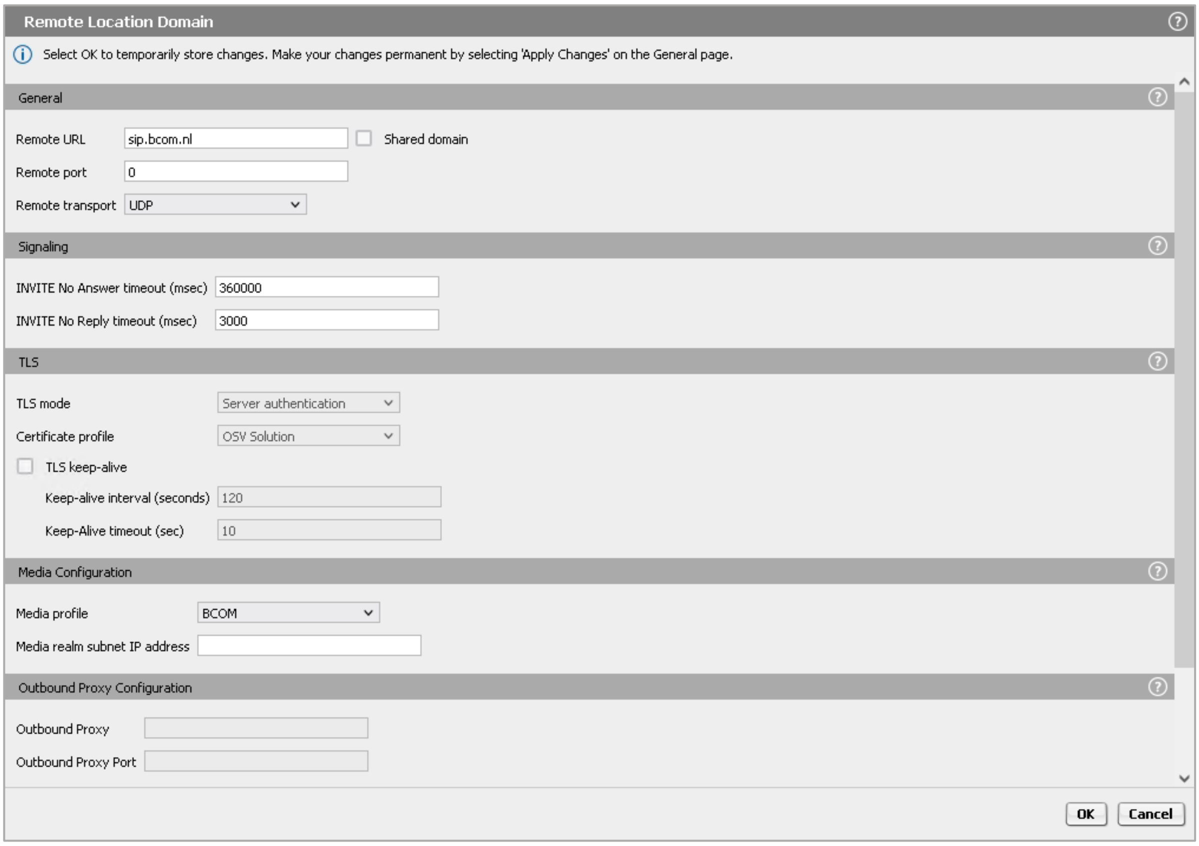

Remote endpoint configuration
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
Remote Location Identification/Routing
Core FQDN
Core realm port
50010
Default core realm location domain name
Default home DN
31850080990
Enable routing based on domain
FQDN
Incoming Routing prefix
Add
Delete
Digest Authentication
Digest authentication supported
Digest authentication realm
sip.bcom.nl
Digest authentication user ID
31850080990
Digest authentication password
OK
Cancel



7. In the **Remote endpoint configuration** window, configure the following:

- **Name**: BCOM (friendly name)
- **Type**: SSP
- **Profile**: BCOM
- **Signaling address type**: DNS SRV
- **Core realm port**: 50010 (refer to sub-section 2.1.3)
- **Default home DN**: 31850080990
- **Digest authentication supported**: Activated
- **Digest authentication realm:** sip.bcom.nl (data provided by the BCOM SSP)
- **Digest authentication user ID:** 31850080990 (data provided by the BCOM SSP)
- **Digest authentication password:** *<password>* (data provided by the BCOM SSP)
- Open external firewall pinhole: Activated

8. Click **Add** in the **Remote Location domain list** area.



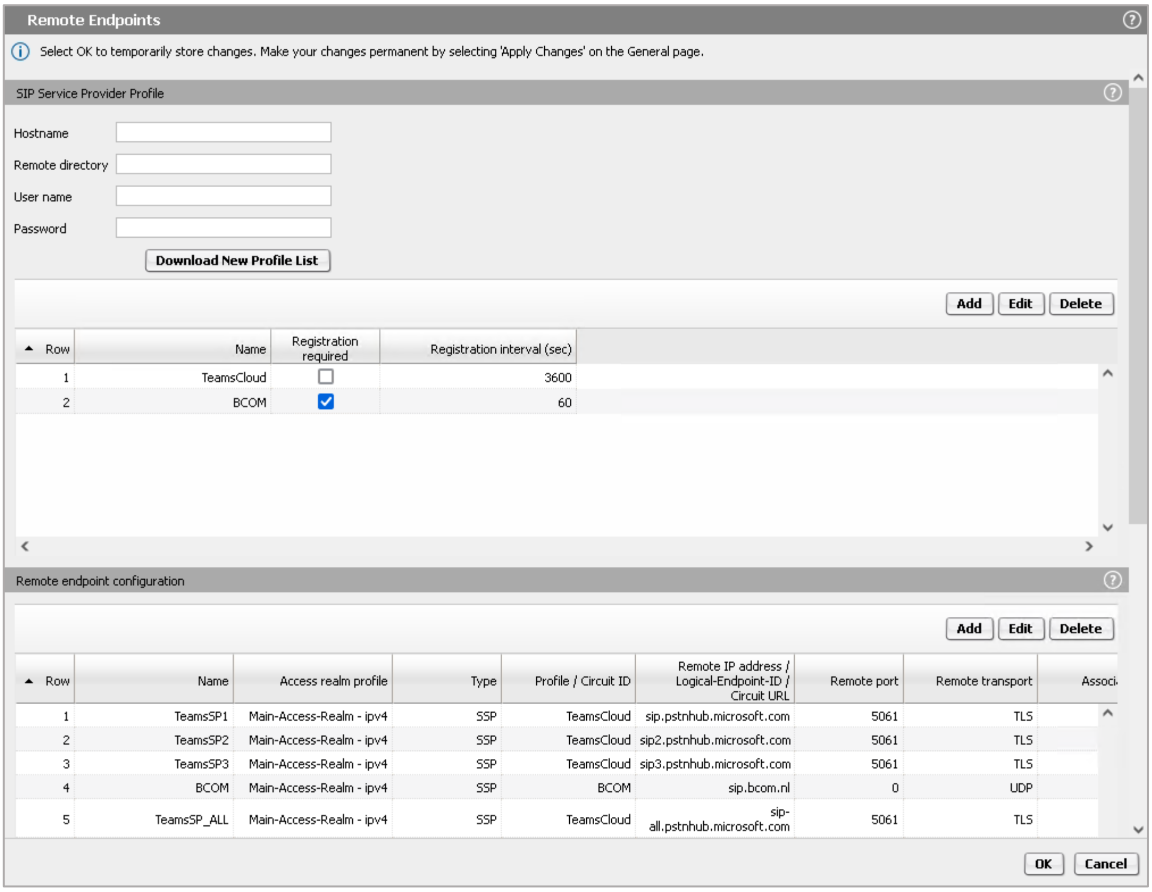
8. In the **Remote Location Domain** window, enter the following:

- **Remote URL**: sip.bcom.nl (BCOM)
- **Remote transport**: UDP
- **Media profile**: BCOM (refer to sub-section 3.3.4)

9. Click **OK** to return to the **Remote endpoint configuration** window.

10. Click **OK** to return to the **Remote Endpoints** window.

The **"Remote Endpoints"** window should look like the figure below:



11. Click **OK** on all open windows.
12. Click **Apply Changes.**