COX SIP Trunking Configuration Guide for Unify OpenScape Voice

Version 9 R4.41.0

Unify OpenScape Voice SBC

Version 9 R4.11.00

Feb 2020

Contents

1 Audience

This document is intended for the SIP trunk customer's technical staff and Value Added Retailer (VAR) having installation and operational responsibilities. This configuration guide provides steps for configuring SIP Trunk to a Unify OpenScape Voice IP-PBX.

1.1 tekVizion Labs

tekVizion Labs™ is an independent testing and Verification facility offered by tekVizion PVS, Inc. ("tekVizion"). tekVizion Labs offers several types of testing services including:

- Remote Testing – provides secure, remote access to certain products in tekVizion Labs for pre-Verification and ad hoc testing
- Verification Testing – Verification of interoperability performed on-site at tekVizion Labs between two products or in a multi-vendor configuration
- Product Assessment – independent assessment and verification of product functionality, interface usability, assessment of differentiating features as well as suggestions for added functionality, stress and performance testing, etc.

tekVizion is a systems integrator specifically dedicated to the telecommunications industry. Our core services include consulting/solution design, interoperability/Verification testing, integration, custom software development and solution support services. Our services helps service providers achieve a smooth transition to packet-voice networks, speeding delivery of integrated services. While we have expertise covering a wide range of technologies, we have extensive experience surrounding our practice areas which include: SIP Trunking, Packet Voice, Service Delivery, and Integrated Services.

The tekVizion team brings together experience from the leading service providers and vendors in telecom. Our unique expertise includes legacy switching services and platforms, and unparalleled product knowledge, interoperability and integration experience on a vast array of VoIP and other next-generation products. We rely on this combined experience to do what we do best: help our clients advance the rollout of services that excite customers and result in new revenues for the bottom line. tekVizion leverages this real-world, multi-vendor integration and test experience and proven processes to offer services to vendors, network operators, enhanced service providers, large enterprises and other professional services firms. tekVizion's headquarters, along with a state-of-the-art test lab and Executive Briefing Center, is located in Plano, Texas.

For more information on tekVizion and its practice areas, please visit tekVizion's website at www.tekVizion.com

2 SIP Trunking Network Components

The network for the SIP trunk reference configuration is illustrated below and is representative of a Unify OpenScape Voice IP-PBX configuration.

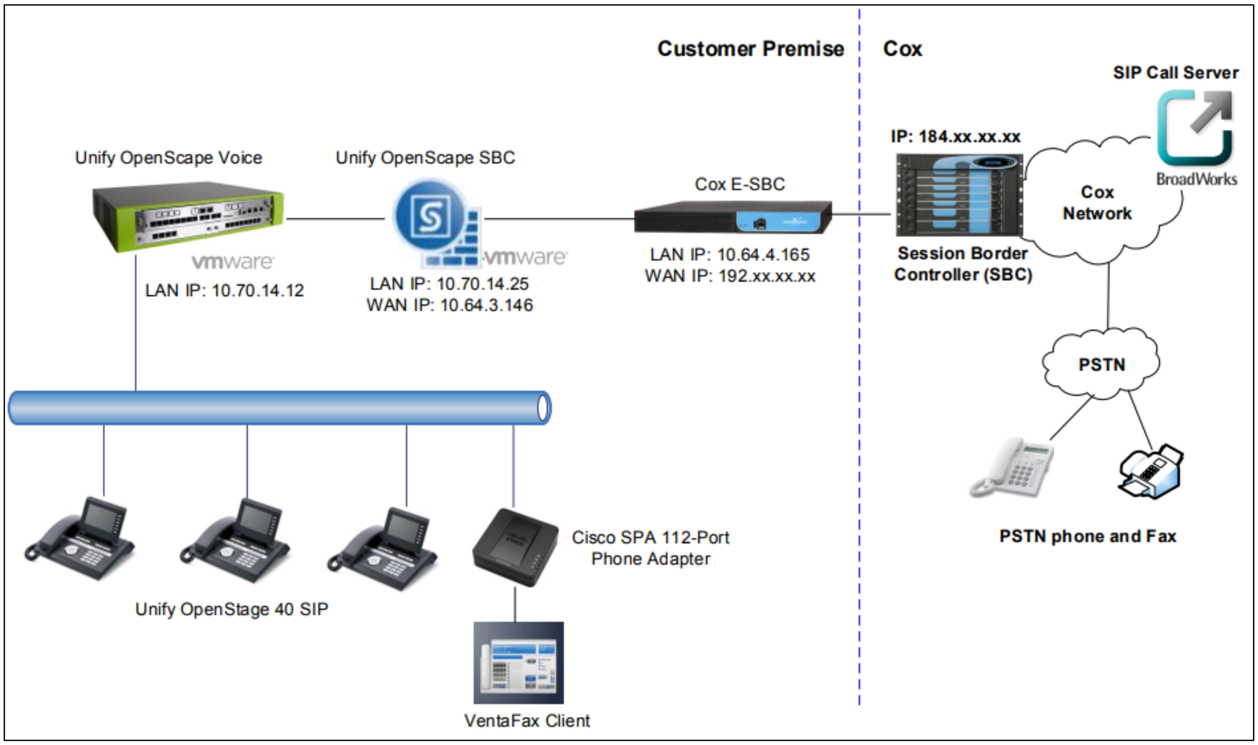

Figure 1: Topology Diagram

2.1 Hardware Components

- Unify OpenScape Voice
- Unify OpenScape Voice SBC

2.2 Software Requirements

- Unify OpenScape Voice Version 9 R4.41.0
- Unify SBC Version 9 R4.11.00

2.3 Features

2.3.1 Features Supported

- Basic Inbound and Outbound calls using G711ulaw
- Long duration call
- Session Timer
- Call Forwarding
- Call Transfer
- Call Hold and Resume
- Three-Way Calling
- RFC2833
- Fax
    - T.38
    - G711u Law Fax Pass-through

2.3.2 Features Not Supported

- Service Provider Voicemail
- SG3 fax speeds are not supported
- G729, G722
- Inband DTMF

2.3.3 Features Not Tested

- ONS Feature
- Calls between Branches
- Voice Mail

2.3.4 Caveats and Limitations

- COX does not provide Voice Mail service
- INVITE from EdgeMarc removes P-Asserted-Identity and Session Expires headers

3 Unify OSV Configuration

These are the main tasks to configure Unify OSV PBX to perform Direct SIP integration with the Unify SBC

3.1 Global Translation and Routing
3.2 Business Group
3.3 Private Numbering Plan List
3.4 Profiles Endpoint
3.5 Members Endpoints
3.6 Profiles Feature
3.7 Members Subscribers
3.8 Destinations and Routes: Destinations
3.9 Translation Prefix Access Codes
3.10 Translation Destination Codes

3.1 Global Translation and Routing

1. Click on the **Configuration** tab and **OpenScape Voice** tab. All OSV provisioning in this guide will be from the **Configuration** and **OpenScape Voice** tabs and will only be noted here.
2. Click on the **Global Translation** and **Routing ICON**.

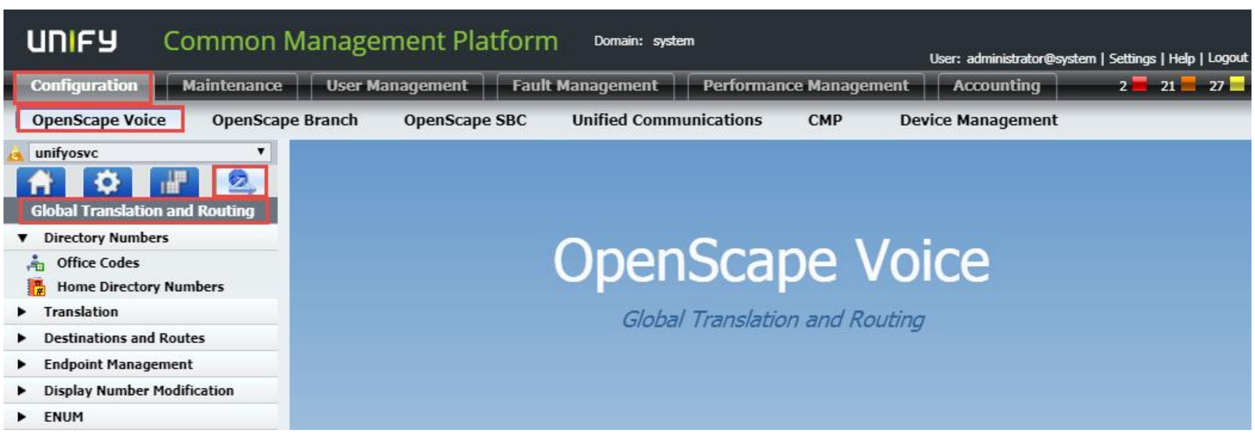

Figure 2 Global Translation and Routing

3. Click on **Directory Numbers** and **Office Codes**.
4. Click **Add** on the right side of screen, not shown here.

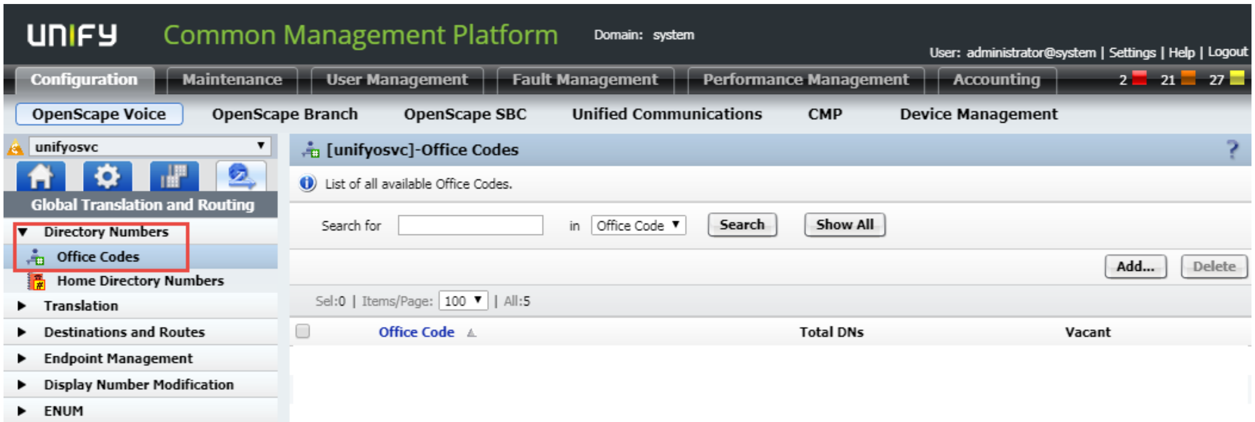

Figure 3 Office Codes

5. Area Code is set to 402 for this example.
6. **Local Office Code** is set to 932 for example.
7. **Directory Number start** is set to 2600 for example.
8. **Directory Number End** is set to 4970 for example.
9. Click **Save** at bottom right of page.

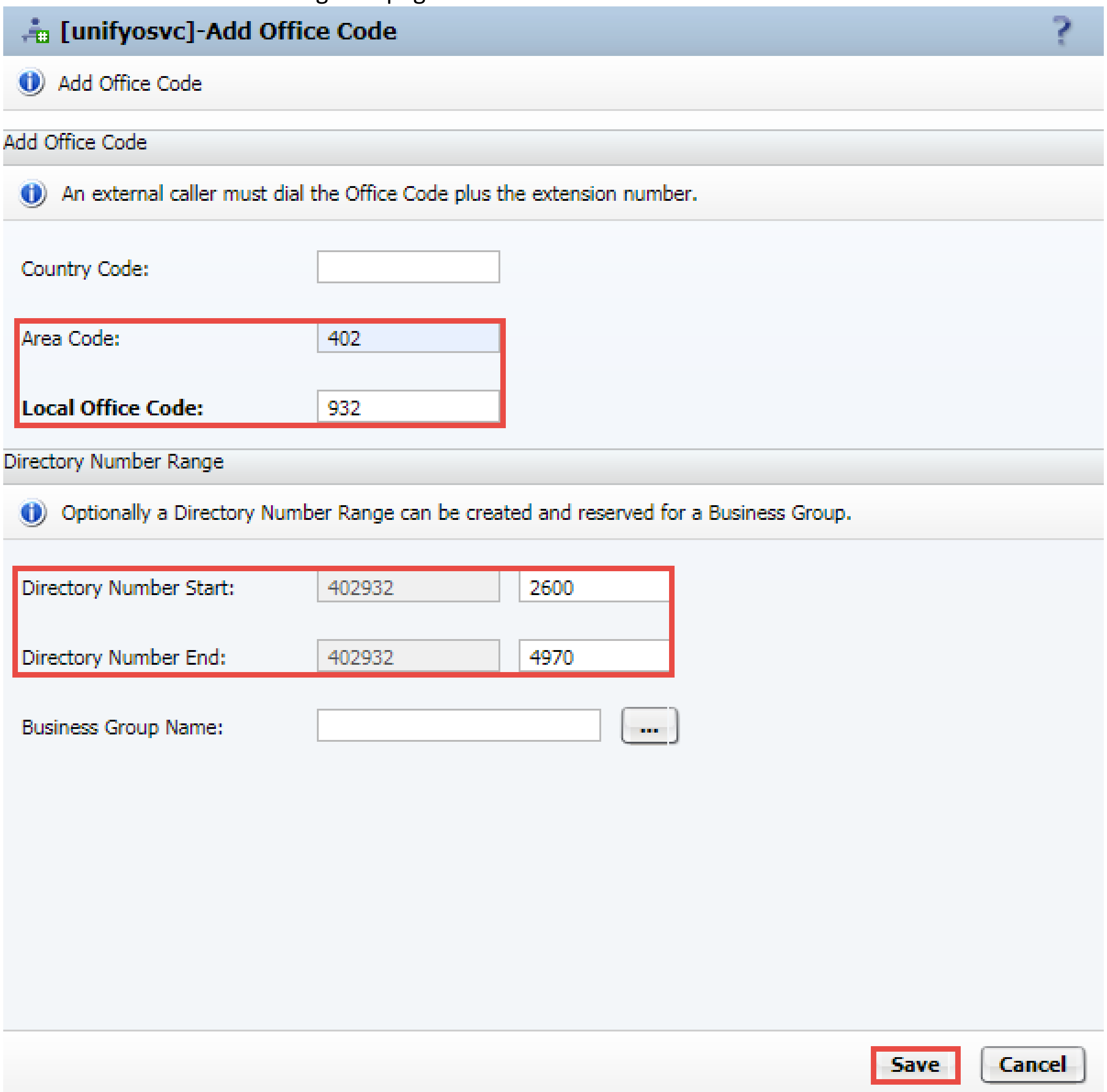

Figure 4 Office Codes-continued

3.2 Business Group

1. Click on the **Business Group ICON** and **Add**.

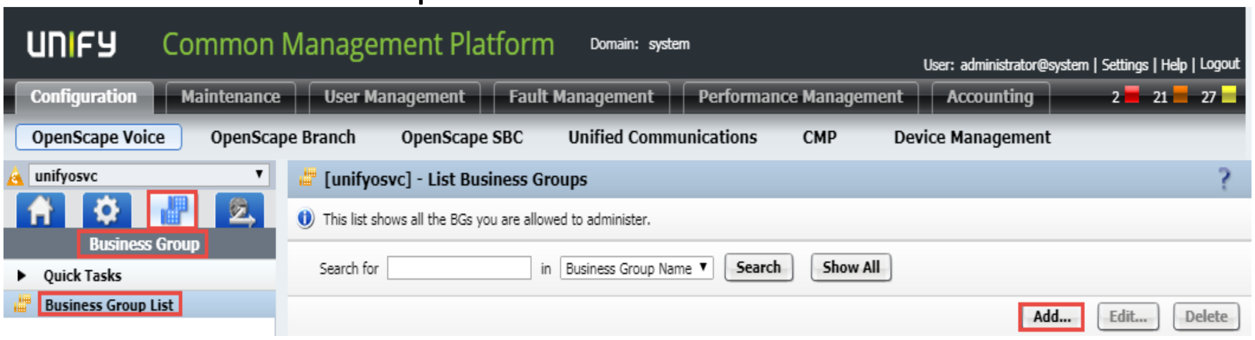

Figure 5 Business Group

2. In the **General** tab **Business Group name** is set to COX_BG for example.
3. **Default Numbering Plan** defaulted to NP_COX_BG for example. The default is based upon the Business Group name previously entered.
4. **Display Number** is set to 4029324962 for example.
5. **Default Office Code** is set to () 402932 for example using the drop down button.
6. Click **Save** at bottom right of page.

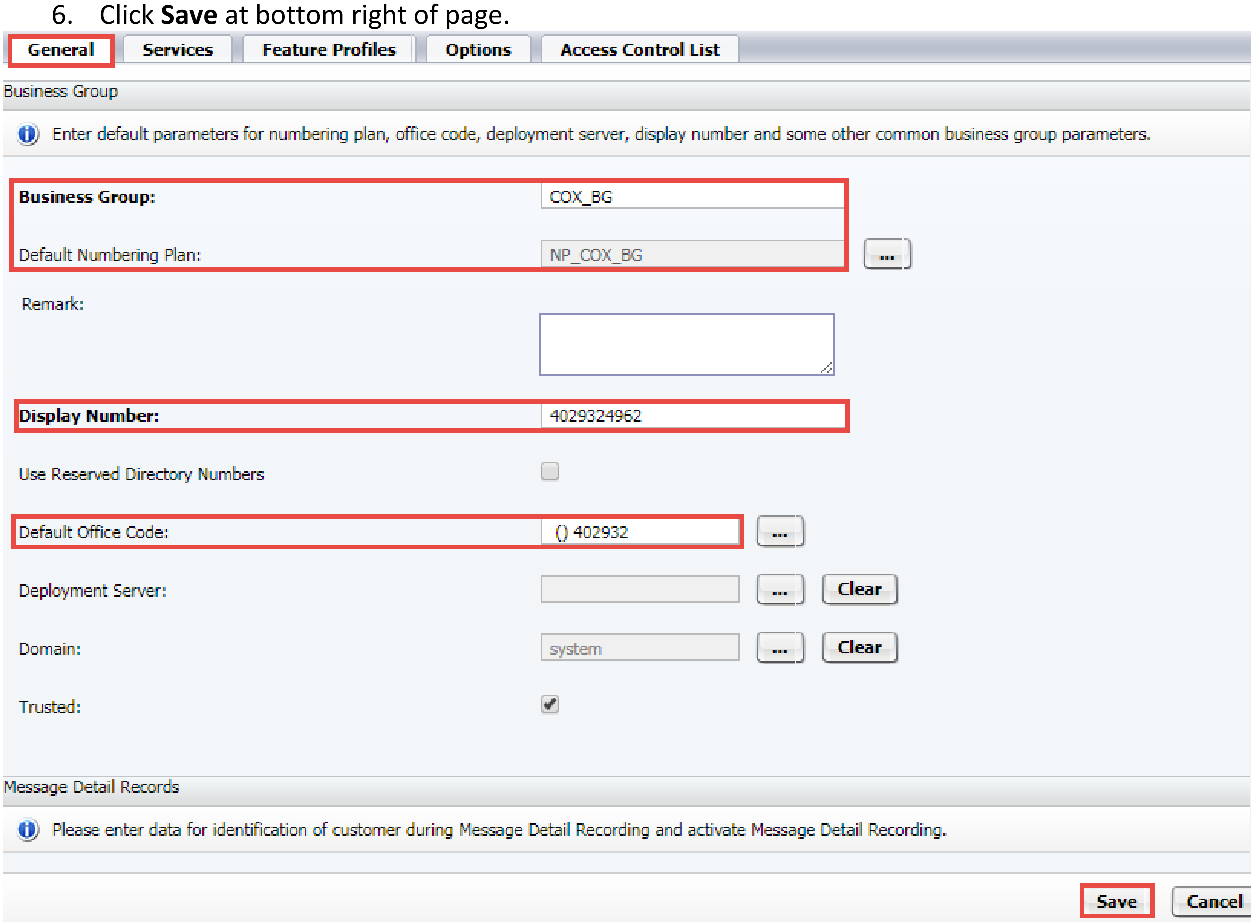

Figure 6 Business Group-continued

7. Click on the **Features Profile** tab and click **Add**.

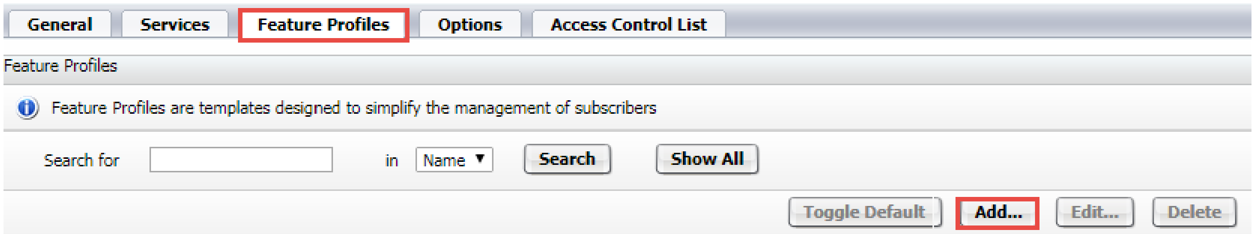

Figure 7 Business Group-continued

8. In the **General** tab **Name** is set to FP_COX for example.
9. **Default** is checked.

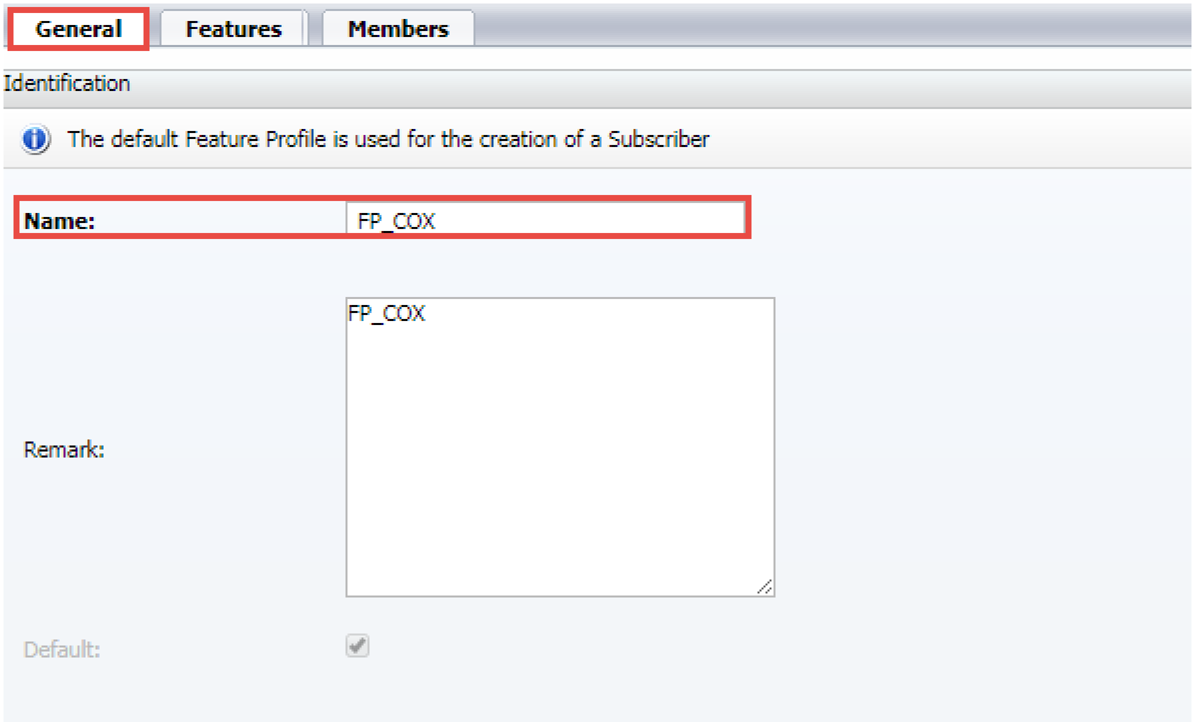

Figure 8 Business Group-continued

10. Click on the **Features** tab.
11. Click into the **Feature Name** field.
12. Check all required features per site specific requirements.
13. Click **Add**.
14. Click **Save**.

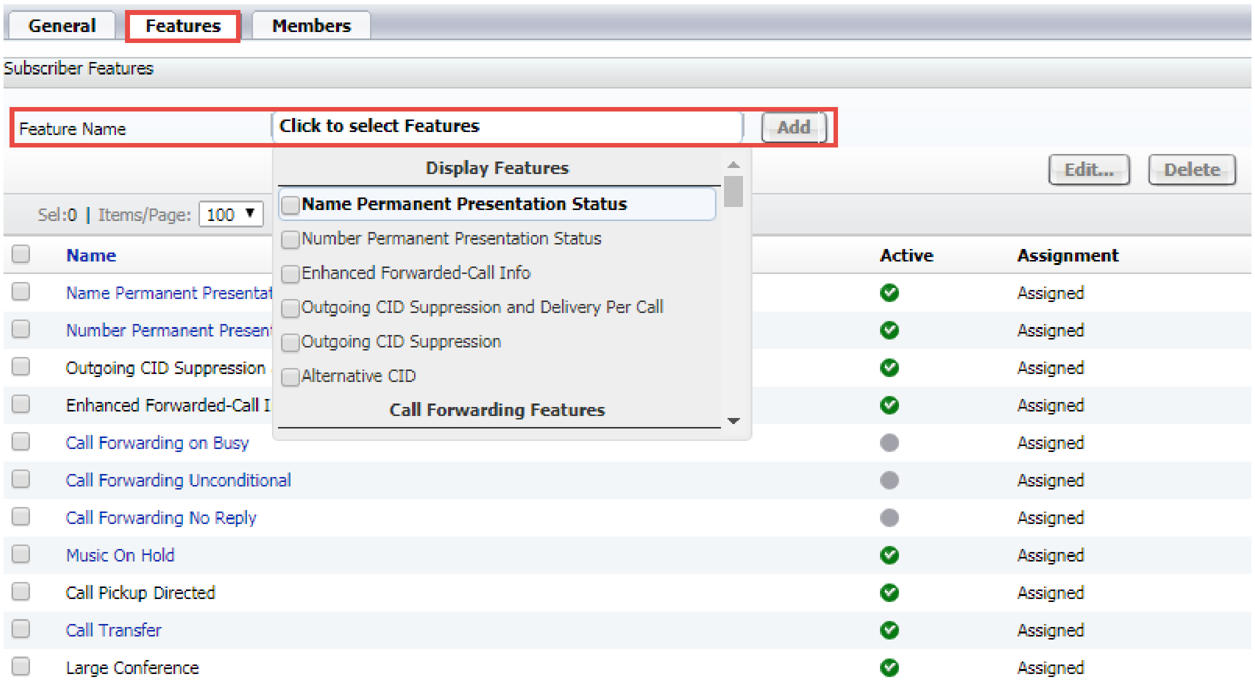

Figure 9 Business Group-continued

3.3 Private Numbering Plan List

1. Click on **Members** and **Private Numbering Plan List**.
2. Click **Add**.

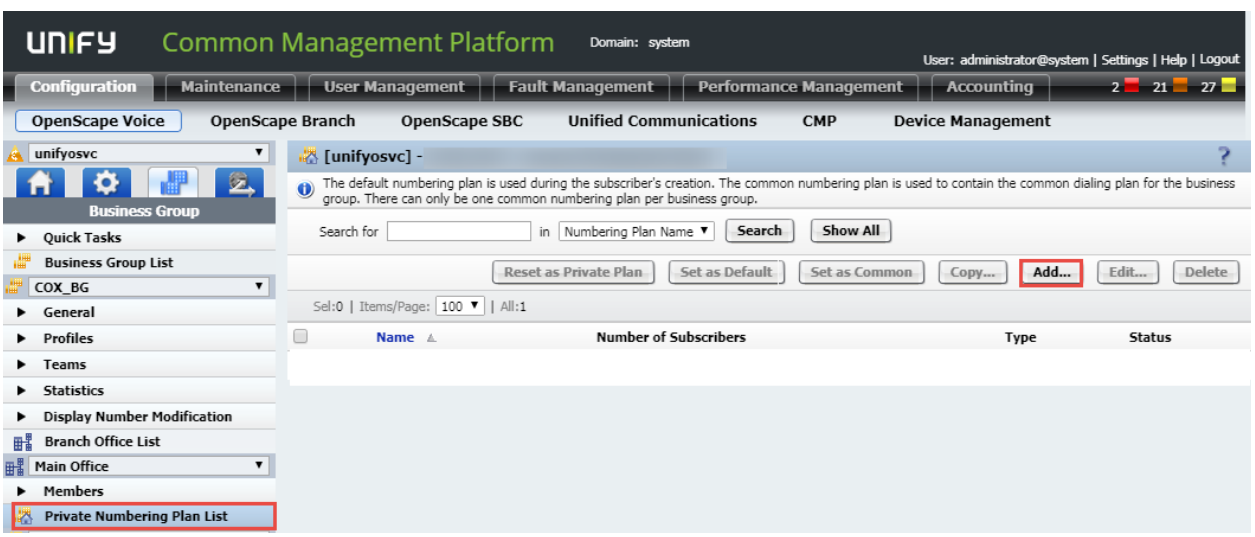

Figure 10 Private Numbering Plan

3. **Name** is set to NP_COX_BG for example.
4. Click **Save**.

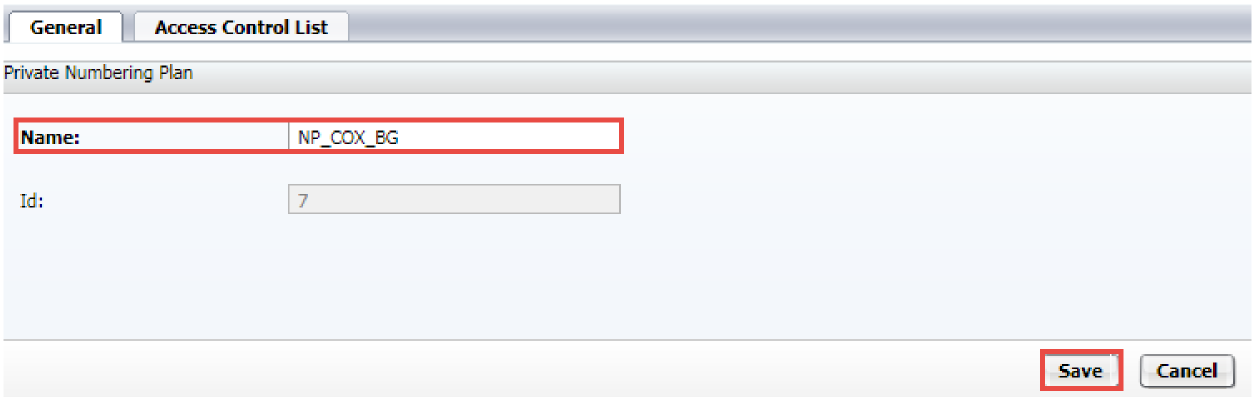

Figure 11 Private Numbering Plan-continued

3.4 Profiles Endpoint

1. Under **Business Group** List, COX_BG is selected for example. The **Business Group** will be **COX_BG** for all OSV provisioning throughout this guide and will only be noted here.
2. Click on **Profiles** and **Endpoint**.
3. Click on **Add**.

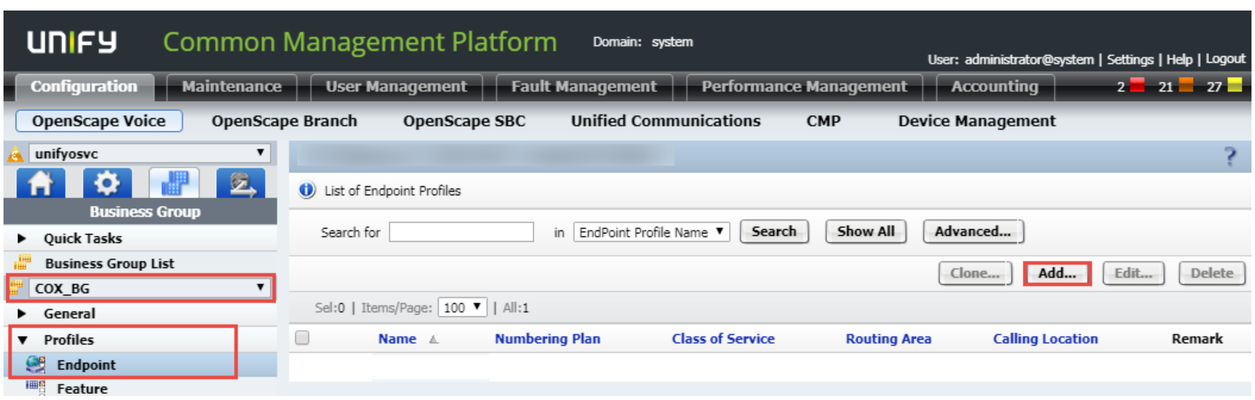

Figure 12 Profiles Endpoint

4. **Name** is set to COX_EP for example.
5. **Numbering Plan** is set to NP_COX_BG for example using the drop down button.
6. **SIP Privacy Support** is set to **full** using the drop down arrow.

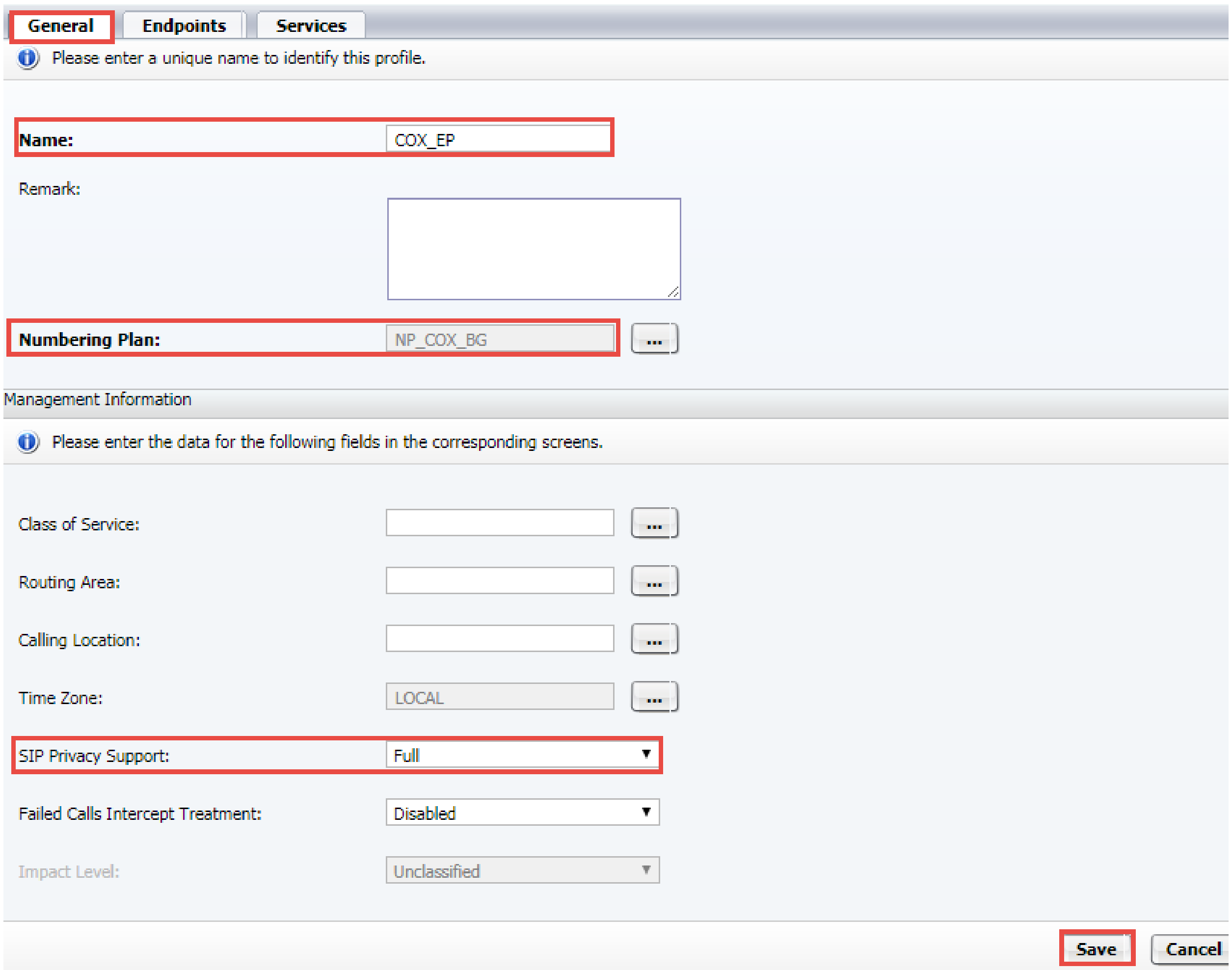

Figure 13 Profiles Endpoint-continued

7. Click on **Services** tab.
8. **Call Waiting** is set to Yes for example using the drop down arrow.
9. **Call Transfer** is set to Yes for example using the drop down arrow.
10. Click **Save**.

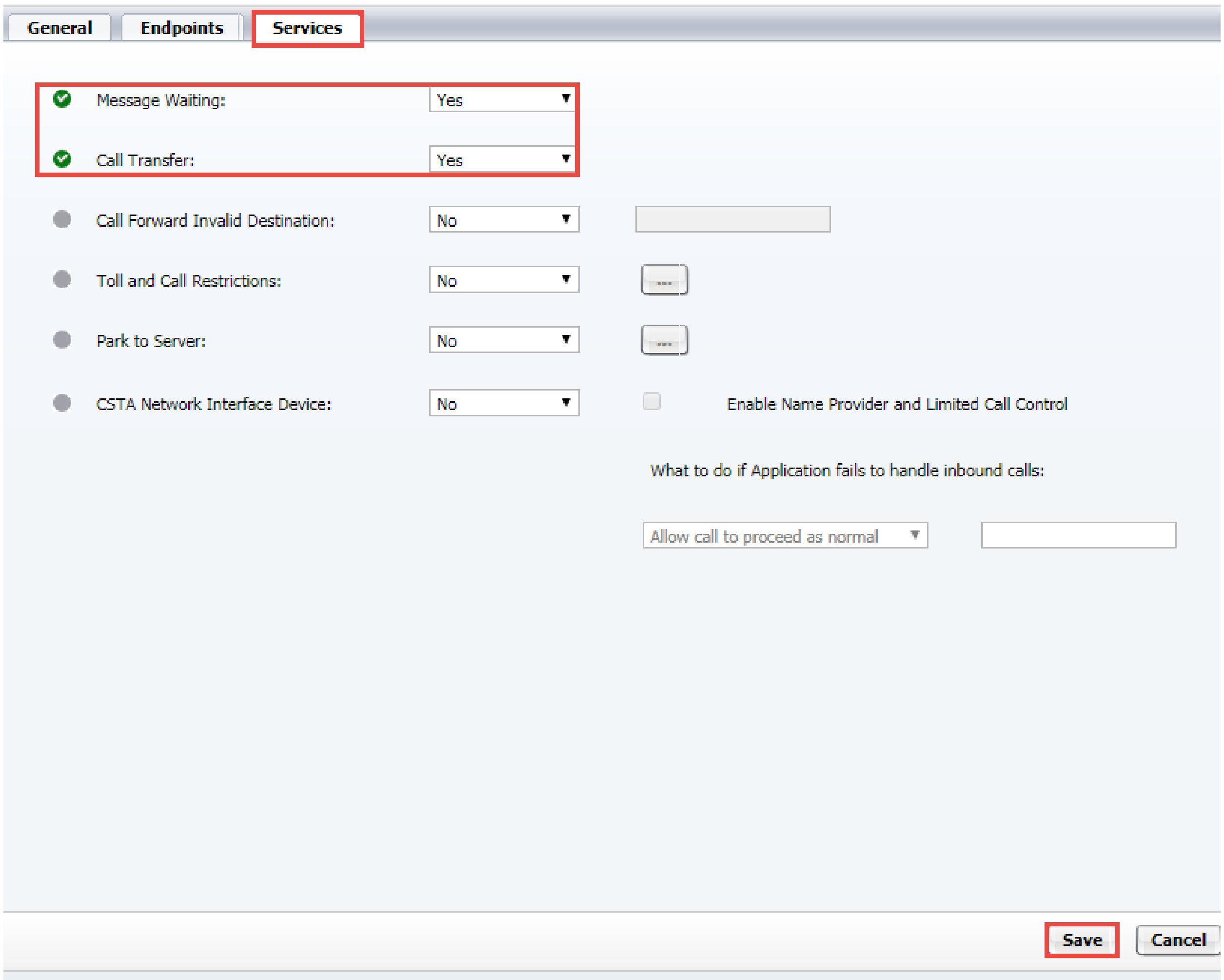

Figure 14 Profiles Endpoint-continued

3.5 Members Endpoints

1. Click on **Members** and **Endpoints**.
2. Click **Add**.

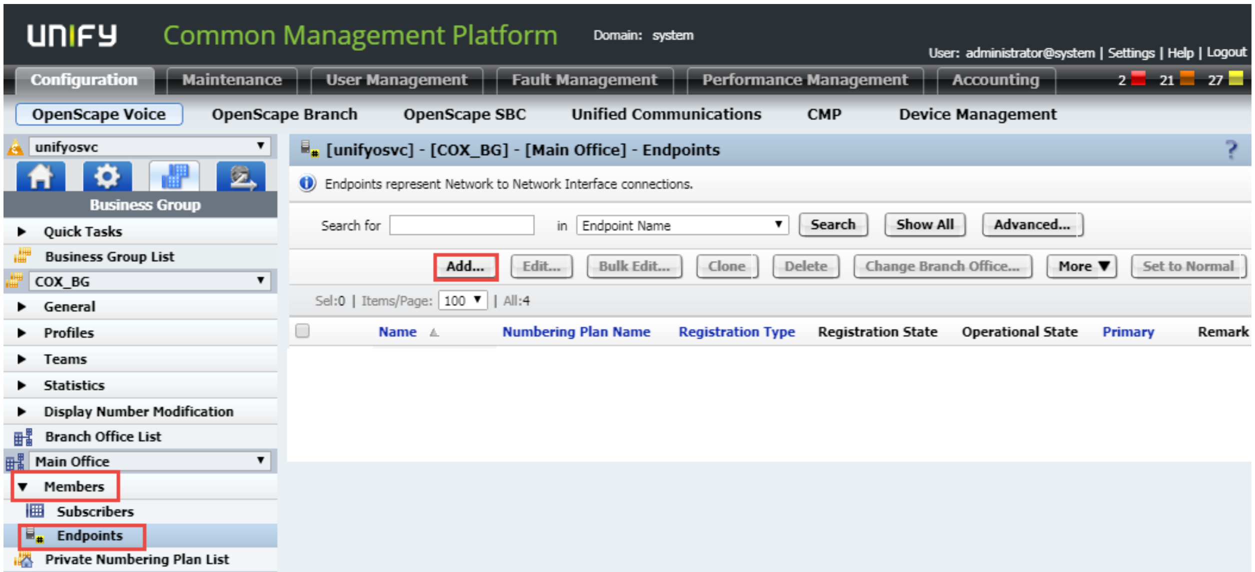

Figure 15 Members Endpoints

3. **Name** is set to COX_SBC for example.
4. **Registered** is checked.
5. **Profile** is set to COX_EP for example using the drop down button.

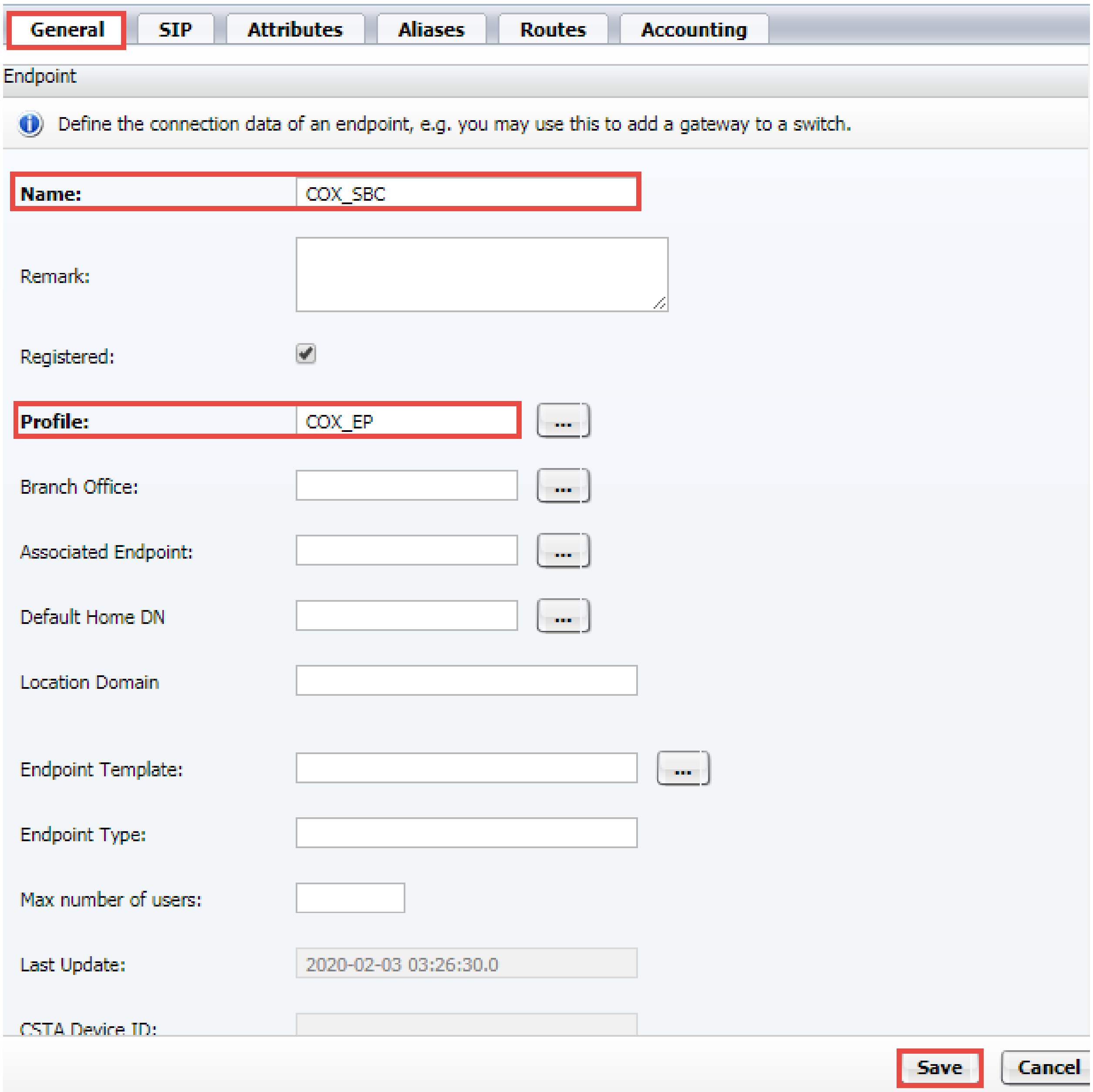

Figure 16 Members Endpoints-continued

6. Click on the **SIP** tab.
7. **SIP Trunking** radial button is selected.
8. **Type** is set to Static.
9. **Signaling Address Type** is set to IP Address or FQDN using the drop down arrow.

10. **Endpoint Address** is set to 10.70.14.25 for example. This is the Unify SBC's LAN IP address.
11. **Port** is set to 5060.
12. **Transport protocol** is set to TCP using the drop down arrow.

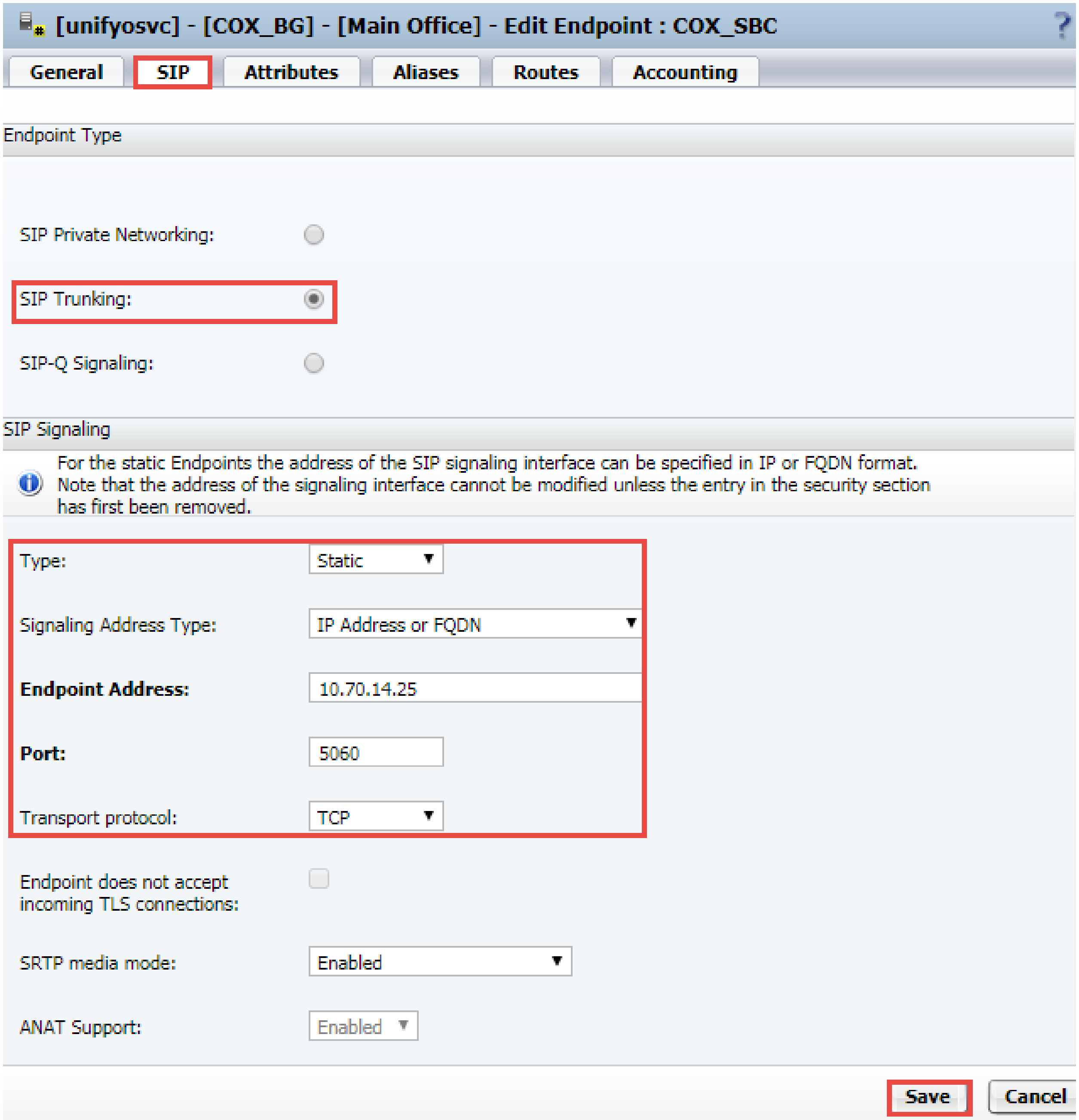

Figure 17 Members Endpoints-continued

13. Click on the **Attributes** tab.

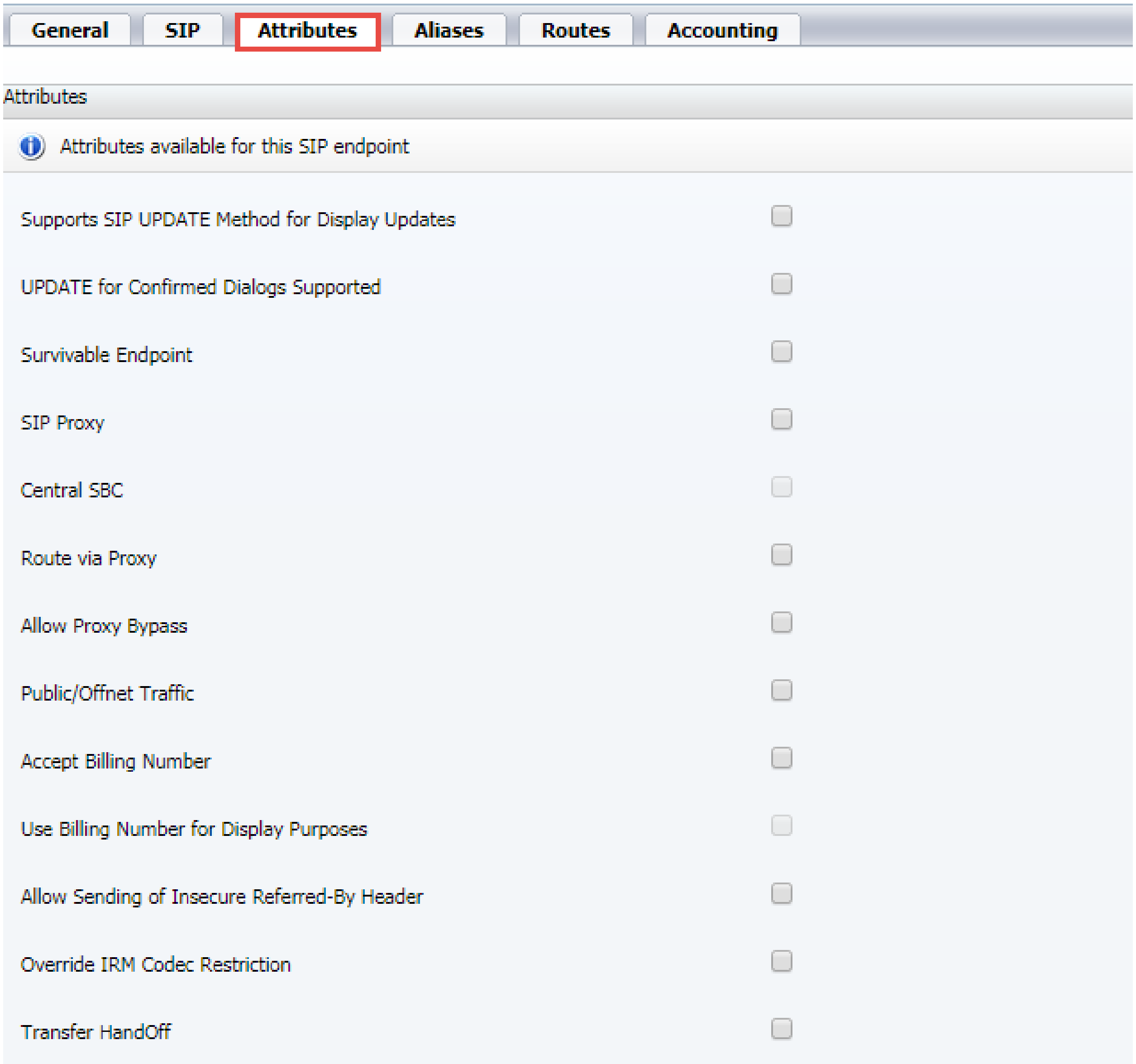

Figure 18 Members Endpoints-continued

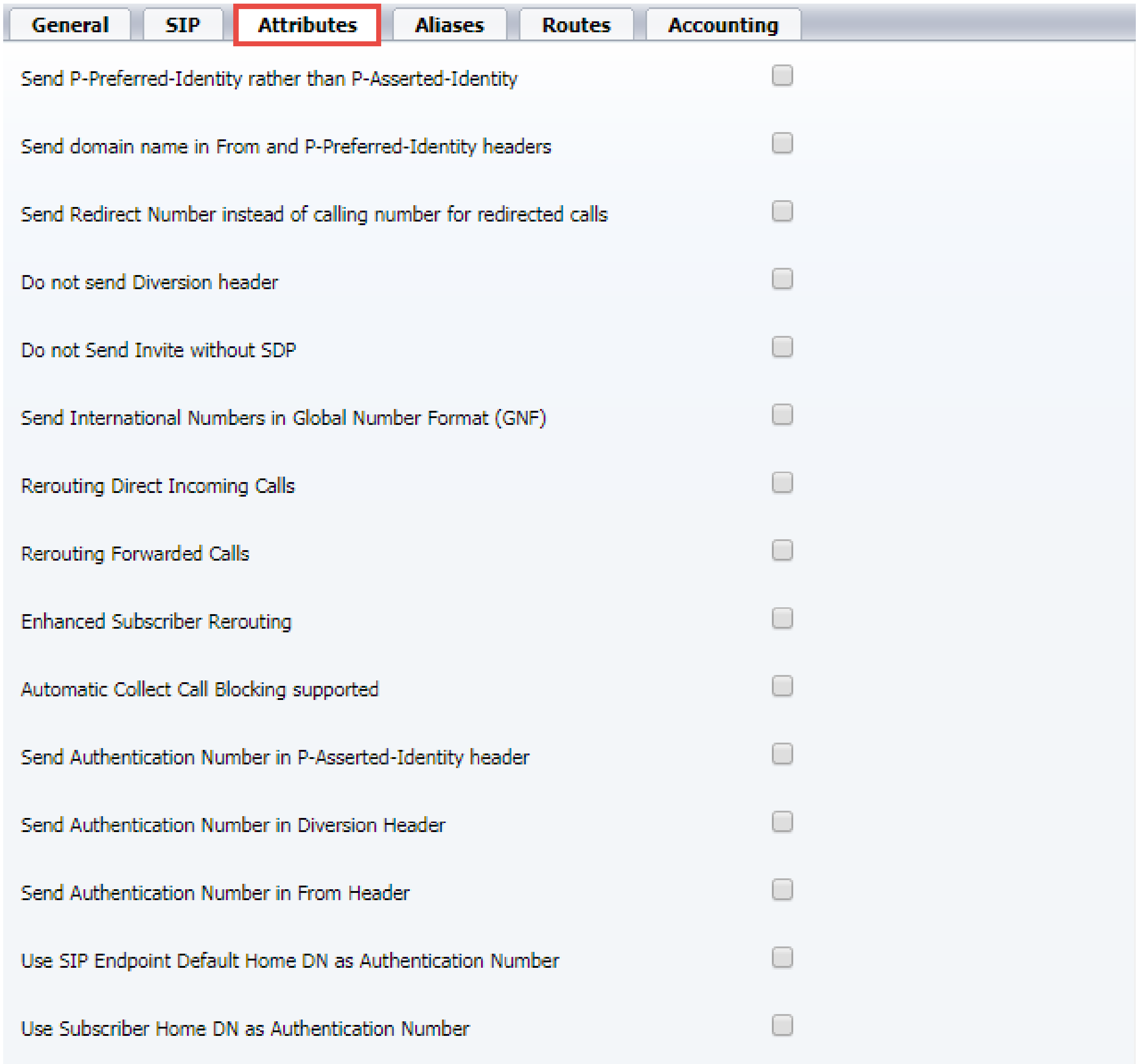

Figure 19 Members Endpoints-continued

14. **Enable Session Timer** is checked.

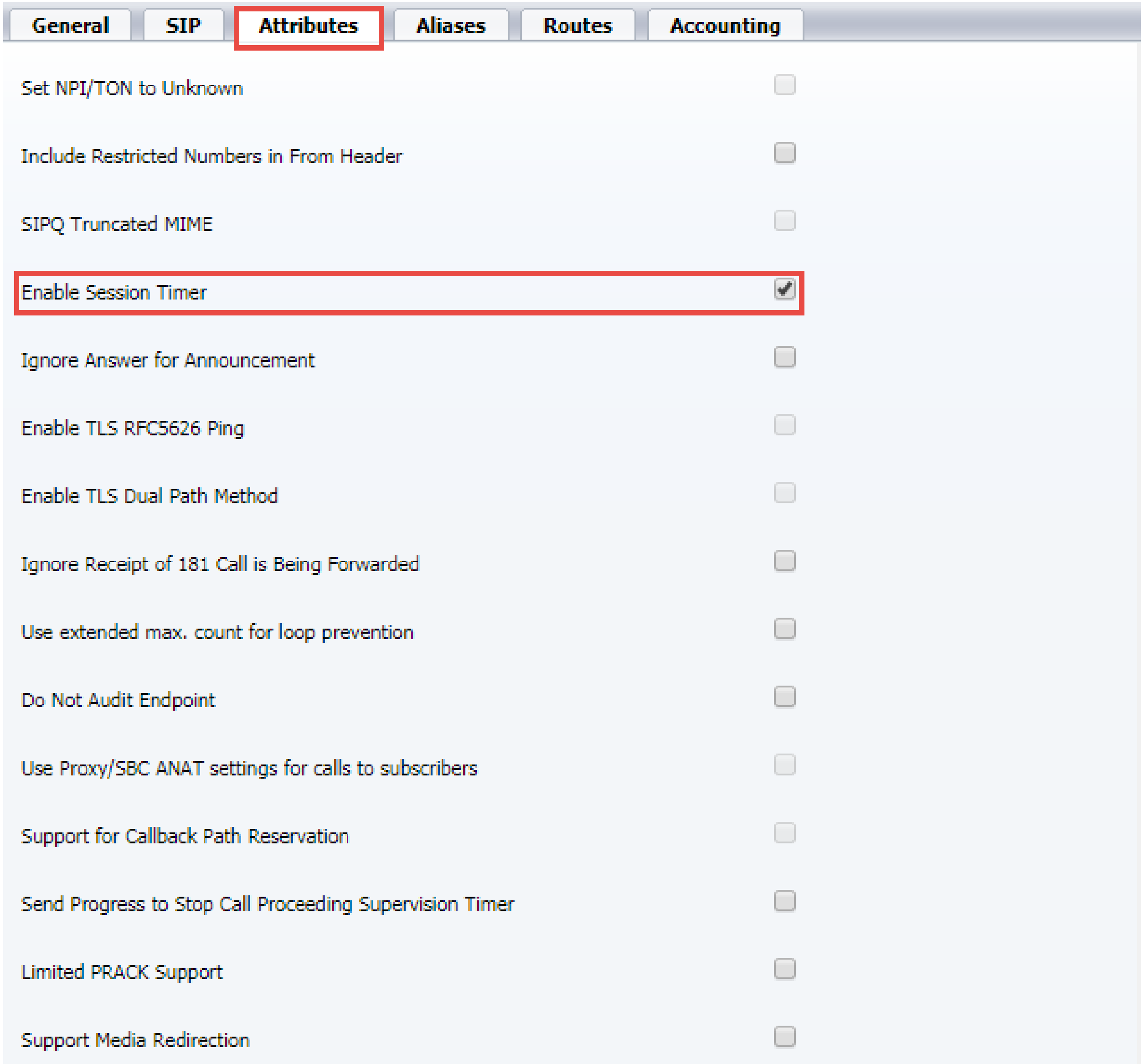

Figure 20 Members Endpoints-continued

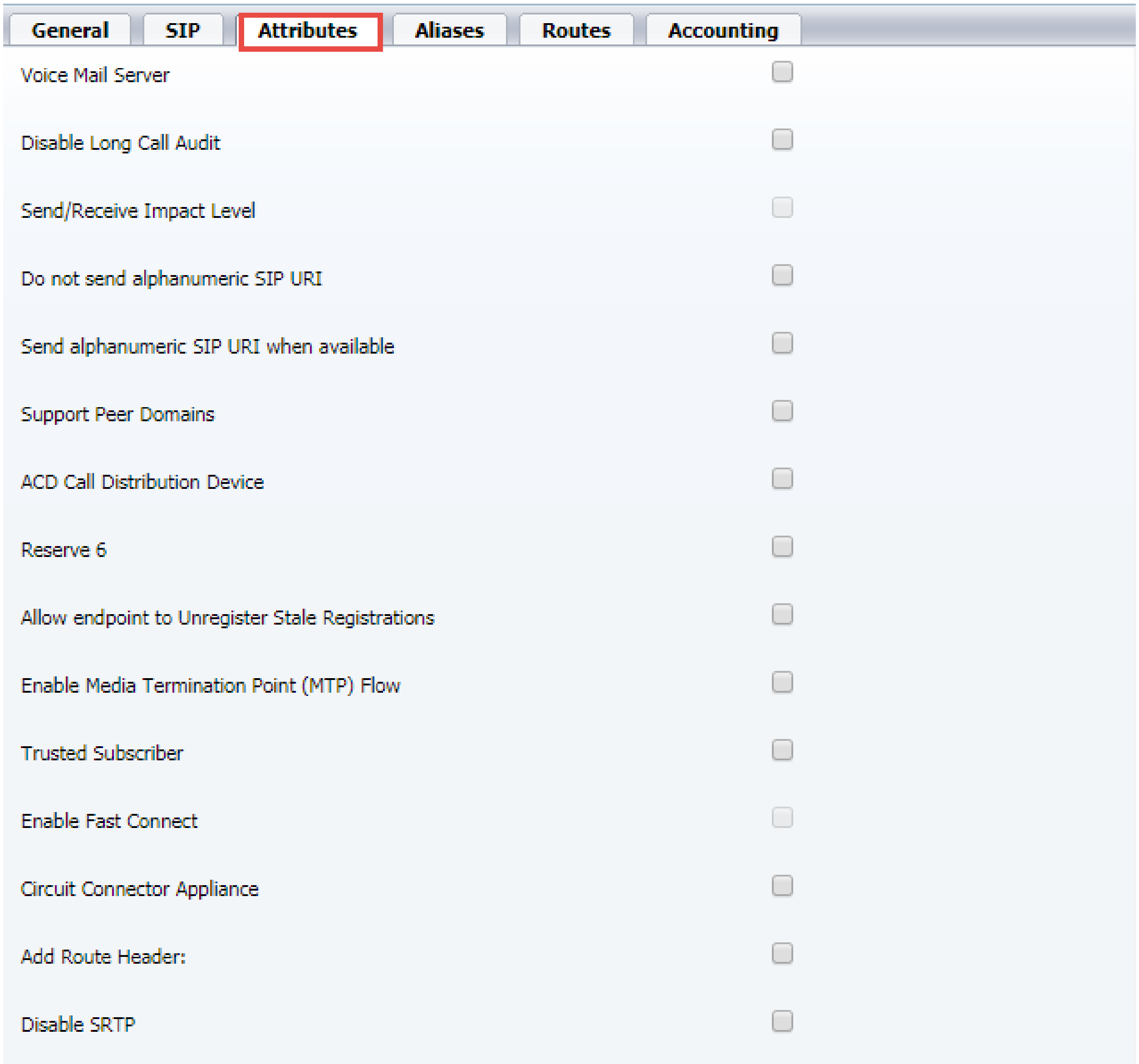

Figure 21 Members Endpoints-continued

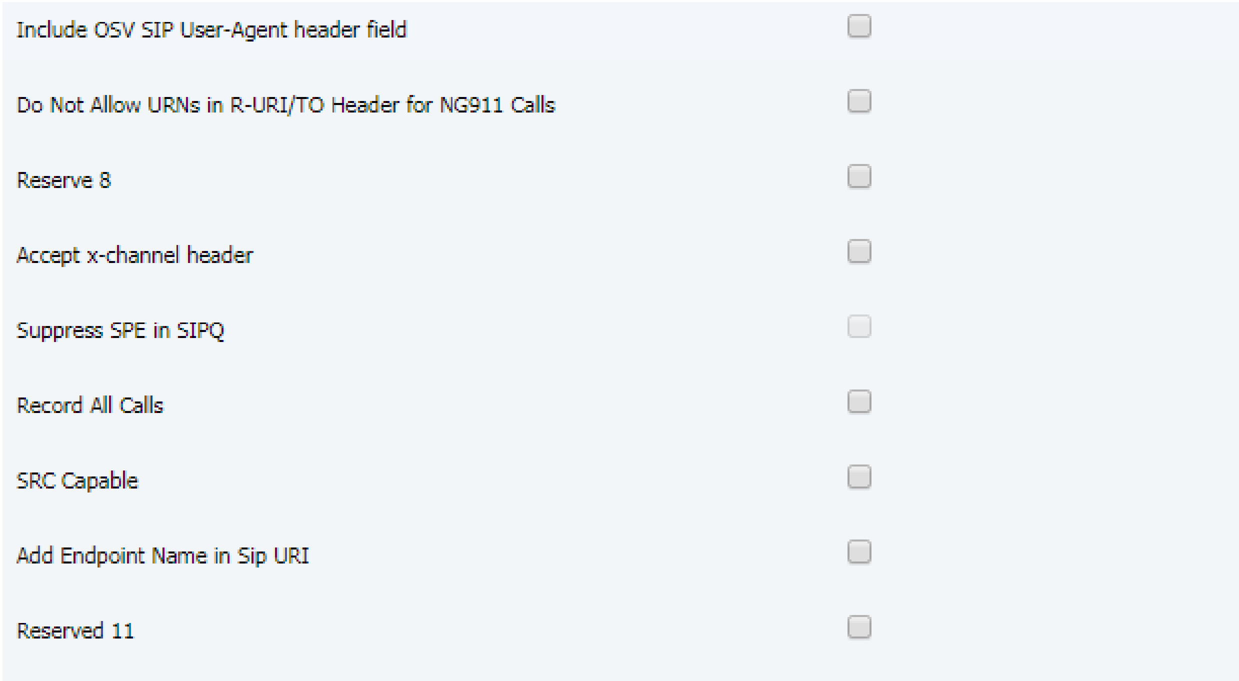

Figure 22 Members Endpoints-continued

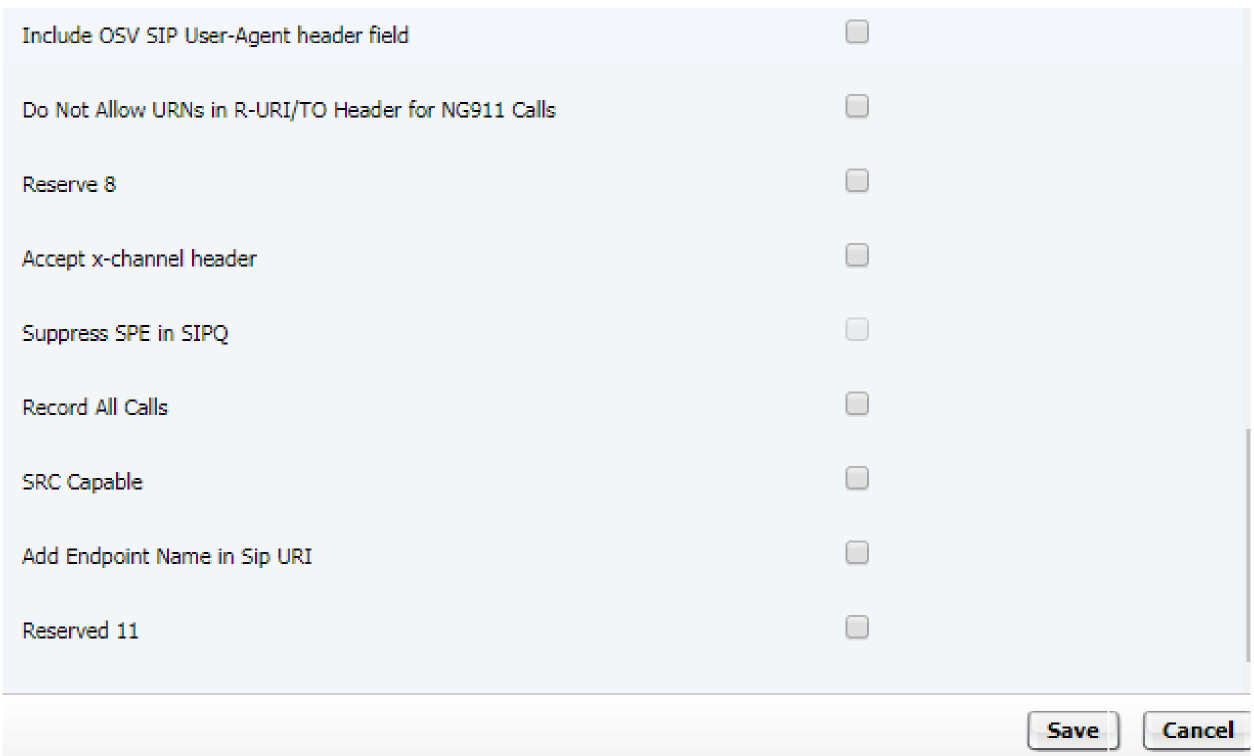

Figure 23 Members Endpoints-continued

15. Click the **Aliases** tab.
16. Click **Add**.
17. Set the name as 10.70.14.25:5060. This is the Openscape Voice SBC LAN IP address.

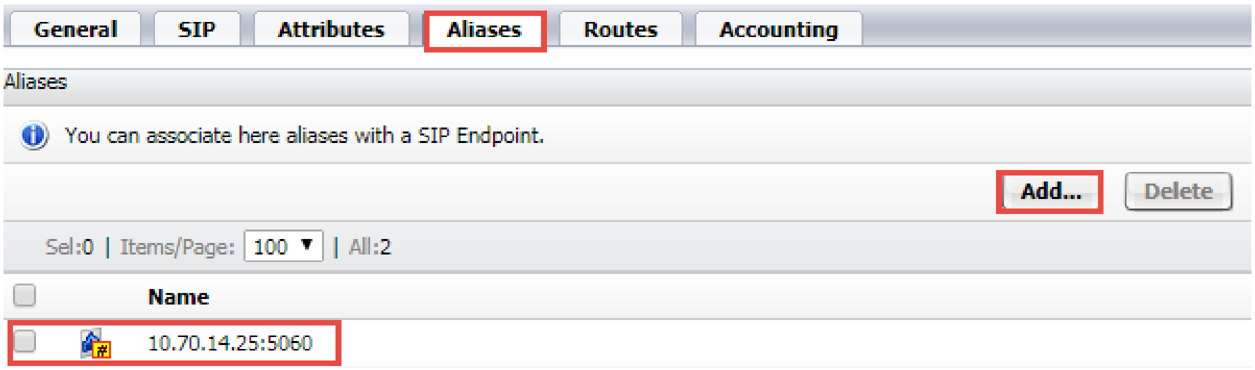

Figure 24 Members Endpoints-continued

18. **Name** is set to COX_SBC_Trunk for example.
19. **Registered** is checked.
20. **Profile** is set to COX_EP for example using the drop down button.

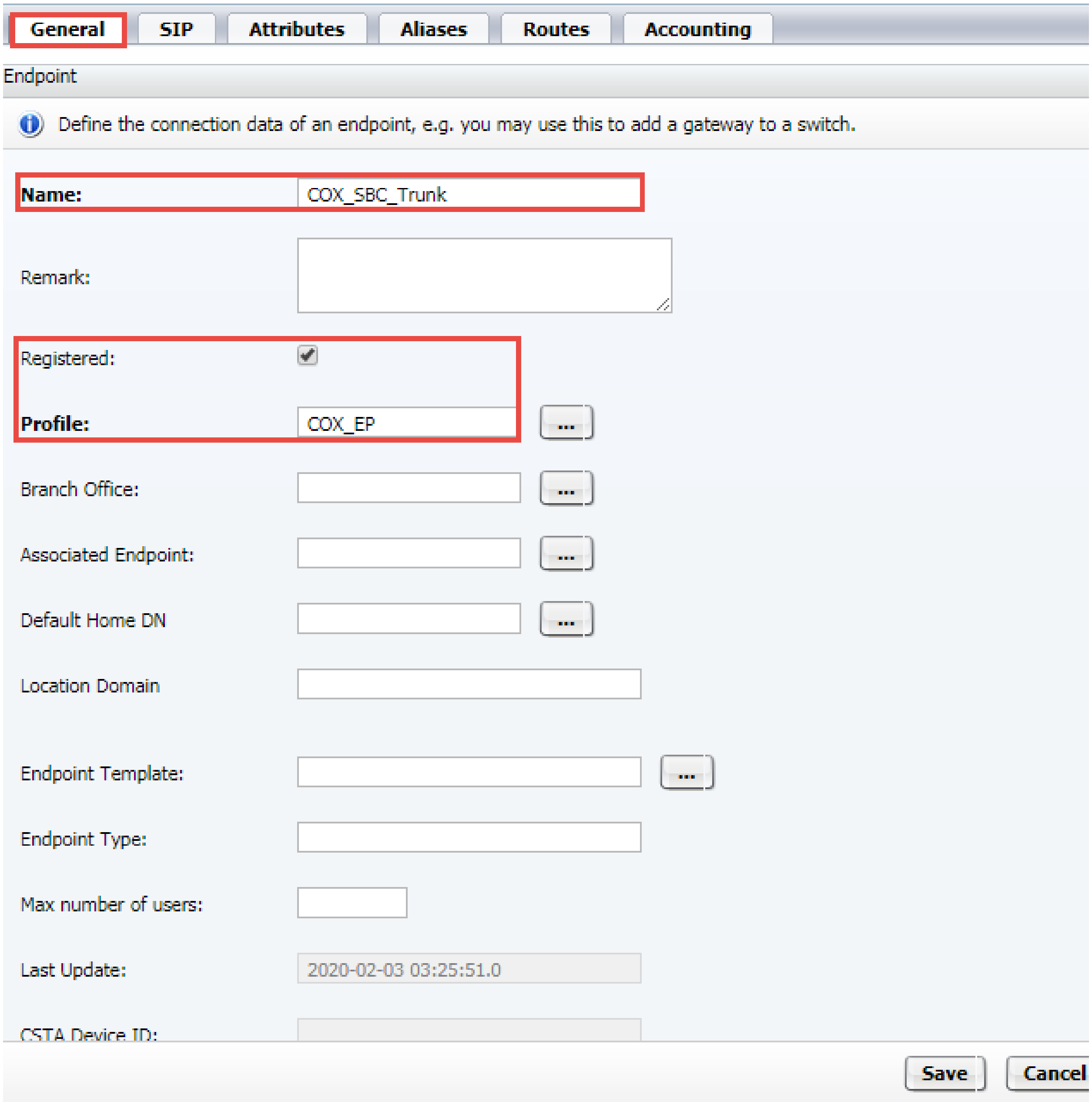

Figure 25 Members Endpoints-continued

21. Click on the **SIP** tab.
22. **SIP Trunking** button is selected.
23. **Type** is set to Static.
24. **Signaling Address Type** is set to IP Address or FQDN using the drop down arrow.
25. **Endpoint Address** is set to 10.70.14.25 for example. This is the Unify SBC's LAN IP address.
26. **Port** is set to 50012
27. **Transport protocol** is set to TCP using the drop down arrow.

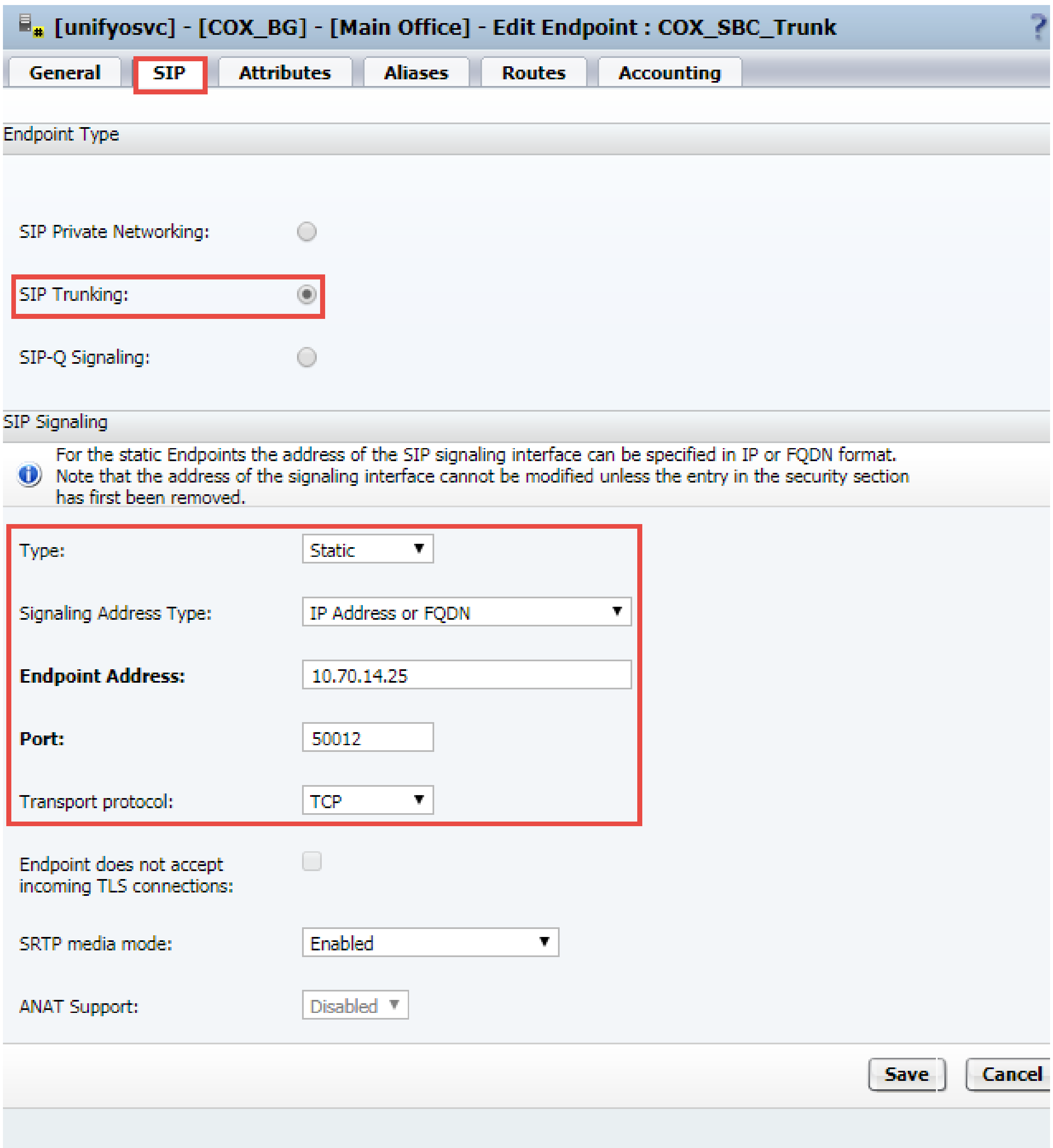

Figure 26 Members Endpoints-continued

28. Click on the **Attributes** tab.

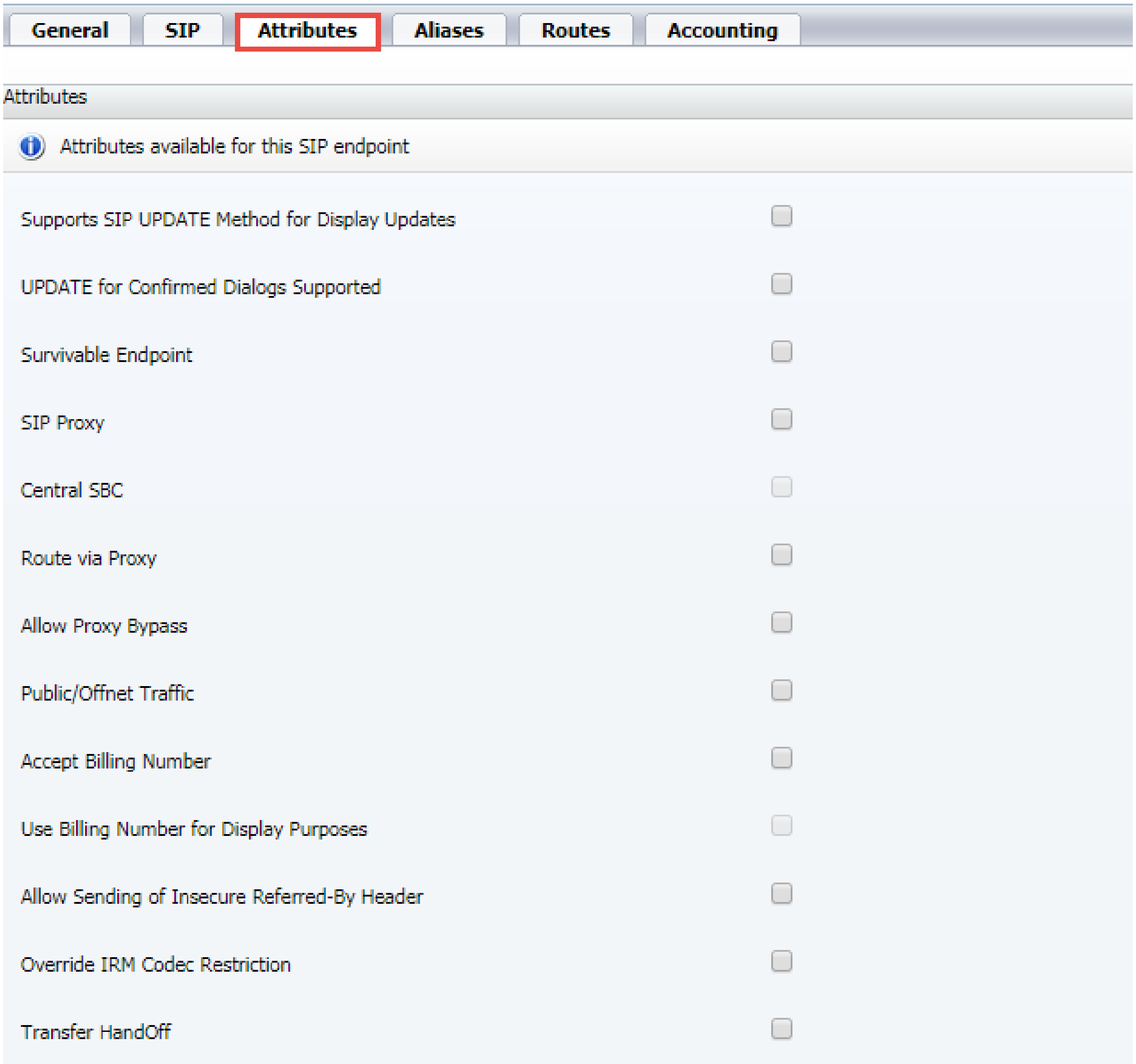

Figure 27 Members Endpoints-continued

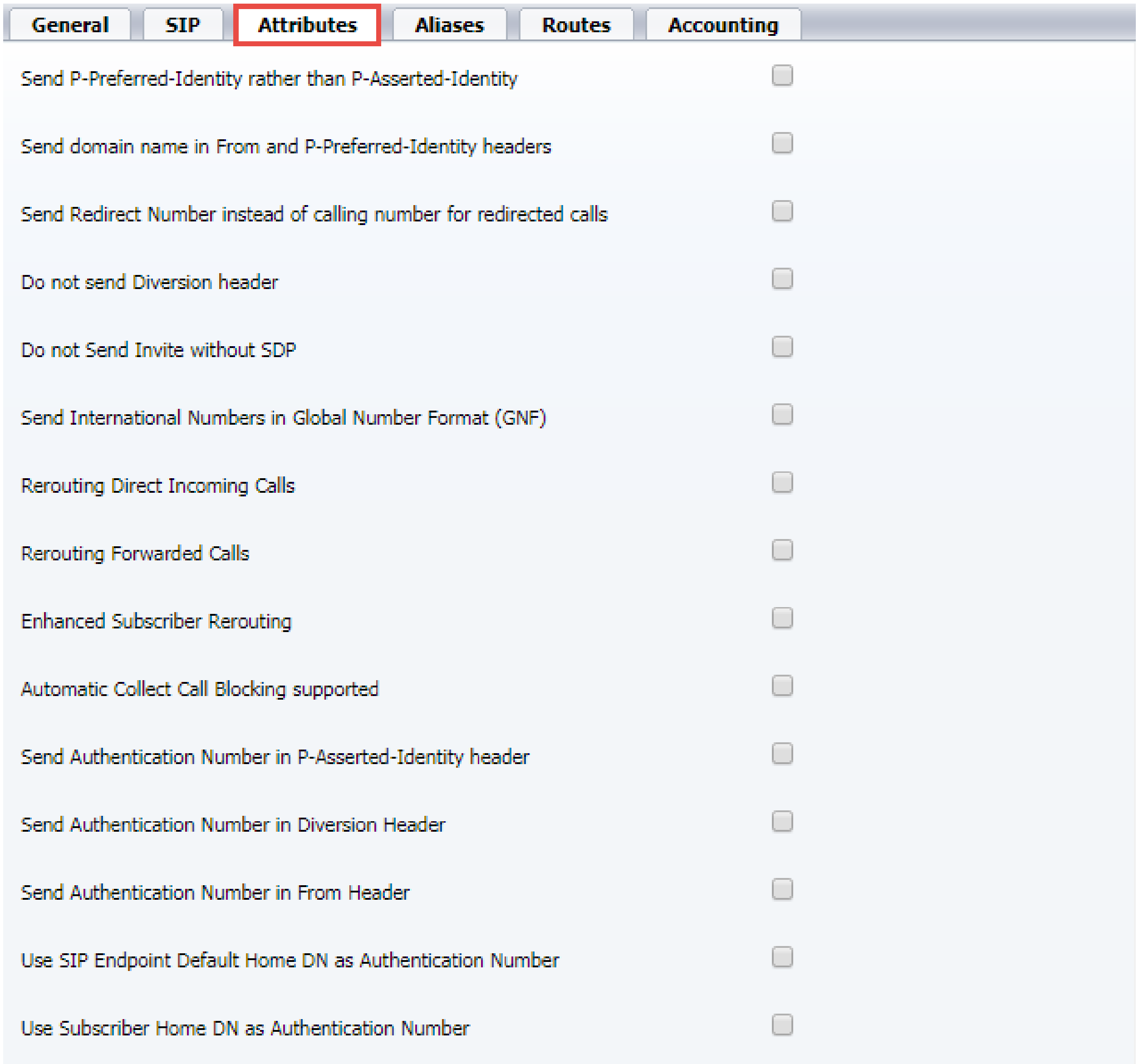

Figure 28 Members Endpoints-continued

29. **Enable Session timer** is checked.

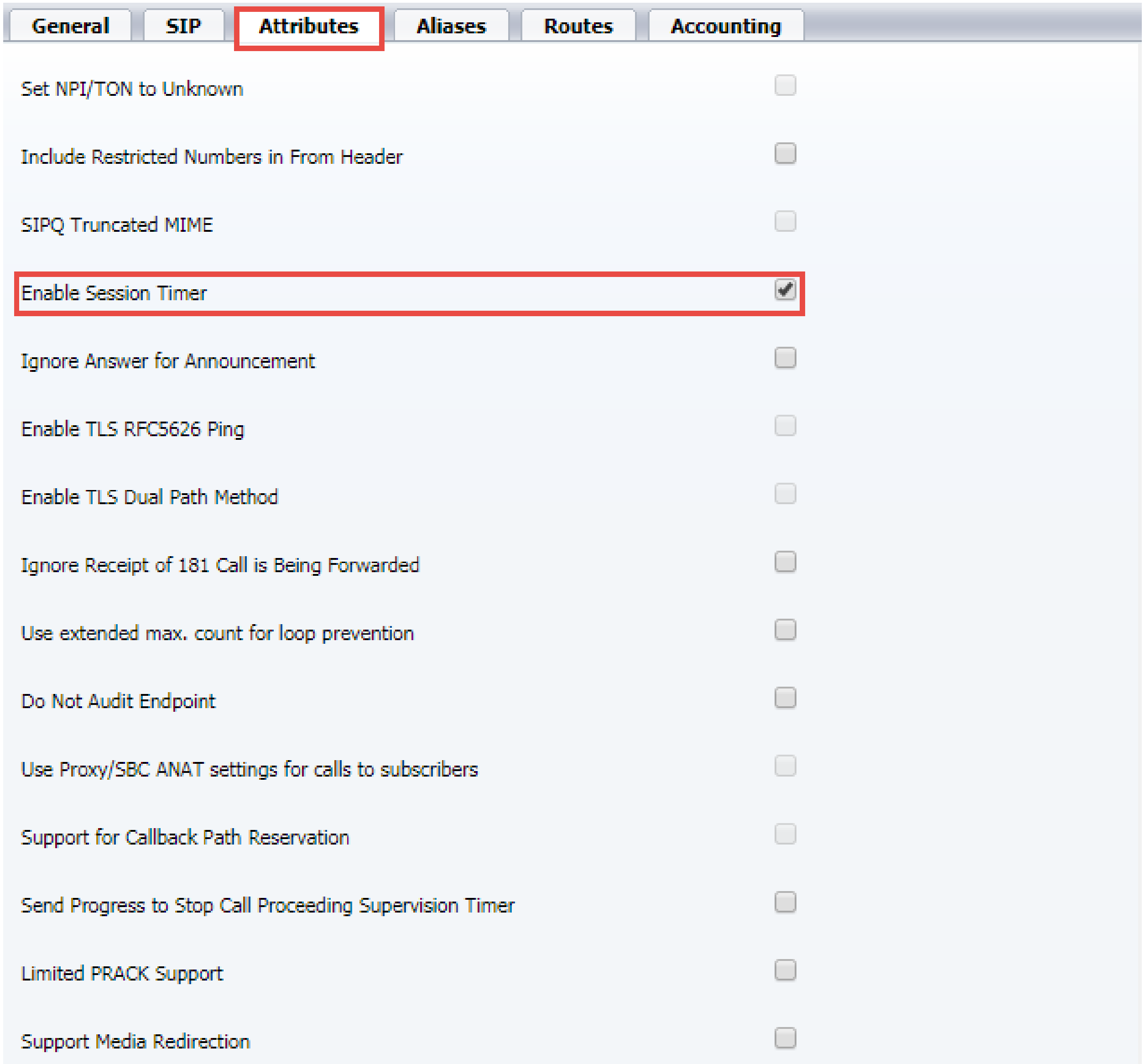

Figure 29 Members Endpoints-continued

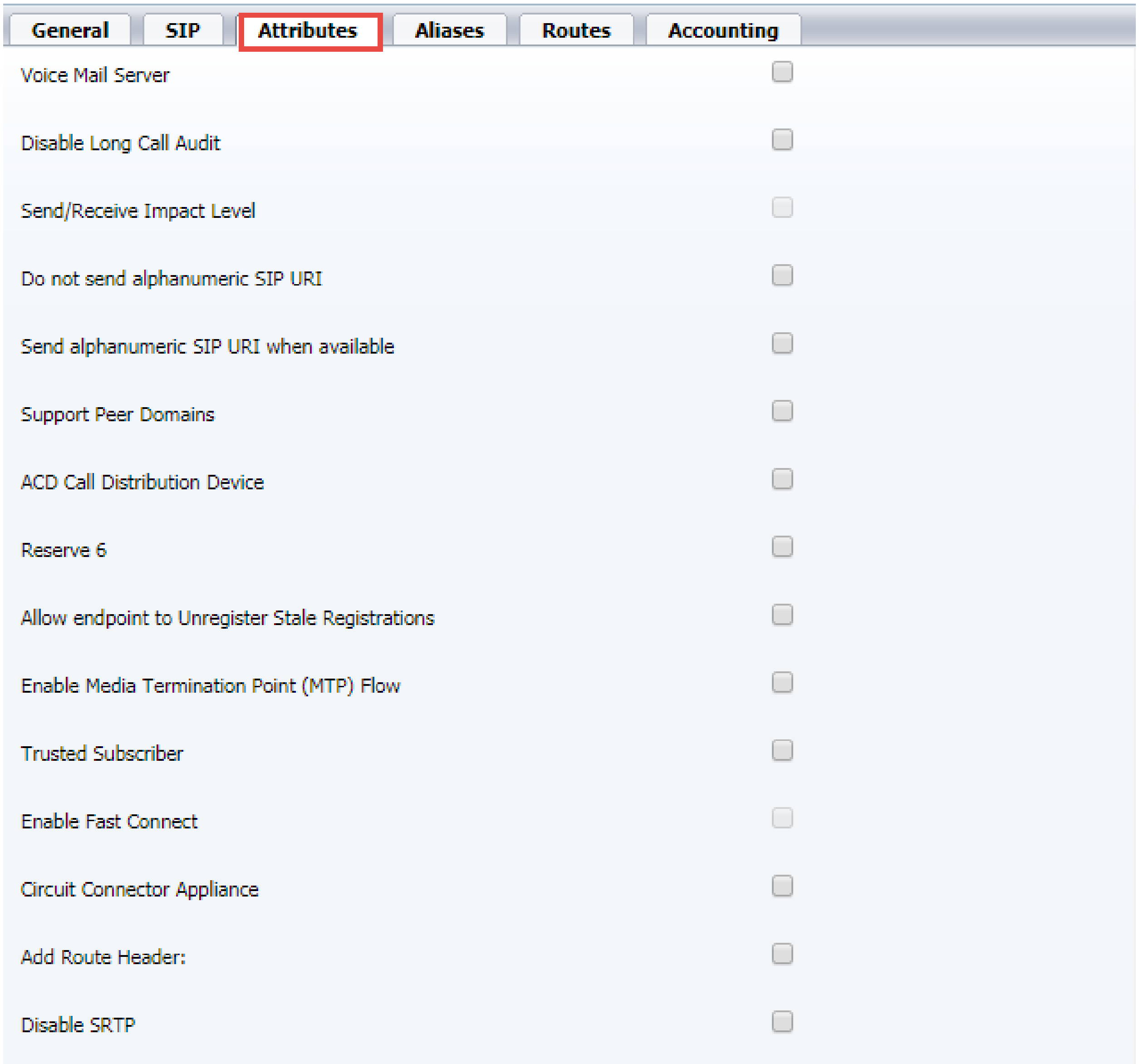

Figure 30 Members Endpoints-continued

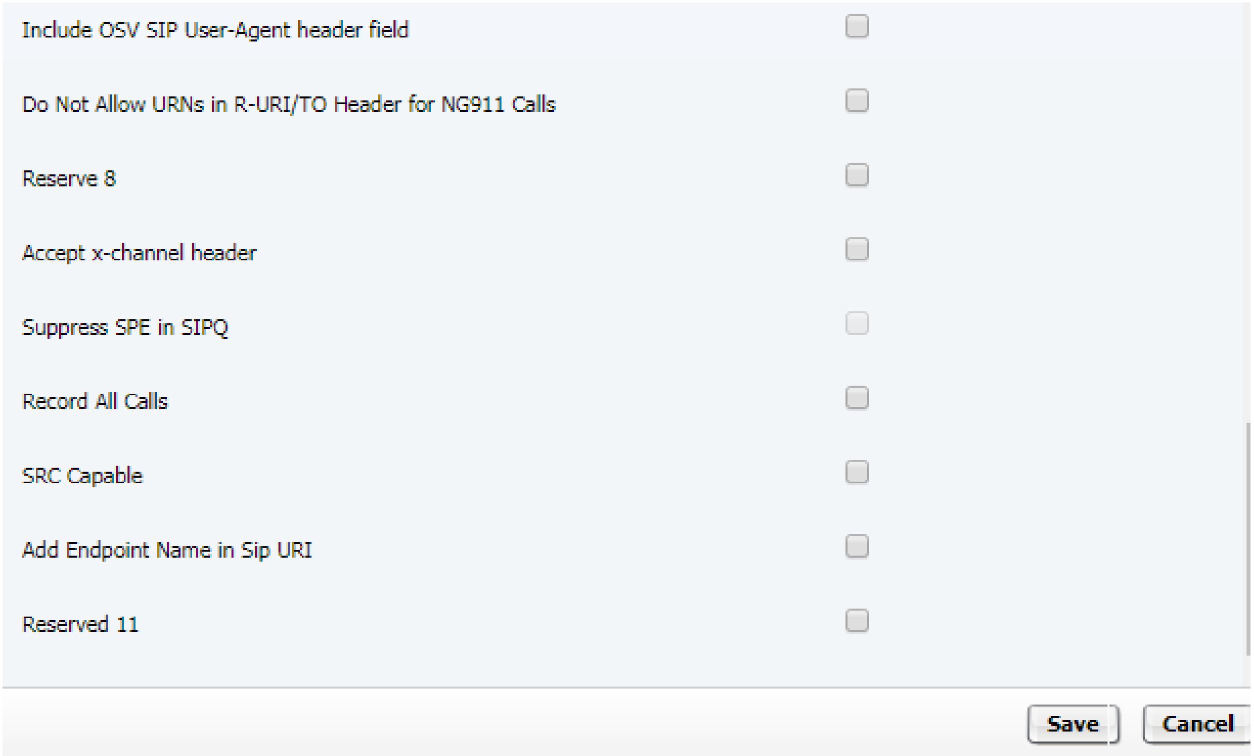

Figure 31 Members Endpoints-continued

30. Click the **Aliases** tab.
31. Click **Add**.
32. Name is set to 10.70.14.25:50010

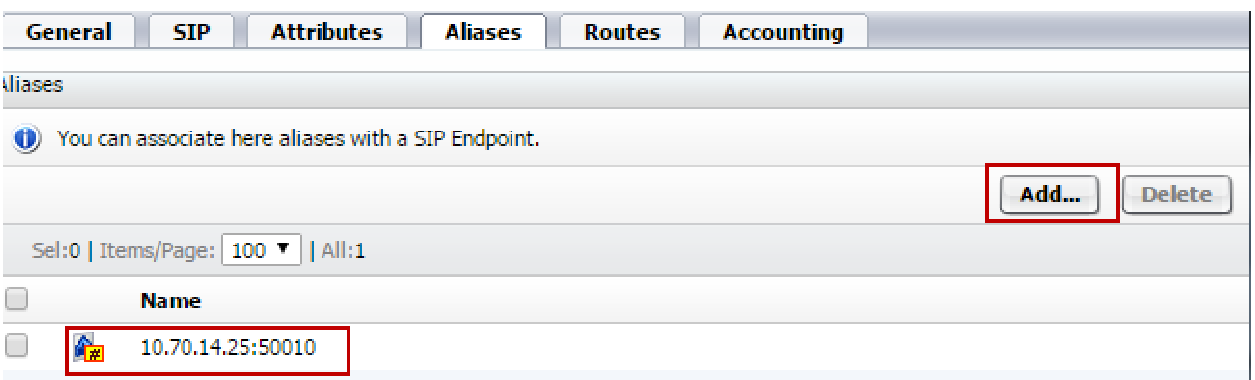

Figure 32 Members Endpoints-continued

33. **Name** is set to COX_MedServer for example.
34. **Registered** in checked.
35. **Profile** is set to COX_EP for example using the drop down button.

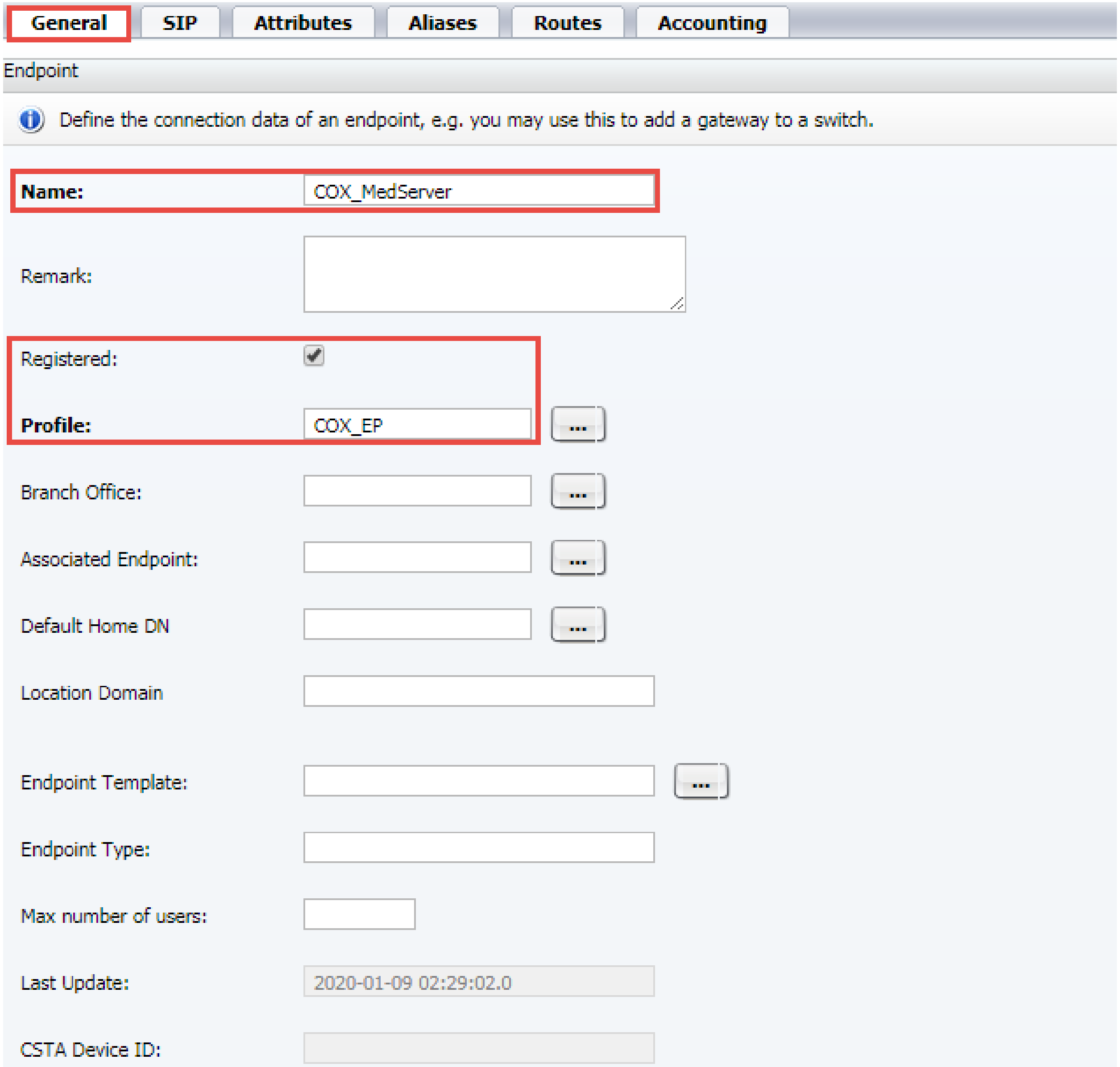

Figure 33 Members Endpoints-continued

36. Click on the **SIP** tab.
37. **SIP Trunking** button is selected.
38. **Type** is set to Static.
39. **Signaling Address Type** is set to IP Address or FQDN using the drop down arrow.
40. **Endpoint Address** is set to 10.70.14.6 for example. This is the Unify OpenScape Voice's media IP address.
41. **Port** is set to 5062.
42. **Transport protocol** is set to TCP using the drop down arrow and Click on Save.

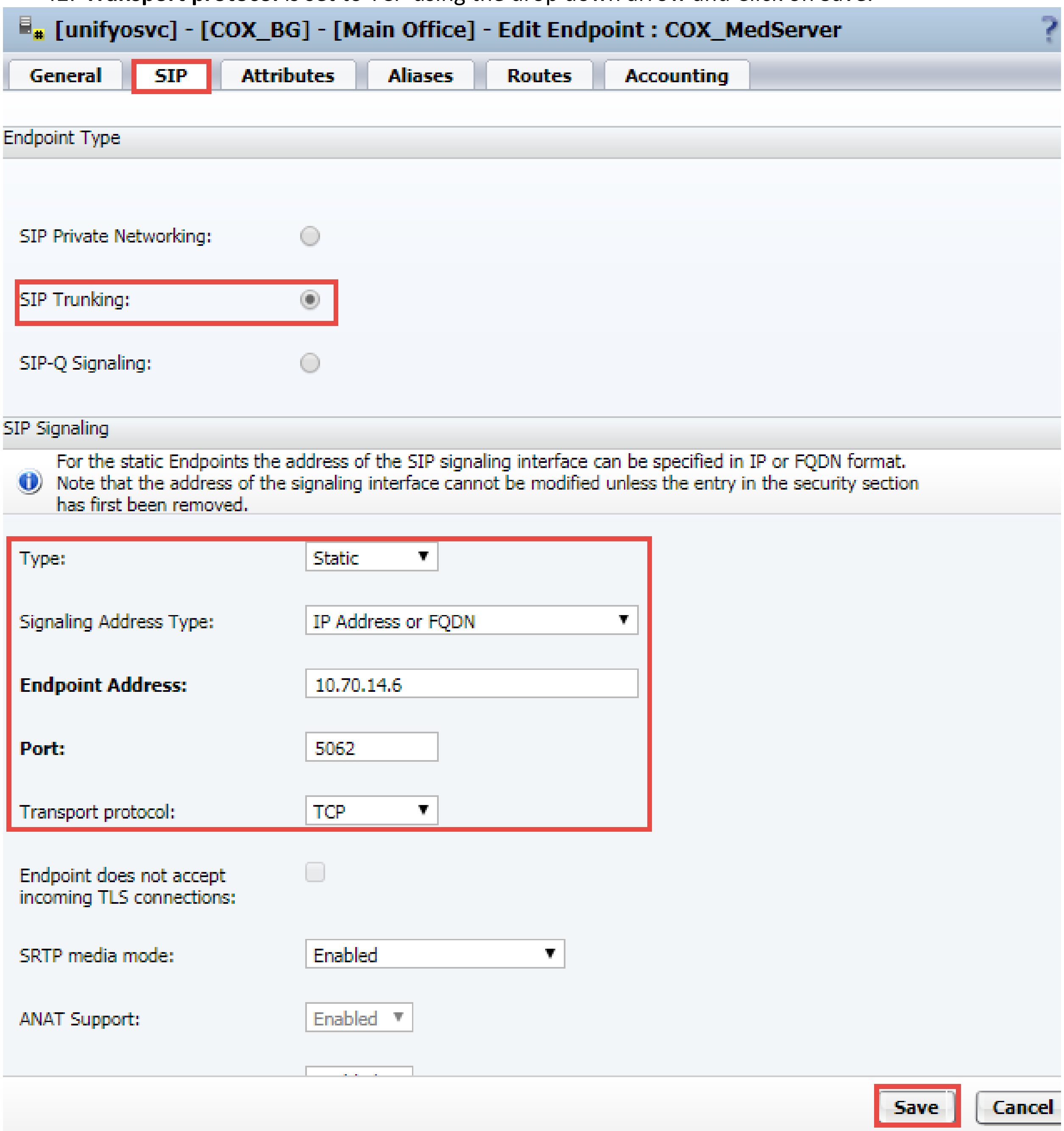

Figure 34 Members Endpoints-continued

3.6 Profiles Feature

1. Click on **Profiles** and **Feature**.
2. Click **Add**.

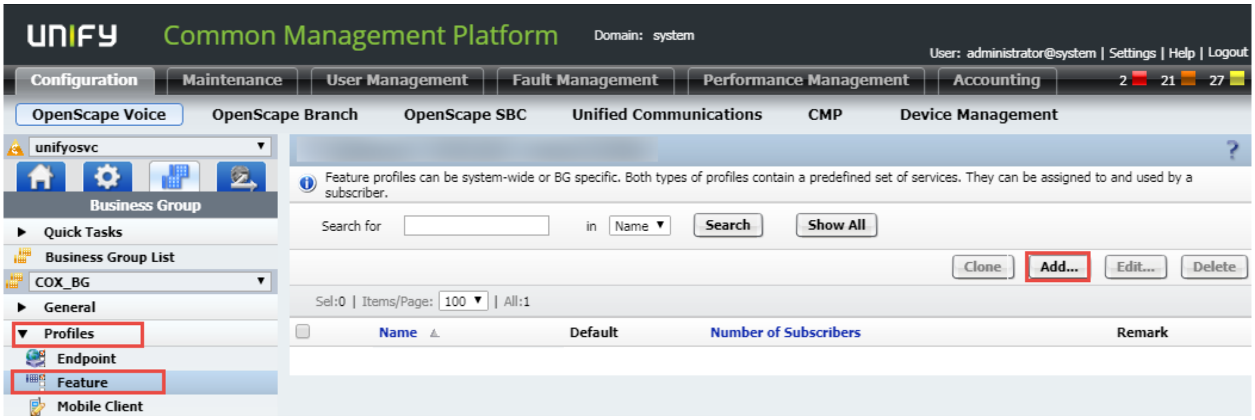

Figure 35 Profiles Feature

3. Set **Name**. FP_COX is given for this example.

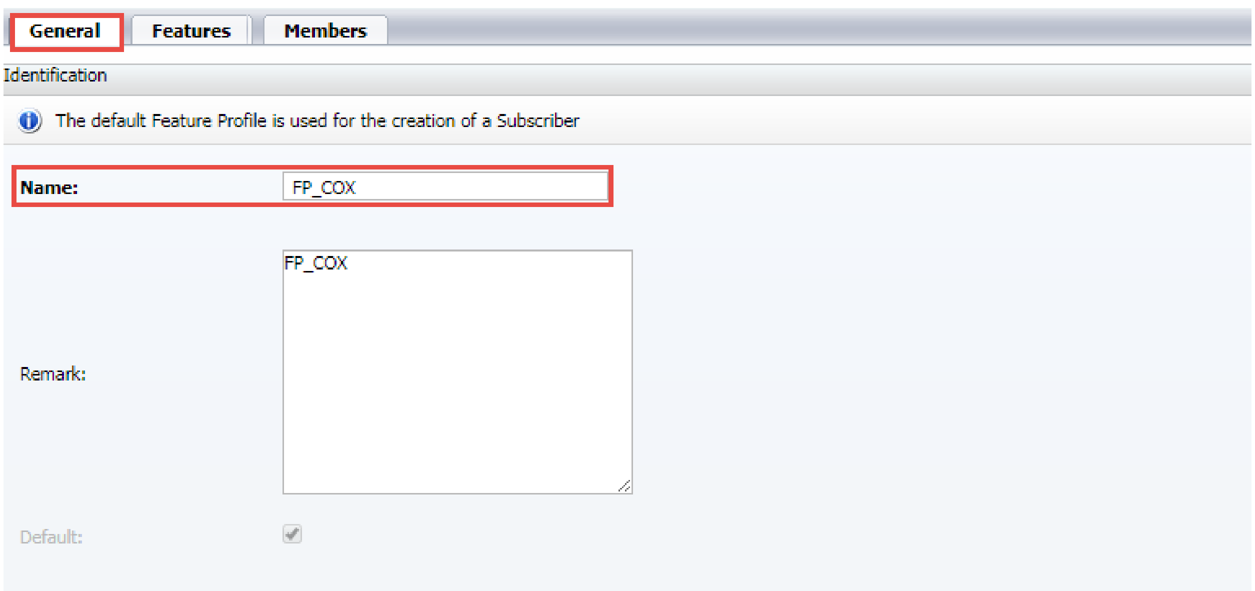

Figure 36 Profiles Feature-continued

4. Click on the **Features** tab.
5. **Features** can be modified based the requirement.
6. Click **Save**.

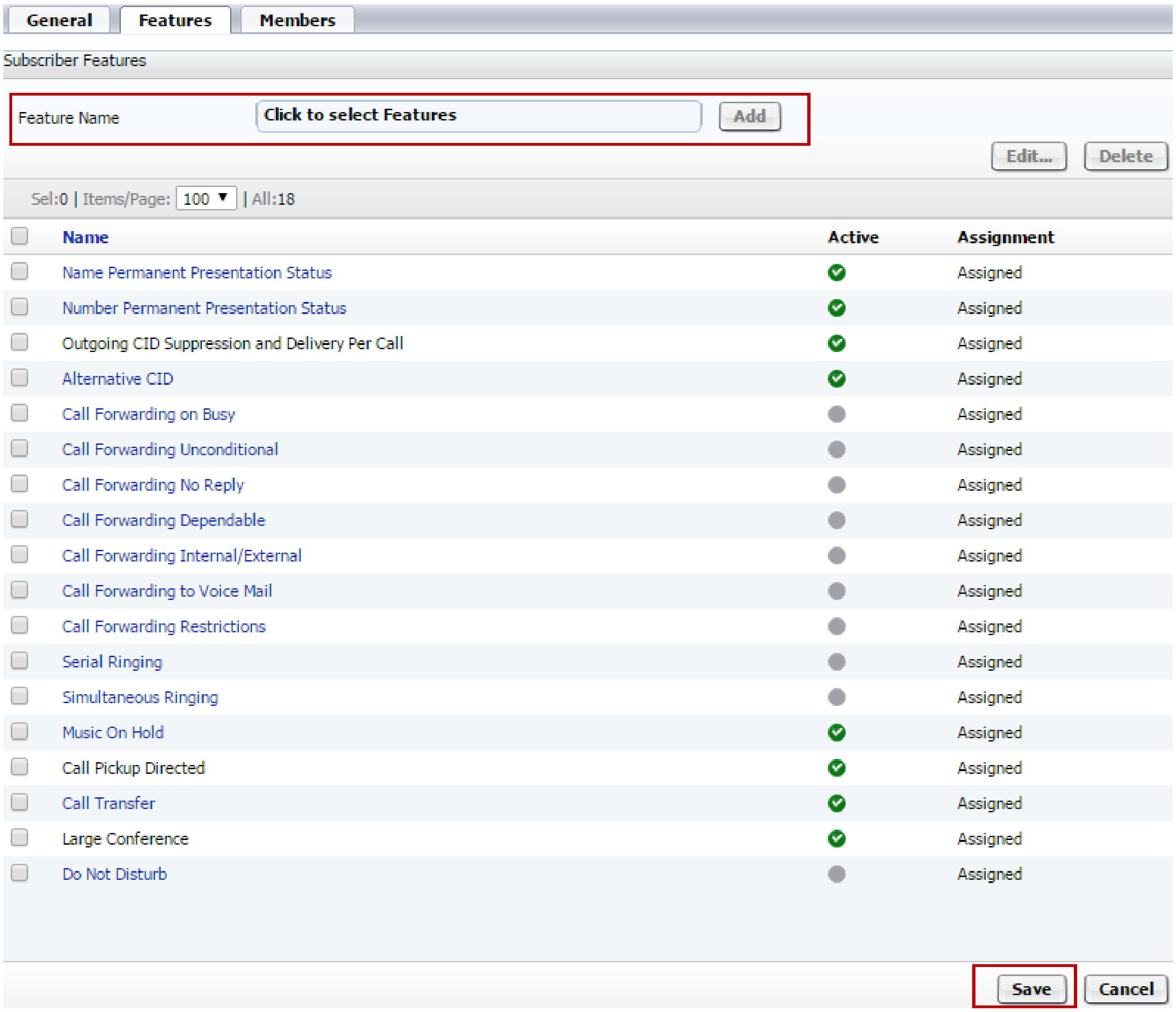

Figure 37 Profiles Feature-continued

3.7 Members Subscribers

1. Click **Members** and **Subscribers**.
2. Click **Add**.

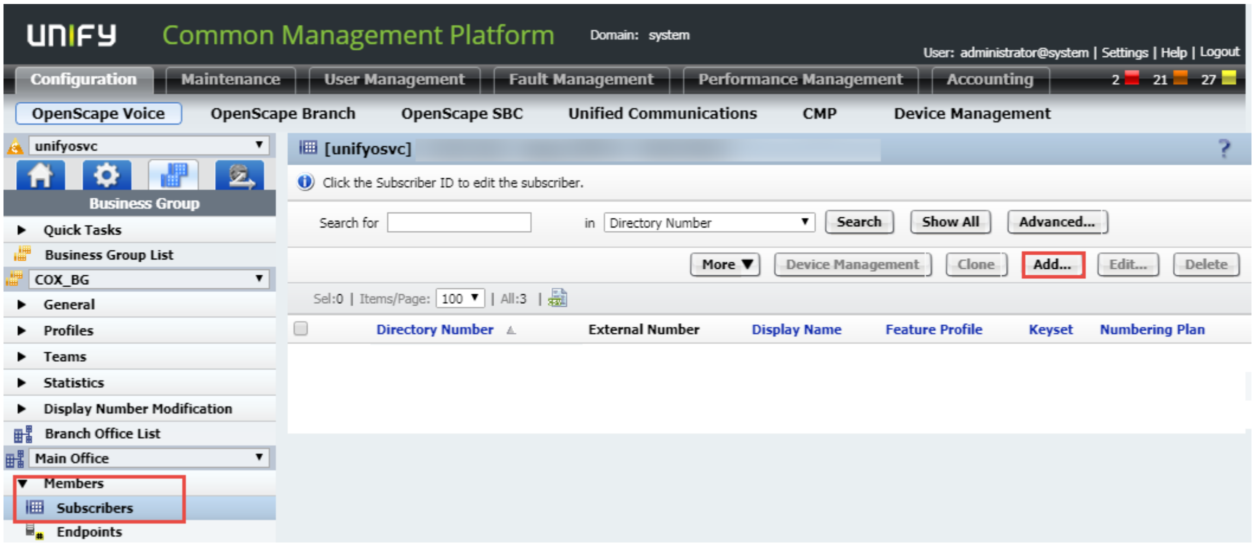

Figure 38 Members Subscribers

3. **Directory Number** is set to ()402932-4962 for example using the drop down button.

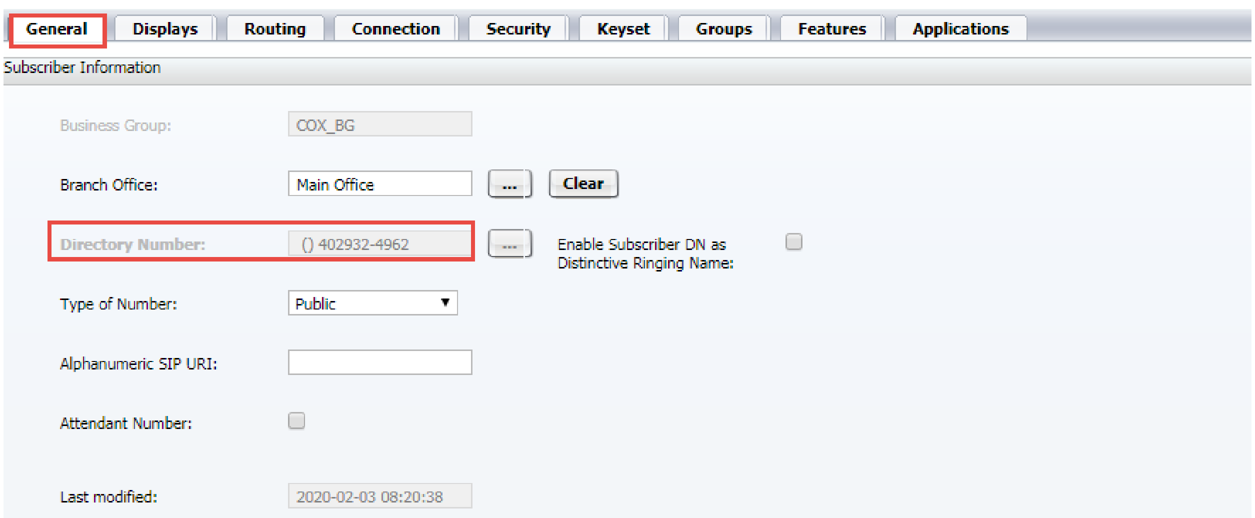

Figure 39 Members Subscribers-continued

4. Click on the **Displays** tab.
5. **Displayed Extension Number** is set to 4029324962 for example.
6. **External Caller ID** is set to 4029324962for example.
7. **Display Name** is set to 4029324962for example.
8. **External Display Name** is set to 4029324962 for example.

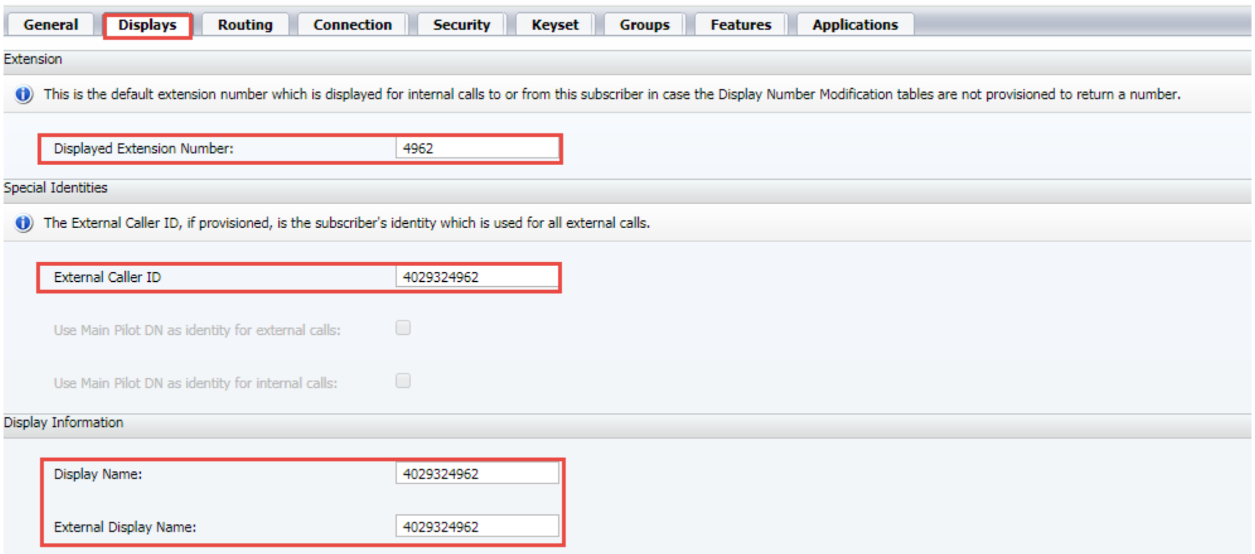

Figure 40 Members Subscribers-continued

9. Click the **Routing** tab.
10. **Numbering Plan** is set to NP_COX_BG for example using the drop down button.

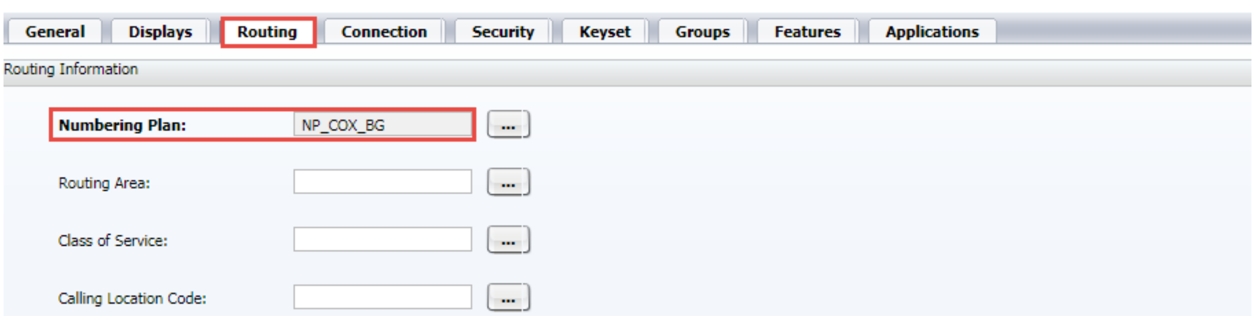

Figure 41 Members Subscribers-continued

11. Click on **Connection** tab
12. **Type** is set to Static for example.
13. **Transport Protocol** is set to TCP.
14. **IP Address** is set to 172.16.31.135 and port 5060 for example. This is the phone's IP address and port.

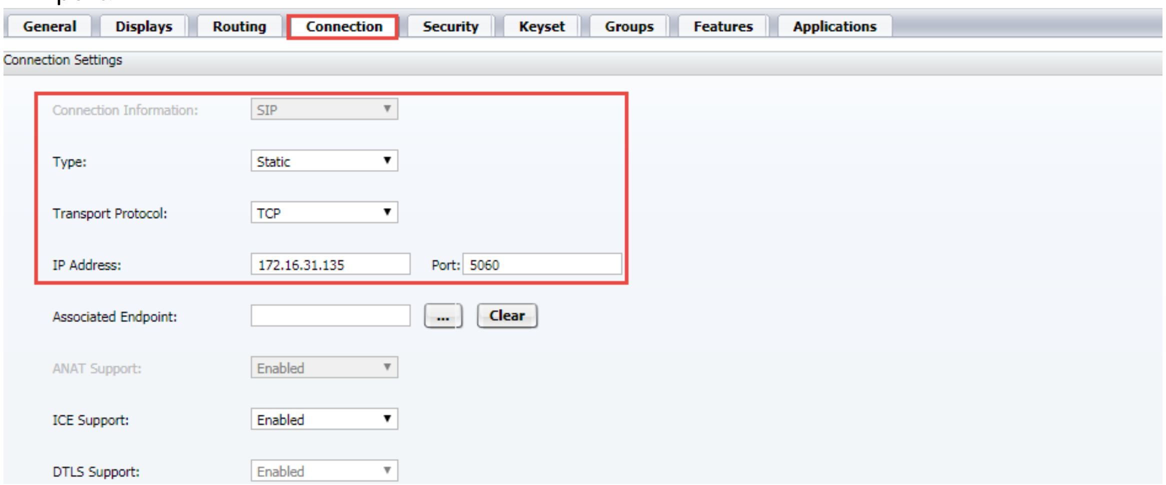

Figure 42 Members Subscribers-continued

15. Click the **Features** tab.
16. Phone features such as call forward can be modified here as needed.
17. Click **Save**.

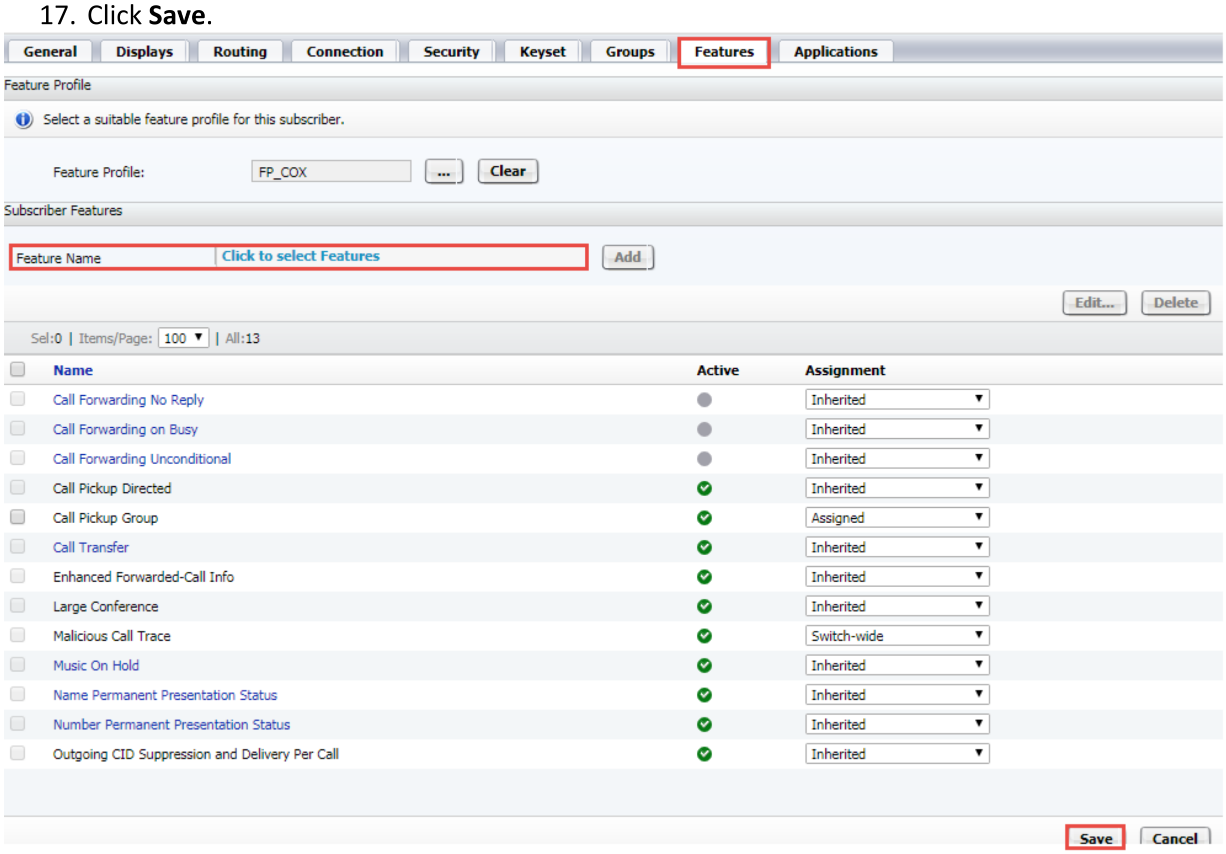

Figure 43 Members Subscribers-continued

3.8 Translation Prefix Access Codes

1. Click on **Translation** and **Prefix Access Codes**.
2. Click **Add**.

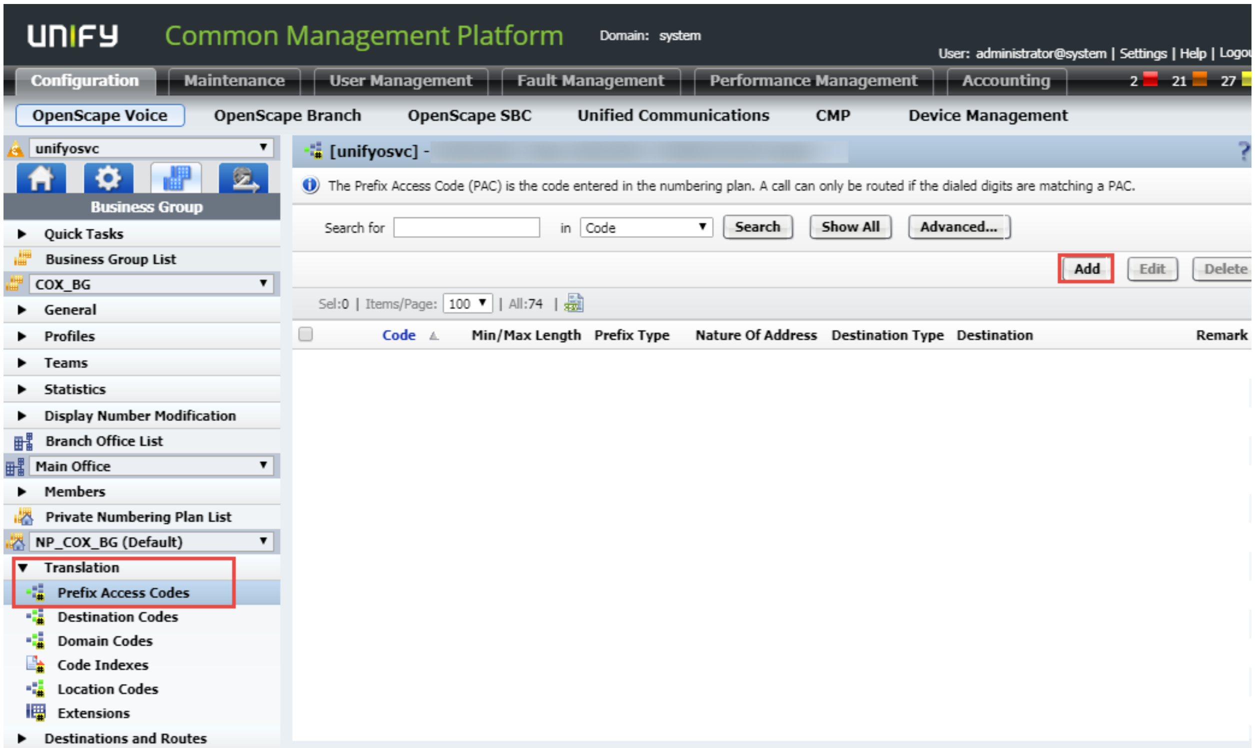

Figure 44 Translation Prefix Access Codes

3.8.1 For outbound Call Routing

3. **Prefix Access Codes** is set to 81 for example.
4. **Minimum Length** is set to 4 for example.
5. **Maximum Length** is set to 20 for example.
6. **Digit Position** is set to 0 for example.
7. **Prefix Type** is set to Off-net Access from the dropdown menu**.**
8. **Nature of Address** is set to National from the dropdown menu.
9. Click **Save.**

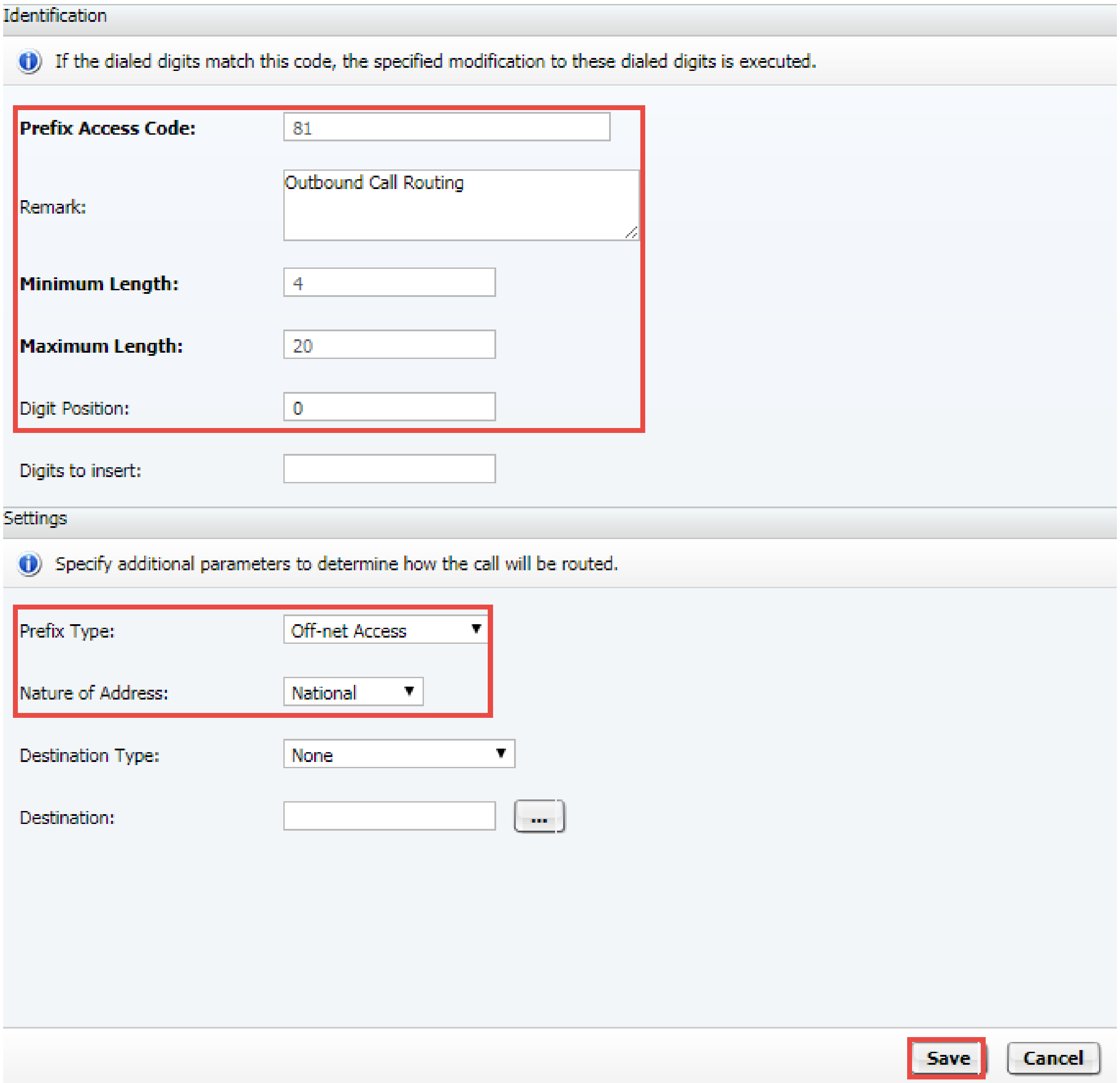

Figure 45 Translation Prefix Access Codes-continued

3.8.2 For Inbound Call Routing

1. **Prefix Access Codes** is set to 4 for example.
2. **Minimum Length** is set to 4 for example.
3. **Maximum Length** is set to 20 for example.
4. **Digit Position** is set to 0 for example.
5. **Prefix Type** is set to Off-net Access from the dropdown menu**.**
6. **Nature of Address** is set to Subscriber from the dropdown menu.
7. Click **Save.**

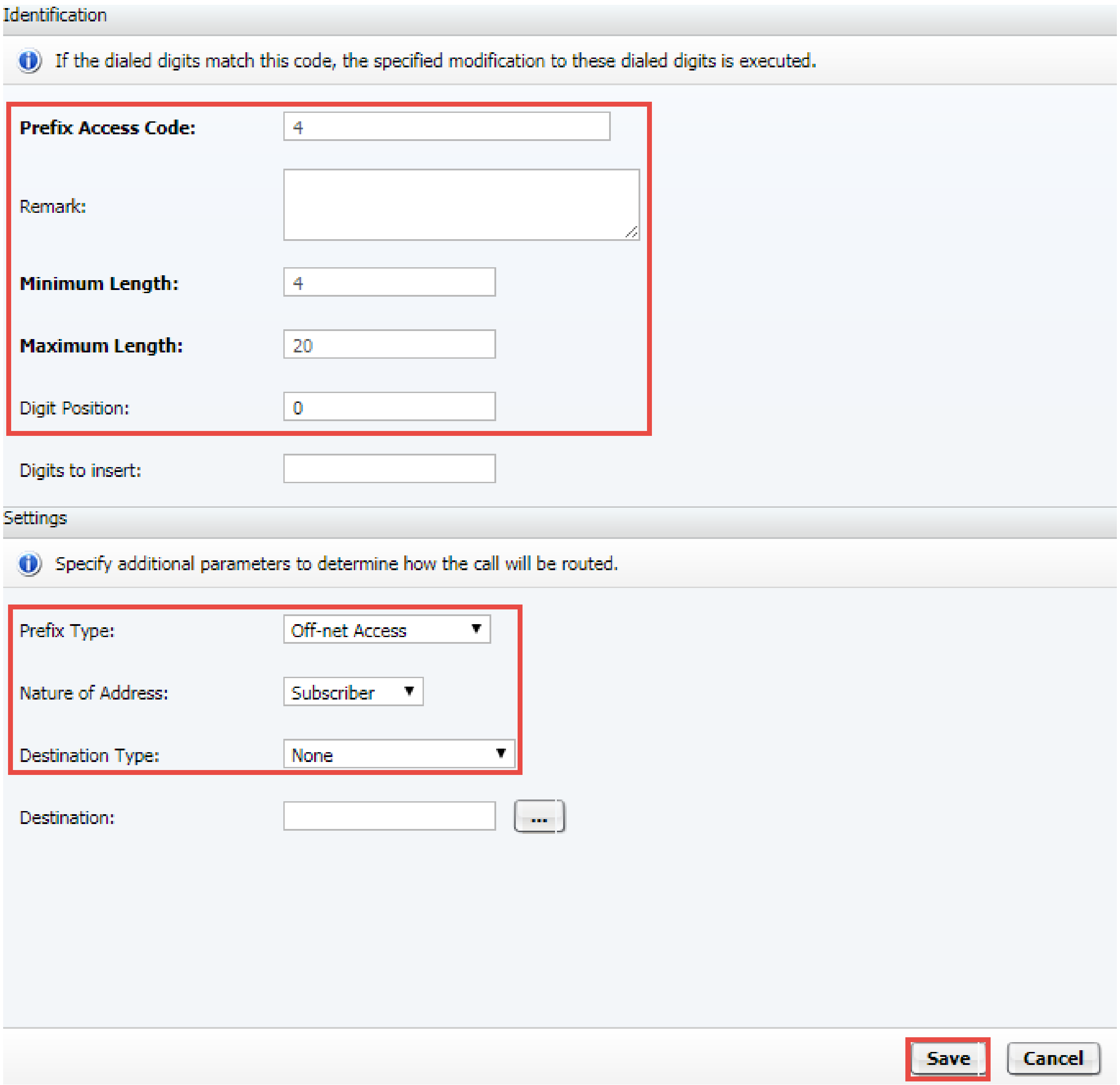

Figure 46 Translation Prefix Access Codes-continued

3.9 Translation Destination Codes

1. Click on **Translation** and **Destination Codes**
2. Click **Add**.

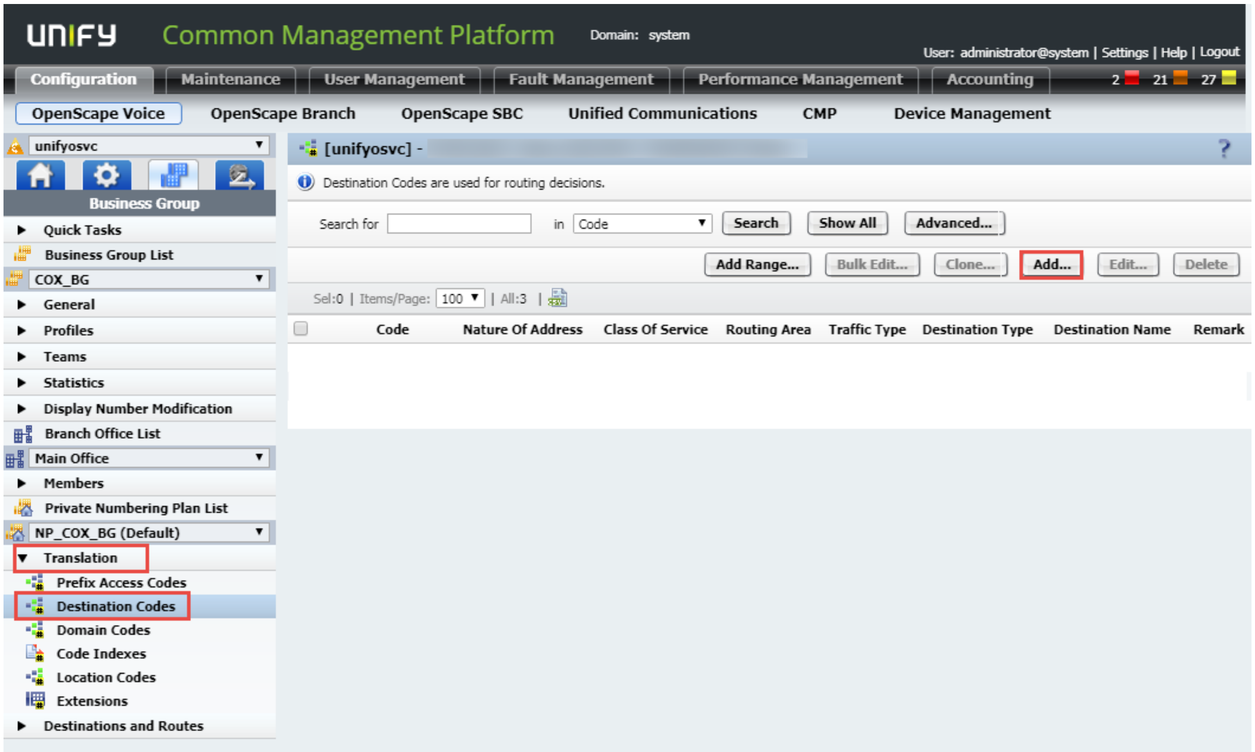

Figure 47 Translation Destination Codes

3.9.1 For Outbound Call Routing

1. **Destination Code**. 81 is selected from the prefix access code defined for outbound call routing.
2. **Nature of Address**. National is selected from drop down menu.
3. **Destination Type**. Destination is selected from the drop down menu.
4. **Destination**. Select the destination which was already configured.
5. Click **Save**.

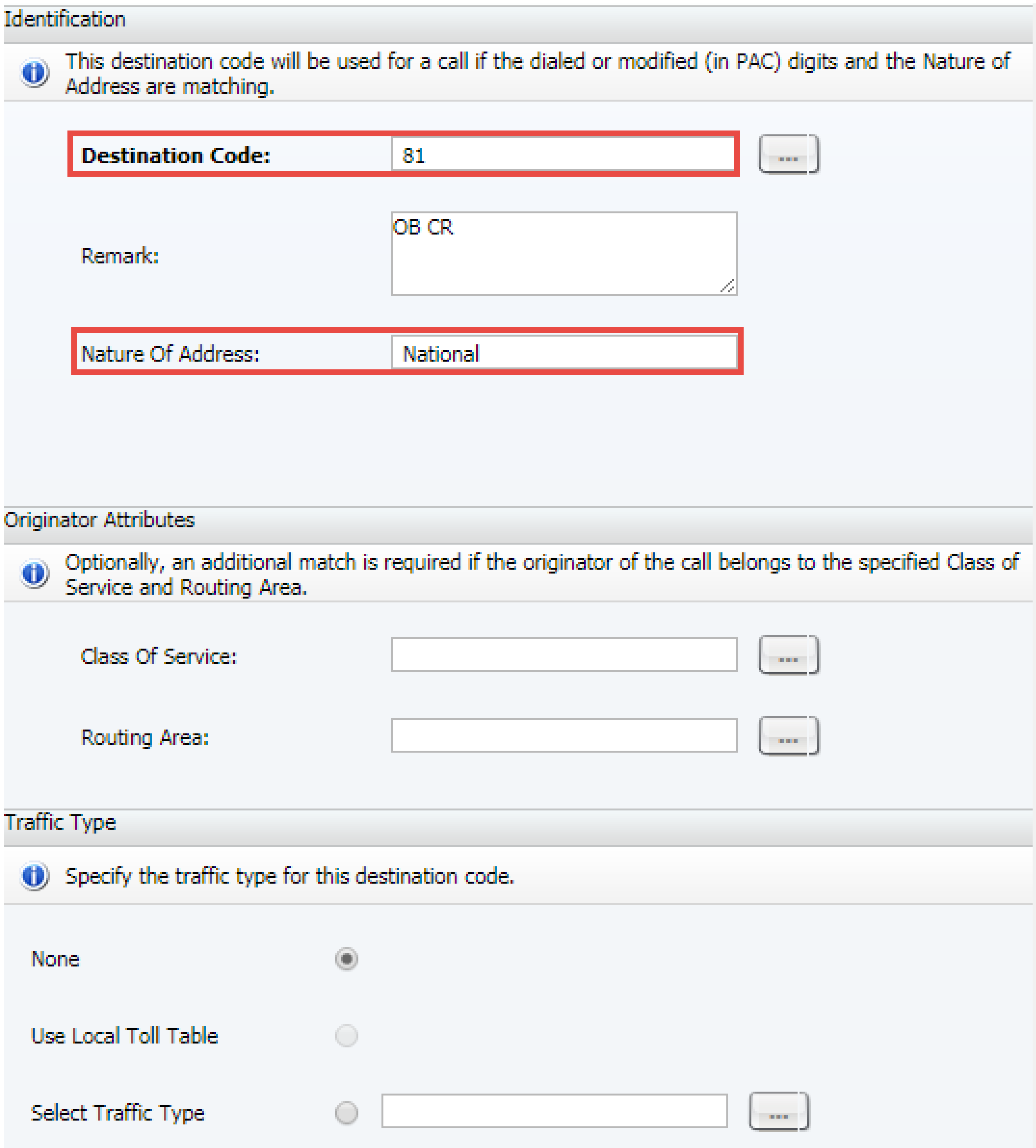

Figure 48 Translation Destination Codes Continued

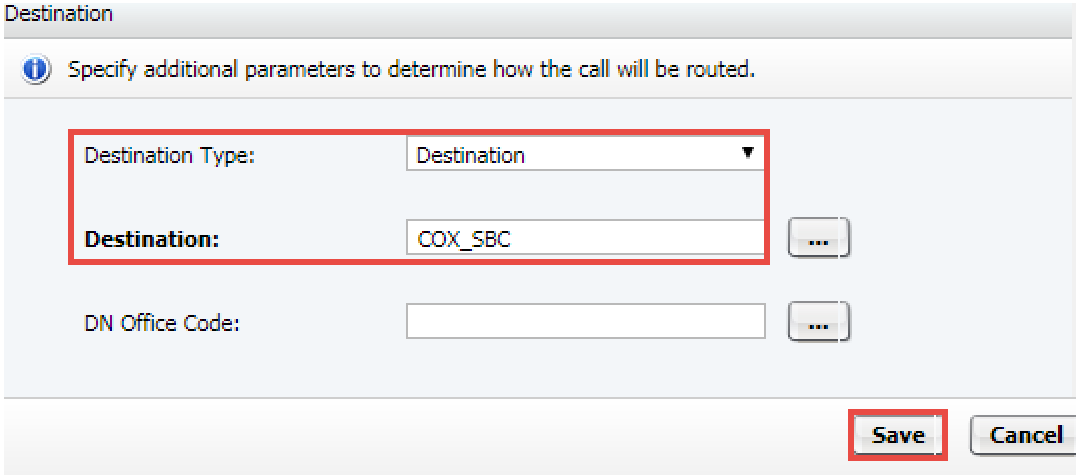

Figure 49 Translation Destination Codes Continued

3.9.2 For Inbound Call Routing

1. **Destination Code** is set to 4 for example using the drop down button.
2. **Nature of Address** is set to Subscriber for example.
3. **Destination Type** is set to Home DN for this example.
4. **Office Code**. Appropriate office is selected.

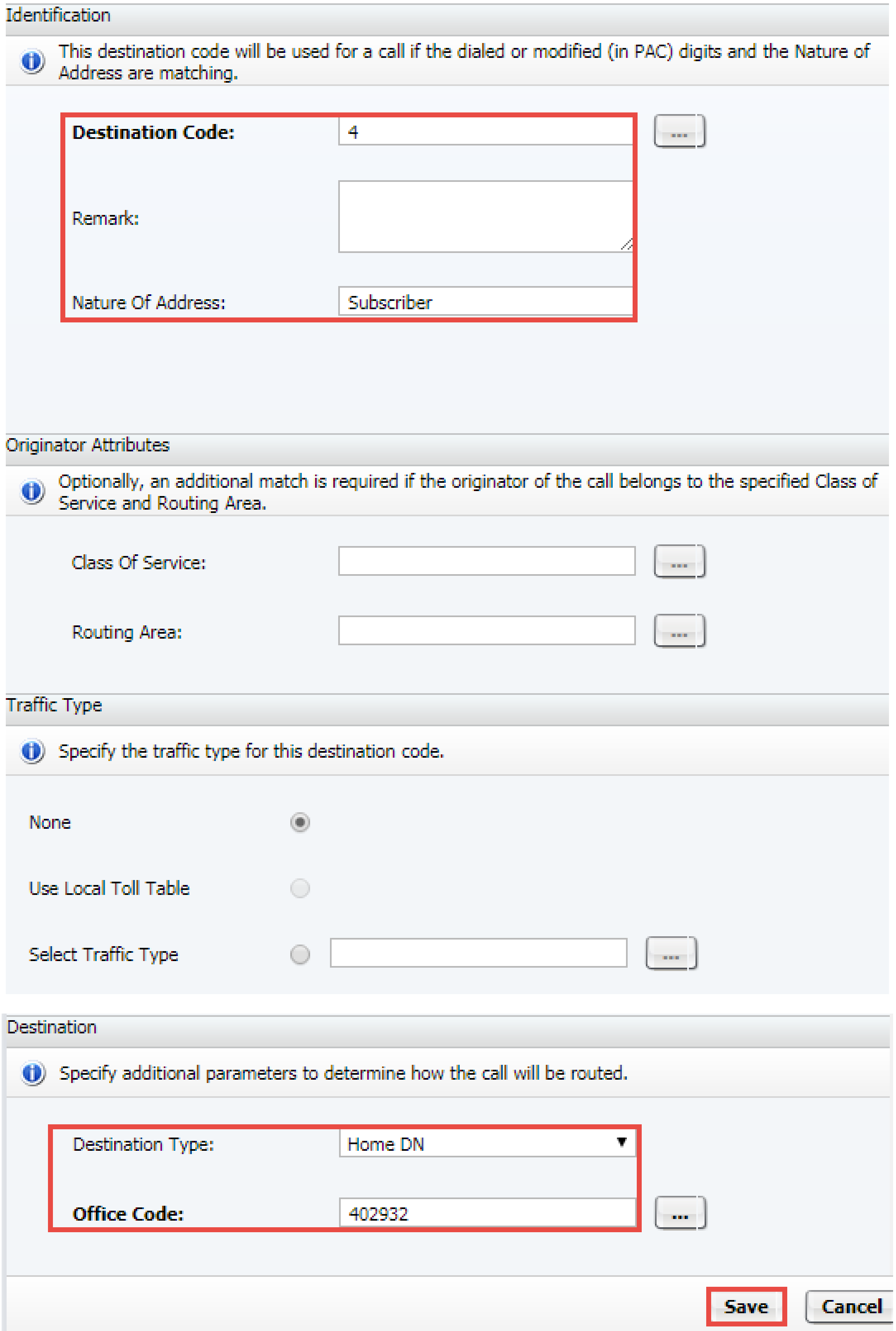

Figure 50 Translation Destination Codes Continued

3.10 Destinations and Routes: Destinations

1. Click on **Destinations and Routes** and **Destinations**.
2. Click **Add**.

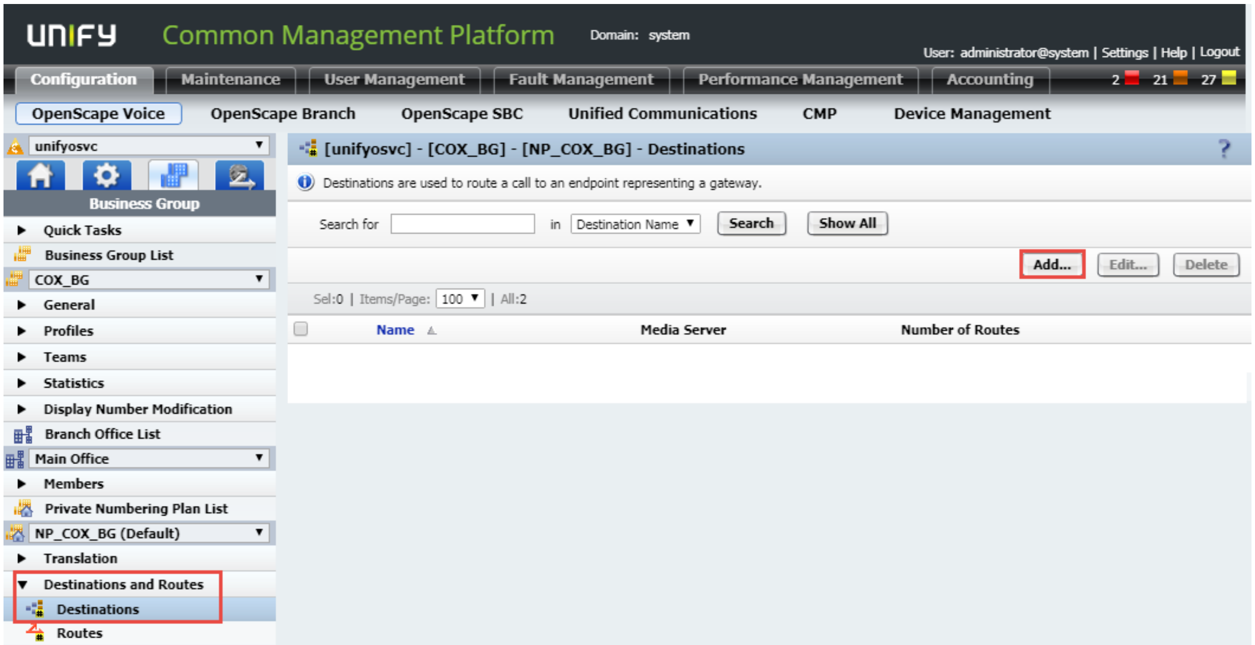

Figure 51 Destination and Routes: Destination

3. **Name** is set to COX_SBC

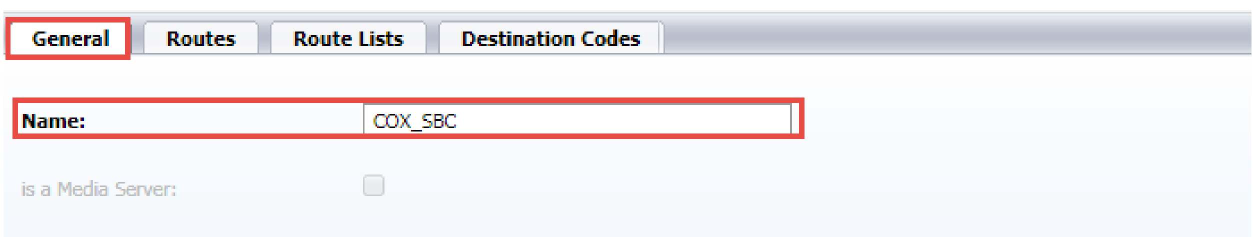

Figure 52 Destination and Routes: Destination--Continued

4. Click **Routes** Tab.
5. Click **Add**.

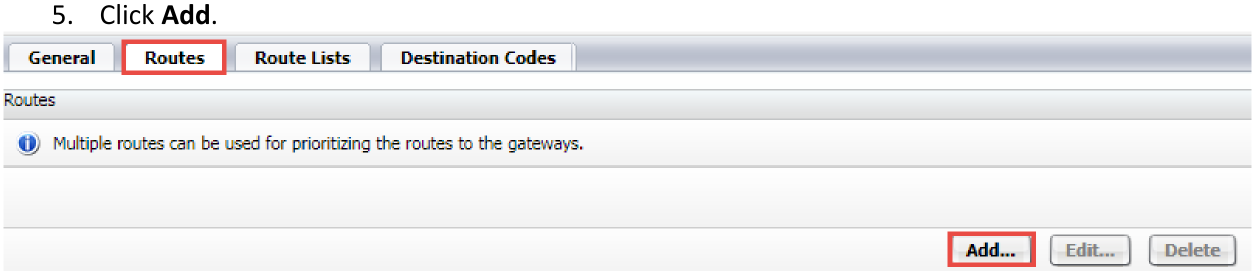

Figure 53 Destination and Routes: Destination--Continue

6. **ID** is set to 3.
7. **SIP Endpoint** COX_SBC_Trunk is selected from the pop up window.
8. **Modification Type** is set to Number manipulation from drop down menu.
9. **Number is digits** to delete is set to 2. Since we use 81 as access code, this field is set to delete access code before sending call out.
10. Click **Save**.

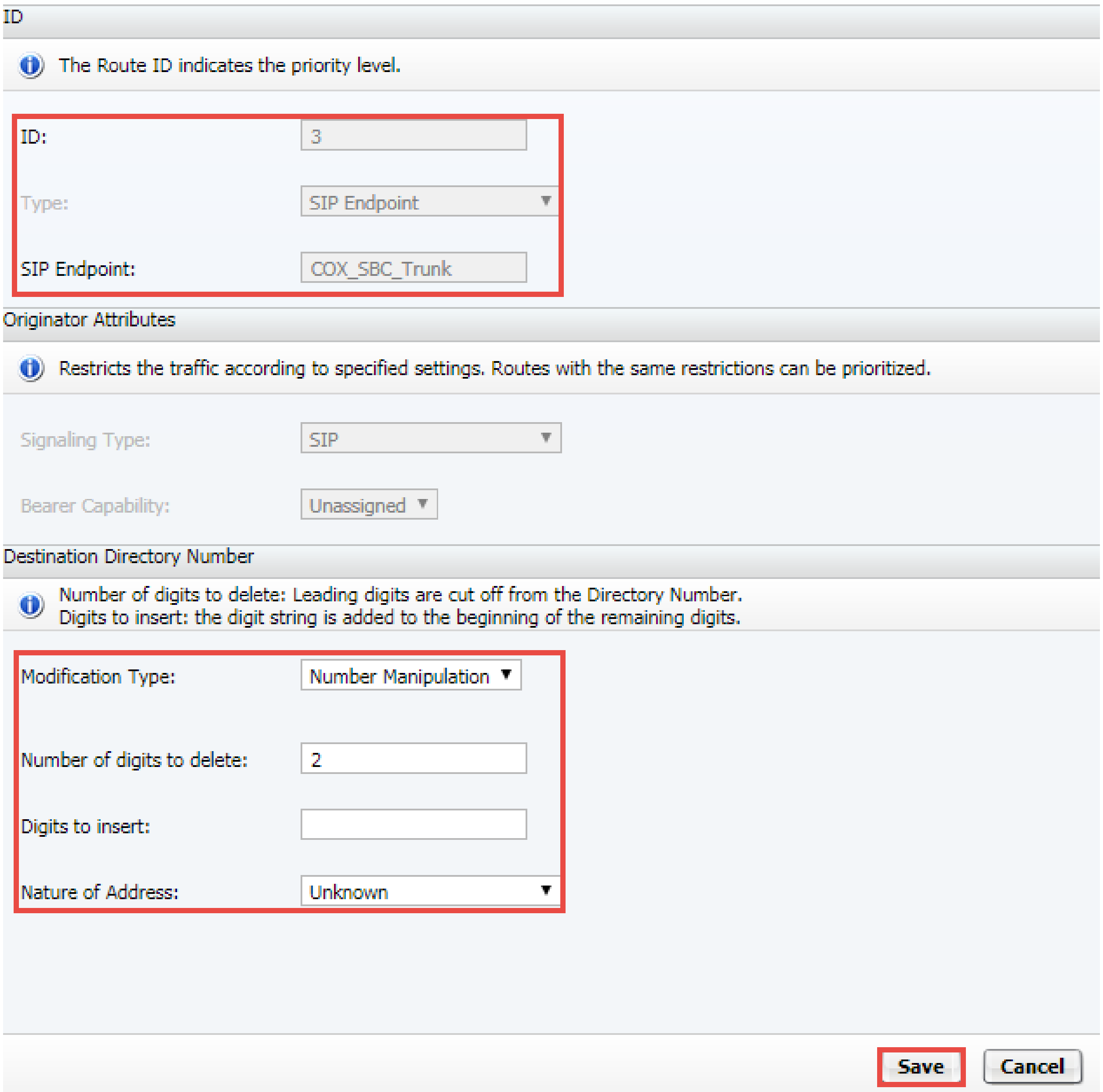

Figure 54 Destination and Routes: Destination--Continued

4 Unify SBC Configuration

These are the main tasks to configure Unify SBC to perform Direct SIP integration with the Unify OSV and COX Network.

4.1 Network/Net Services: Settings
4.2 Network/Net Services: DNS
4.3 VoIP: SIP Server Settings
4.4 Features: Enable Remote Endpoints

4.1 Network/Net Services: Settings

1. Click on **Network/Net Services** and **Settings**.

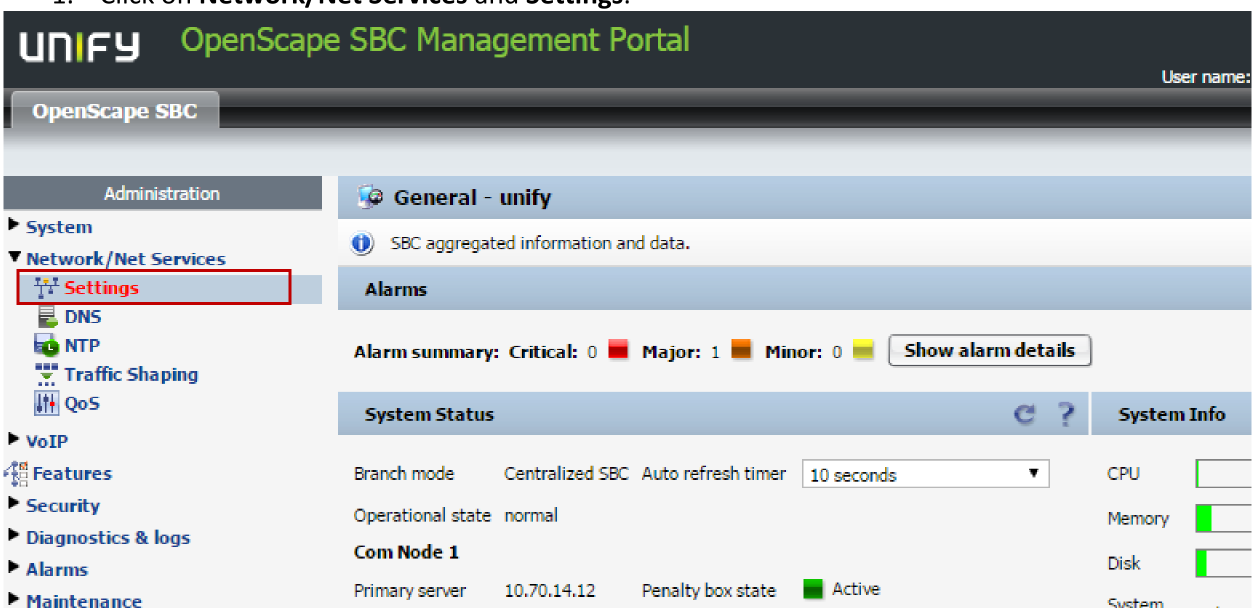

Figure 55 Network/Net Services: Settings

2. Under the **Physical Network Interface** section, verify eth0 and eth1 are enabled.
3. Under the **Core realm configuration** section, click Add.
4. **Type** is set to Main IPv4.
5. **Network ID** is set to Main-Core-IPv4.
6. **IP Address** is set to 10.70.14.25 for example. This is the Unify SBC's LAN IP address.
7. **Subnet mask** is set to 255.255.255.0 for example. This is the Unify SBC's LAN IP mask.
8. **SIP-UDP** is set to 5060.
9. **SIP-TCP** is set to 5060.
10. **SIP-TLS** is set to 5061.

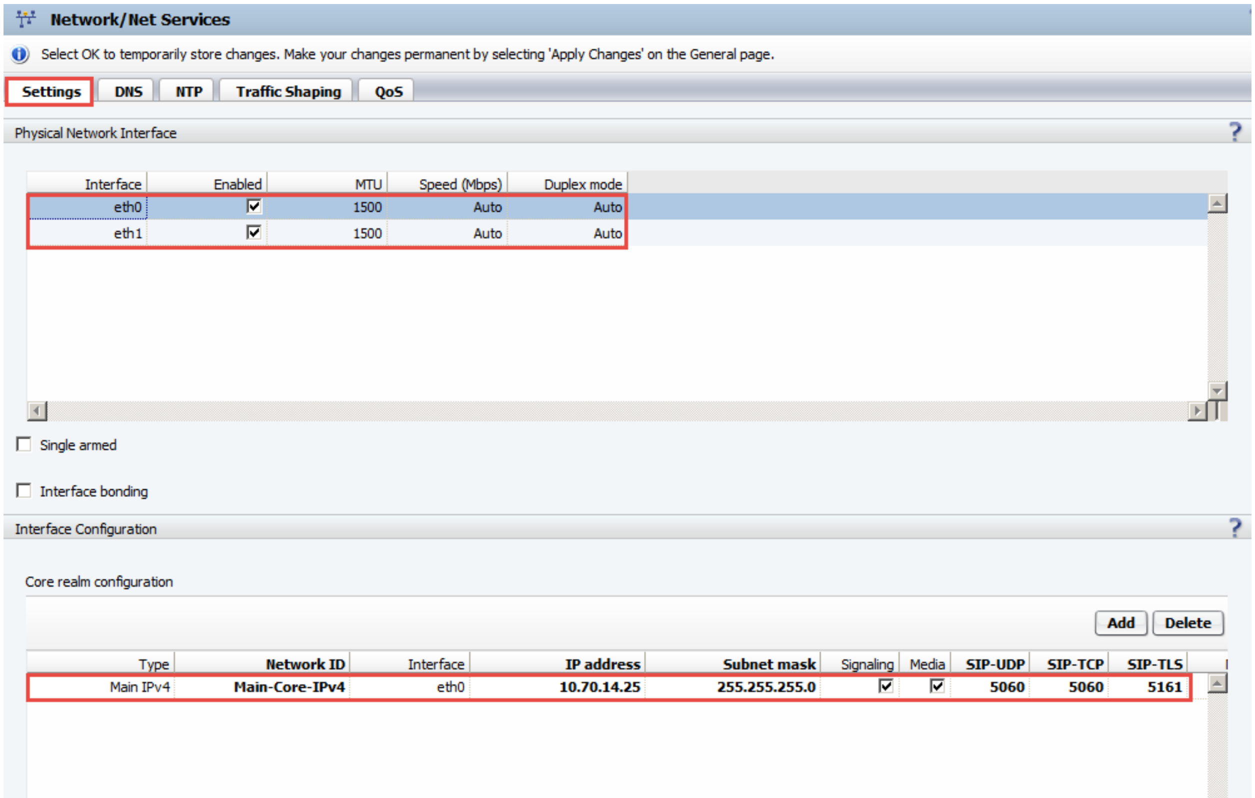

Figure 56 Network/Net Services: Settings-continued

11. Scroll down to **Access** and **Admin realm configuration** section.
12. **Type** is set to Main IPv4.
13. **Network ID** is set to Main-Access-IPv4.
14. **IP Address** is set to 10.64.3.146 for example. This is the Unify SBC’s WAN IP address.
15. **Subnet mask** is set to 255.255.0.0 for example. This is the Unify SBC’s WAN IP mask.
16. **SIP-UDP** is set to 5060.
17. **SIP-TCP** is set to 5060.
18. **SIP-TLS** is set to 5061.
19. Scroll down to **Realm Profile** section and map the Realm profile, Realm, Signaling network Id and Media network ID as show below

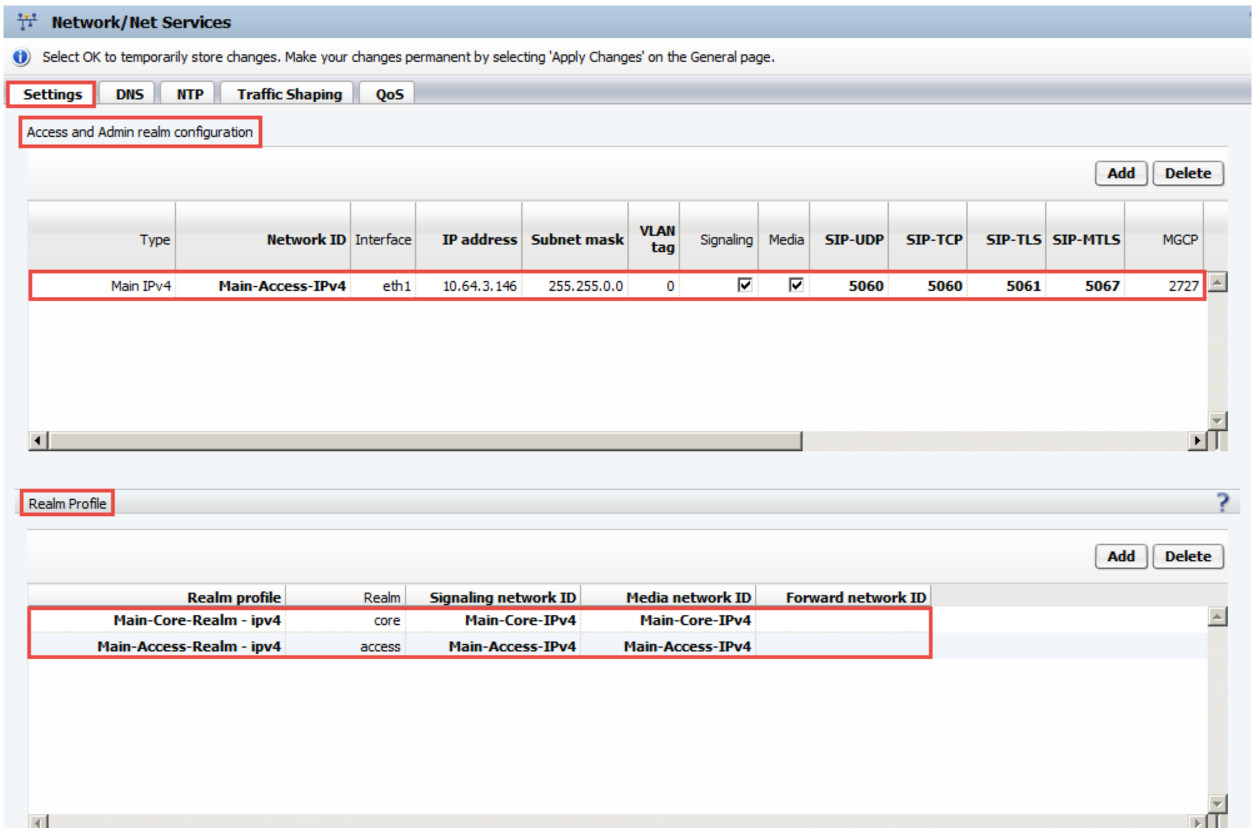

Figure 57 Network/Net Services: Settings-continued

20. **Default gateway address** is set to 10.64.1.1 for example.
21. Click **OK** at the bottom right of page

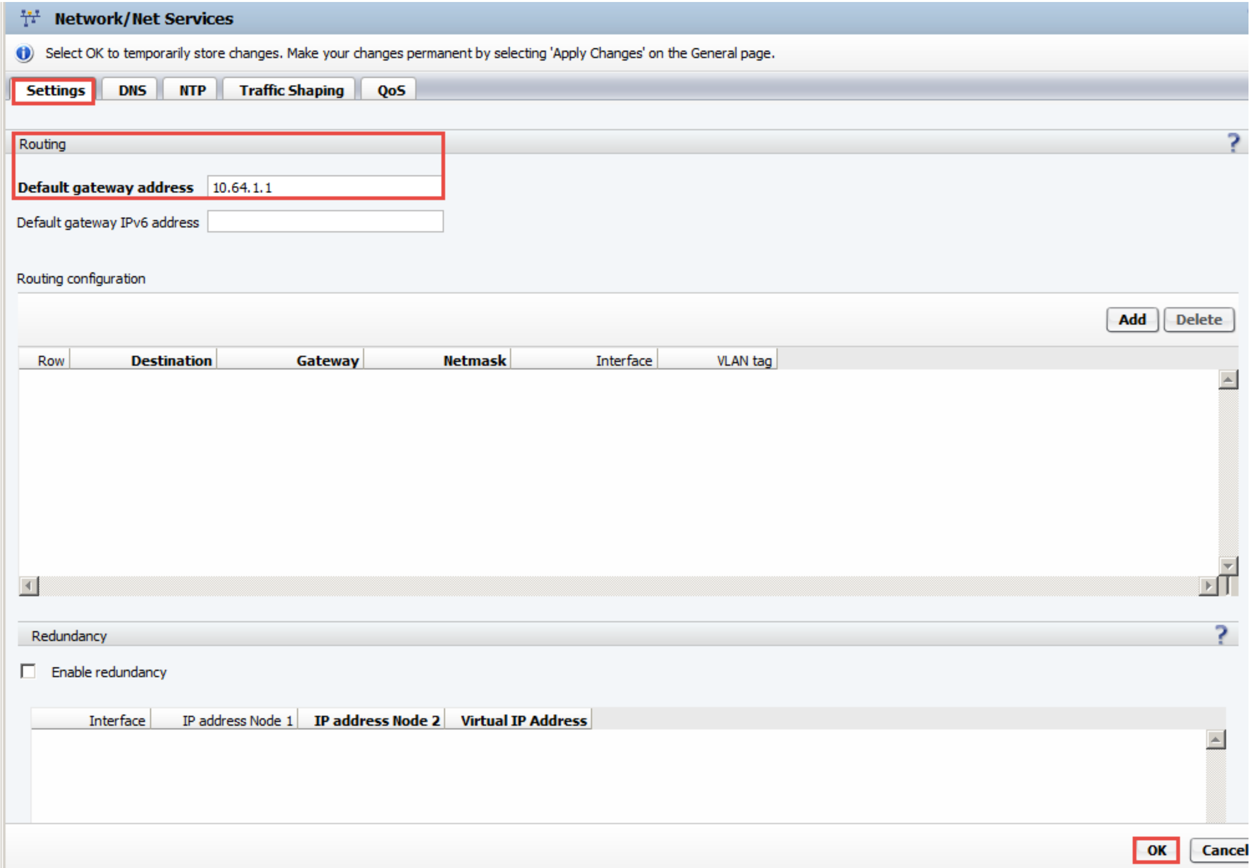

Figure 58 Network/Net Services: Settings-continued

22. On the main screen, click **Apply** changes at bottom right of page.

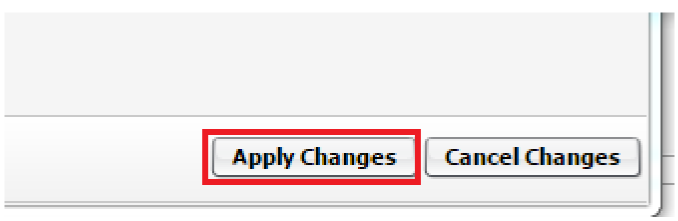

Figure 59 Network/Net Services: Settings-continued

4.2 Network/Net Services: DNS

1. Click on **Network/Net Services** and **DNS**.

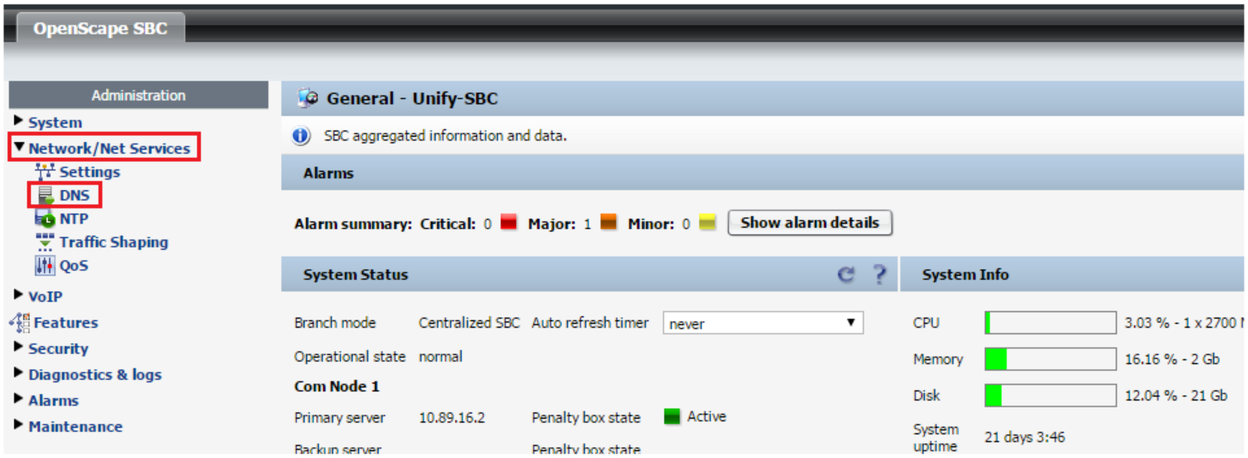

Figure 60 Network/Net Services: DNS

2. **DNS server IP address** is set to 10.64.1.3 for example.
3. Click **Add**.
4. Click **OK** at bottom right of page, not shown here.

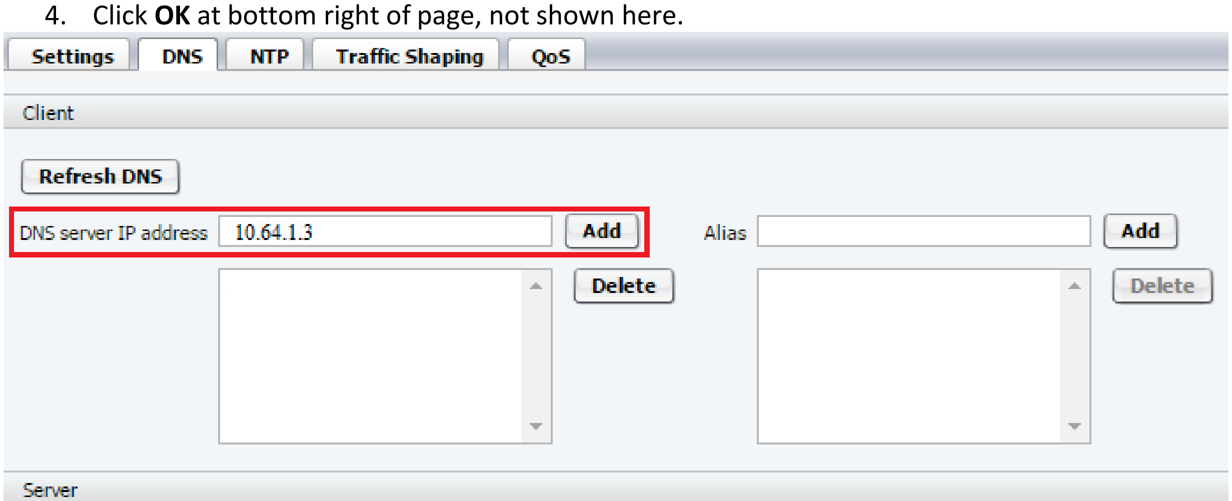

Figure 61 Network/Net Services: DNS-continued

5. On the main screen, click **Apply** changes at bottom right of page.

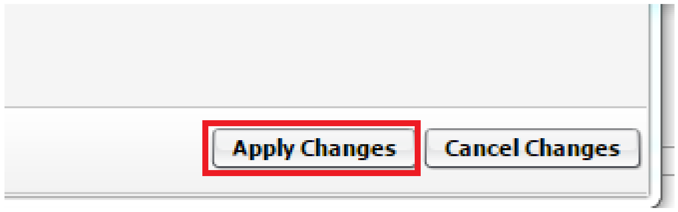

Figure 62 Network/Net Services: DNS-continued

4.3 VoIP: SIP Server Settings

1. Click **VoIP** and **SIP Server Settings**.

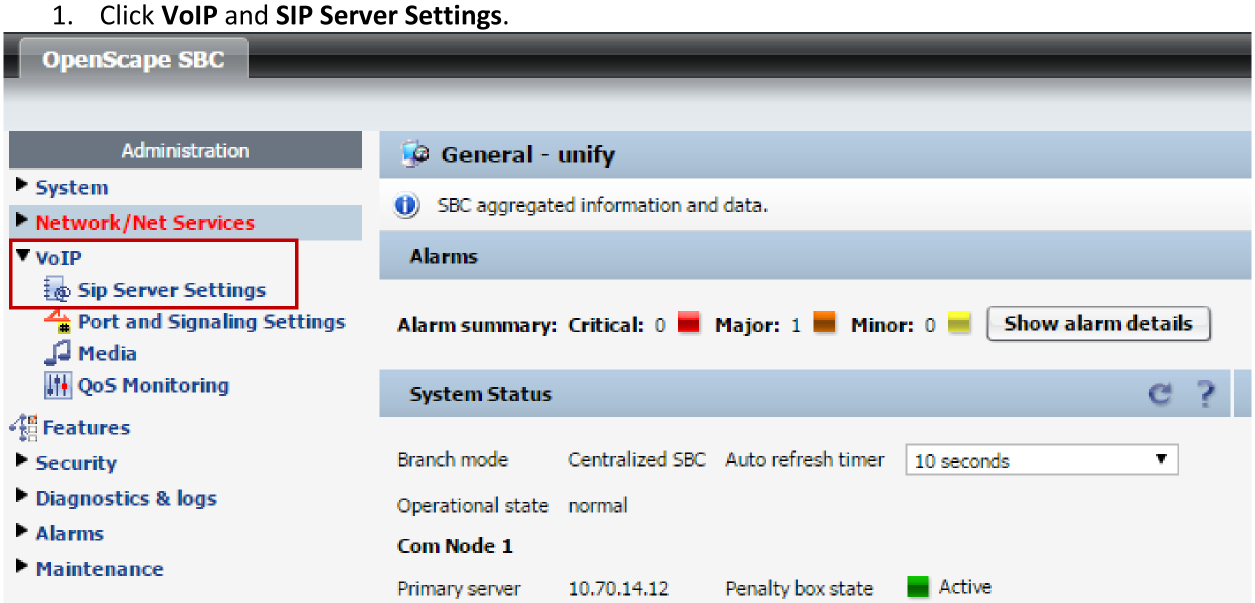

Figure 63 VoIP: SIP Server Settings

2. **Comm System Type** is set to Simplex for example.
3. **Target type** is set to Binding for example.
4. **Primary server** is set to 10.70.14.12, TCP, and 5060 for example. This is the Unify OSV's IP address and port information.
5. Click **OK** at bottom right of screen, not shown here.

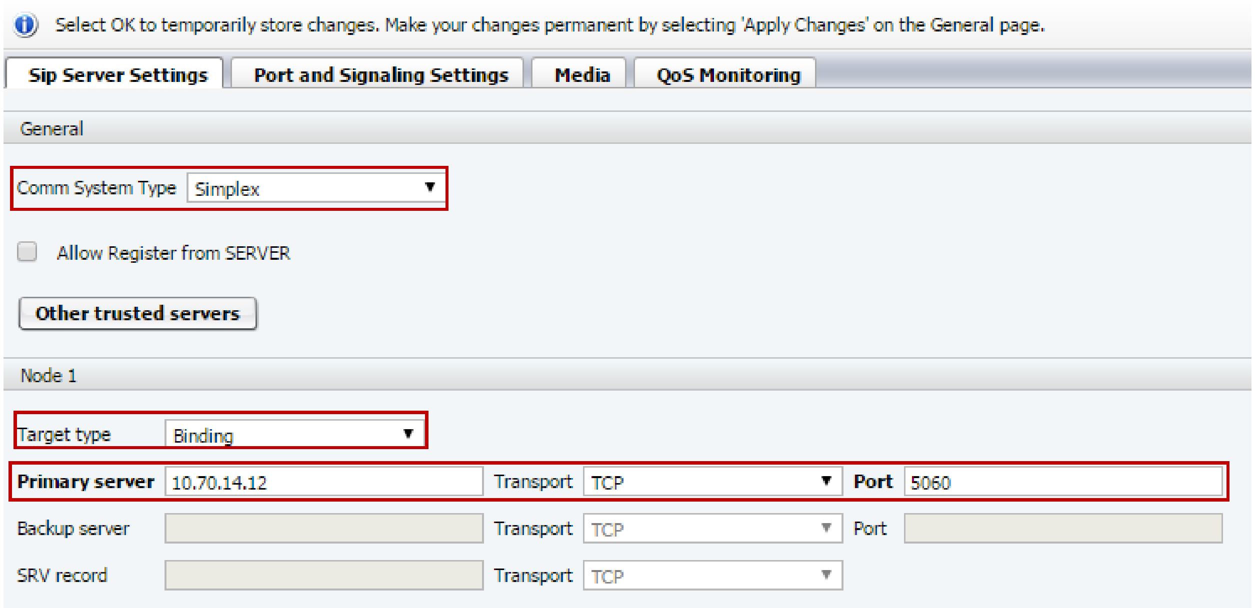

Figure 64 VoIP: SIP Server Settings-continued

6. On the main screen, click **Apply** changes at bottom right of page.

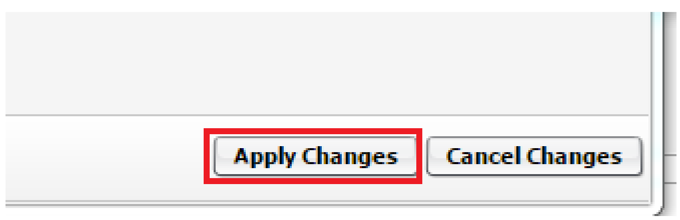

Figure 65 VoIP: SIP Server Settings-continued

4.4 Features: Enable Remote Endpoints

1. Click on **Features.**

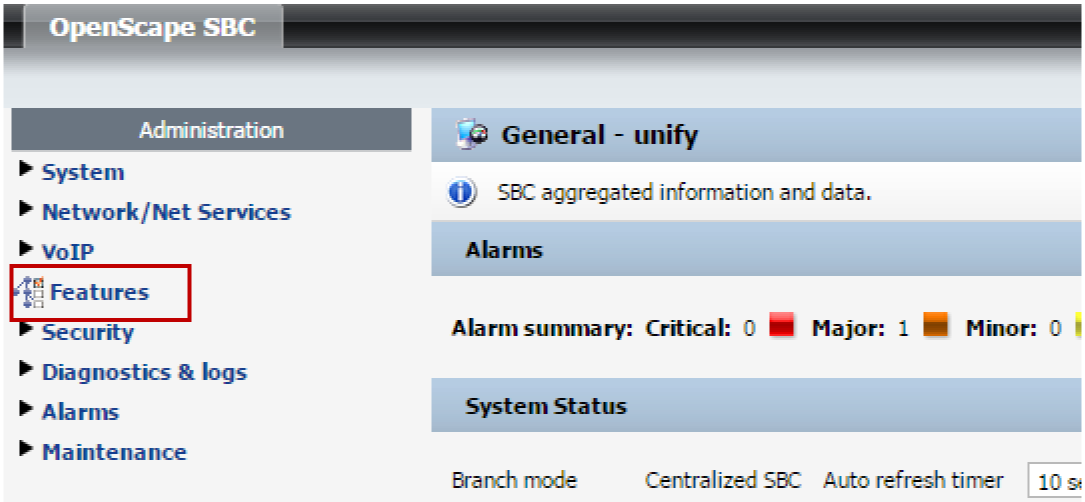

Figure 66 Features: Enable Remote Endpoints

2. Enable **Remote Endpoints** is checked.
3. Click the **Configure** button.

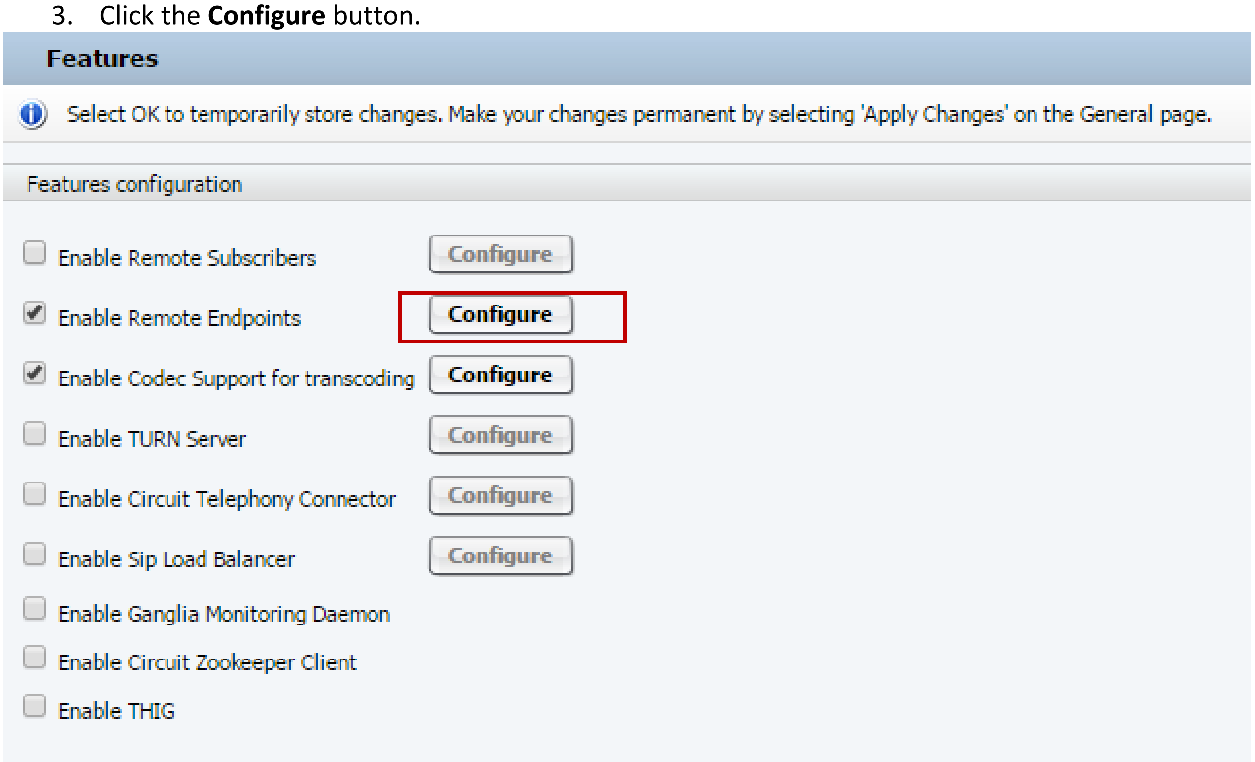

Figure 67 Features: Enable Remote Endpoints-continued

4. Under the **SIP Service Provider Profile** section, click **Add**.

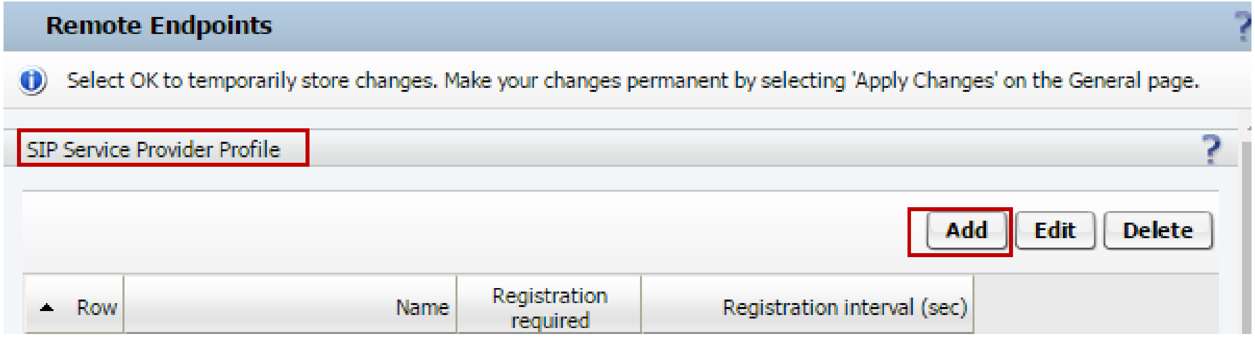

Figure 68 Features: Enable Remote Endpoints-continued

5. **Name** is set to COXServiceProvider for example.
6. Click **OK**

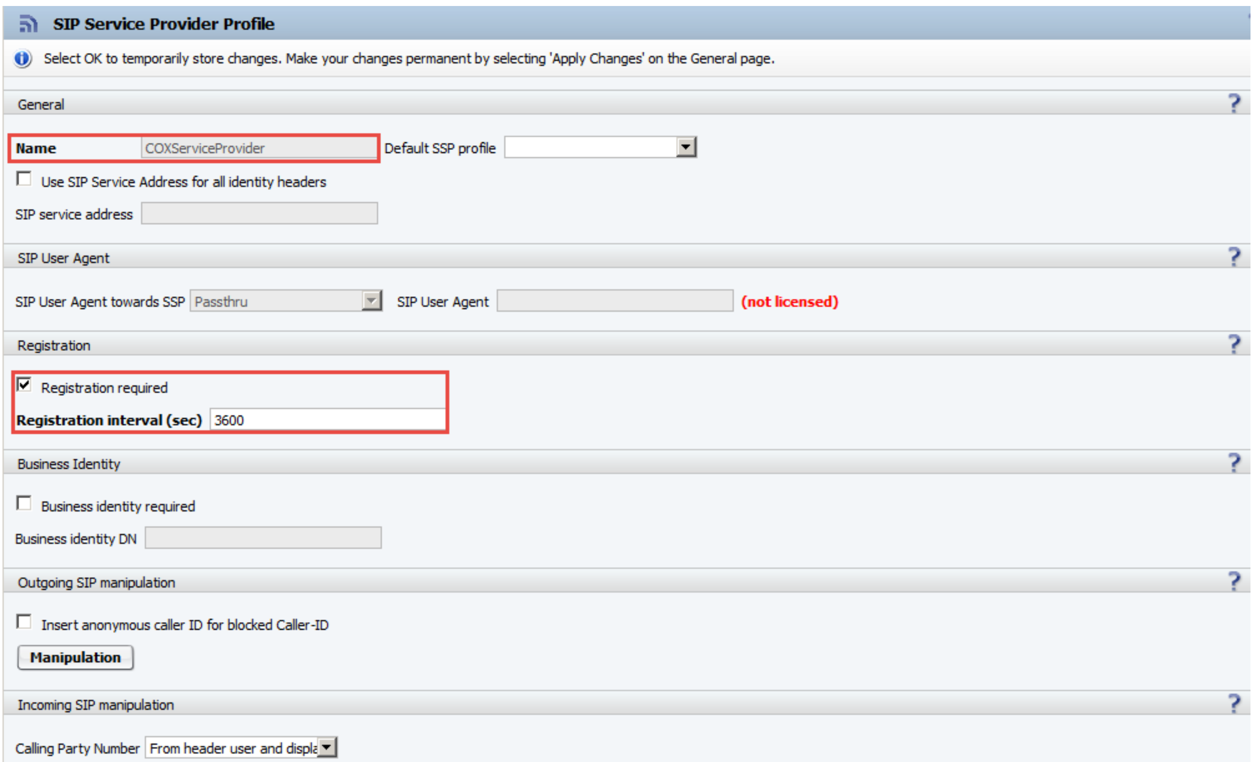

Figure 69 Features: Enable Remote Endpoints-continued

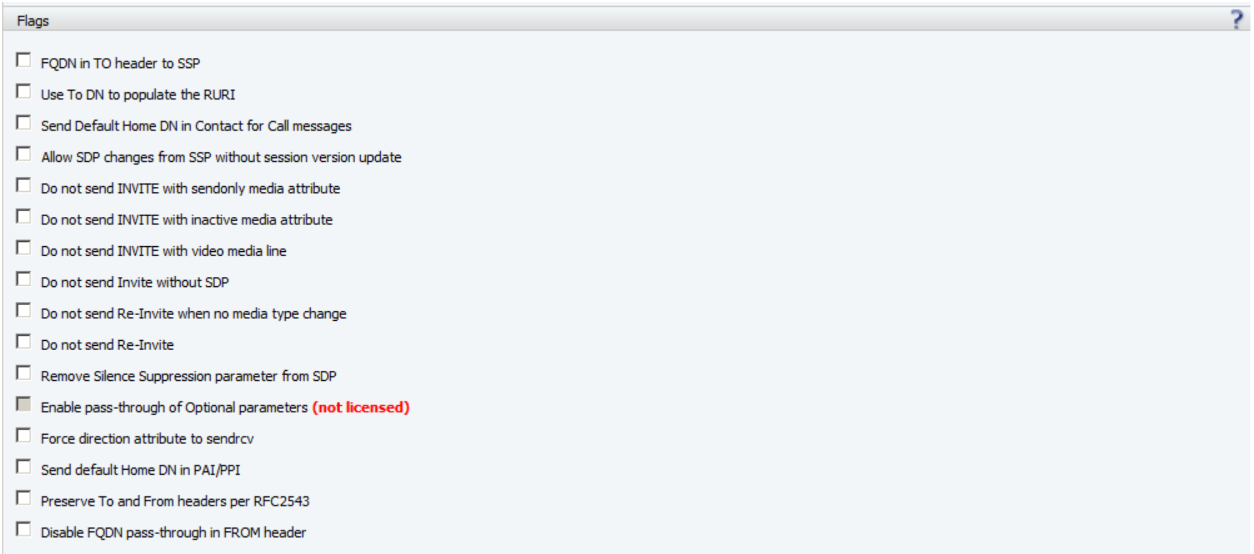

Figure 70 Features: Enable Remote Endpoints-continued

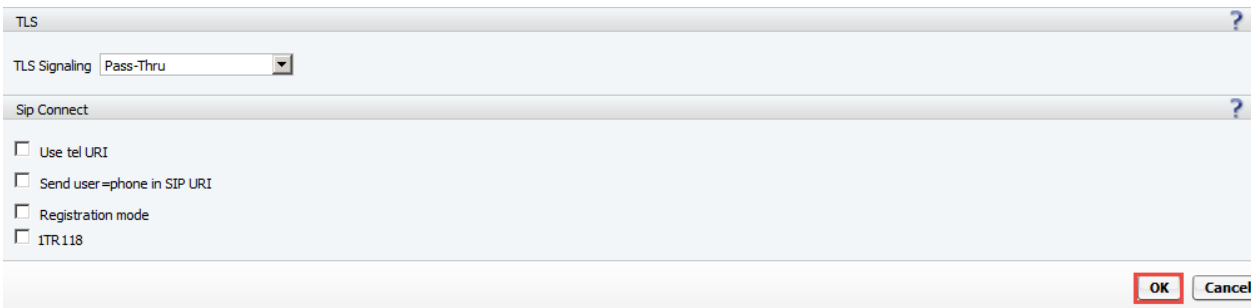

Figure 71 Features: Enable Remote Endpoints-continued

7. In the **Remote endpoint configuration** section click **Add**.

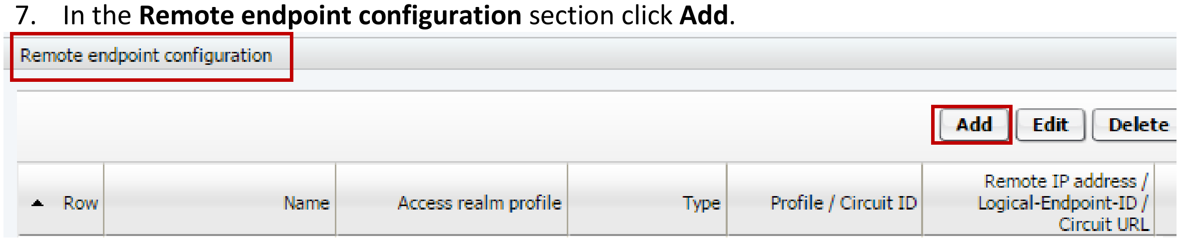

Figure 72 Features: Enable Remote Endpoints-continued

8. **Name** is set to COX_ServiceProvider for example.
9. **Type** is set to SSP for example.
10. **Profile** is set to COXServiceProvider for example.
11. Choose the appropriate **Access realm profile** and **Core realm profile** from the drop down list.
12. Scroll down to **Remote Location domain list** section, click **Add**.
13. **Remote URL** is set to 10.64.4.165 for example. This is the COX provisioned IP address for SIP trunk connection. .
14. **Media IP** is set to 10.64.3.146 for example. This is the Unify SBC's WAN IP address.

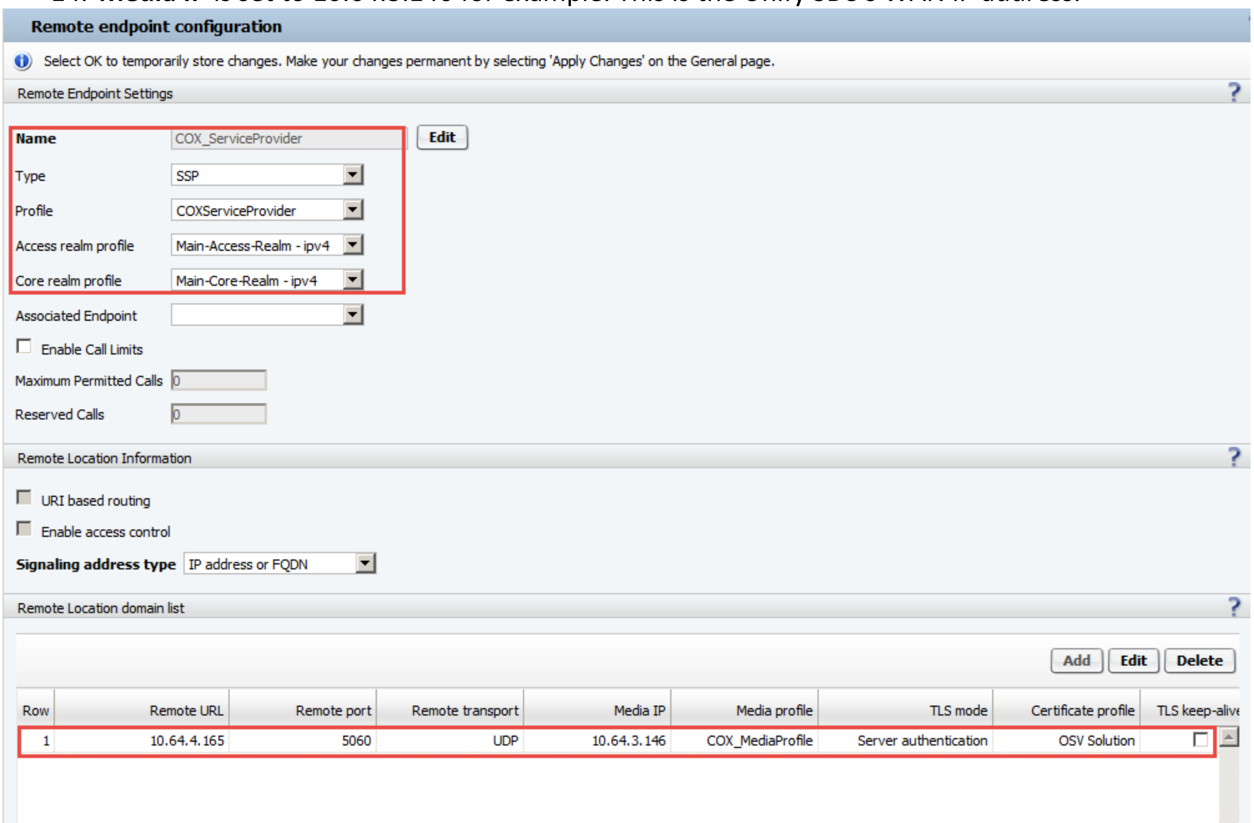

Figure 73 Features: Enable Remote Endpoints-continued

15. In the **Remote Location Identification/Routing** section set **Core realm port** to 50012
16. In the **Digest Authentication** section, Enter required registration details as provided by the Service Provider and click on OK.

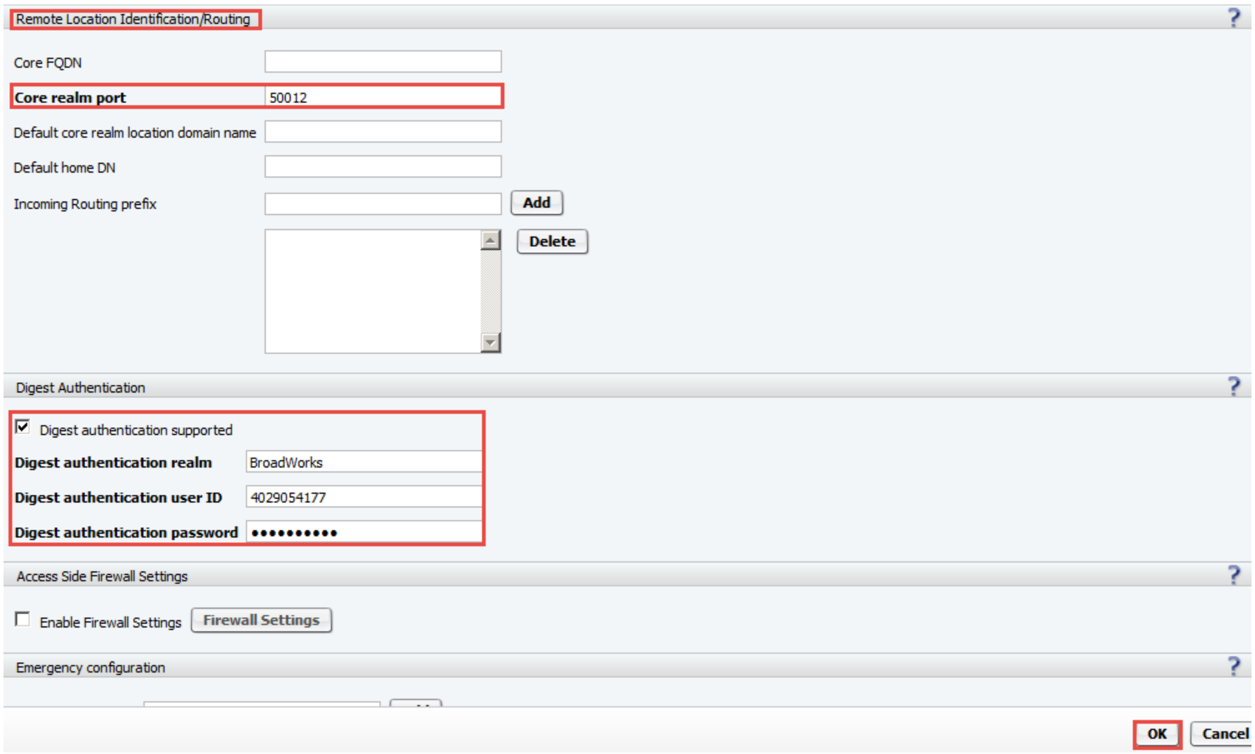

Figure 74 Features: Enable Remote Endpoints-continued

17. On the main screen, click **Apply** changes at bottom right of page.

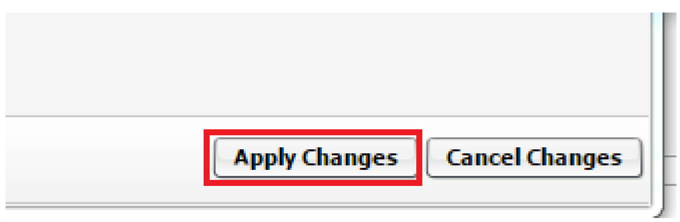

Figure 75 Features: Enable Remote Endpoints-continued

5 Test Results

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
01-01 BC AB1	Basic call	Basic call A to B	A calls B = check display on B side B answers = check display on A side Verify speech path in both directions A clears call = both parties idle again	PASSED	
01-02 BC AB2	Basic call	Basic call A to B	A calls B = check display on B side B answers = check display on A side Verify speech path in both directions B clears call = both parties idle again	PASSED	
01-03 BC BA1	Basic call	Basic call B to A	B calls A = check display on A side A answers = check display on B side Verify speech path in both directions A clears call = both parties idle again	PASSED	
01-04 BC BA2	Basic call	Basic call B to A	B calls A = check display on A side A answers = check display on B side Verify speech path in both directions B clears call = both parties idle again	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
01-05 BC AB CELL	Basic call	Basic call A to B cell phone	Party B is a cell phone subscriber A calls B = check display on B side B answers = check display on A side Verify speech path in both directions B clears call = both parties idle again	PASSED	
01-06 BC BA CELL	Basic call	Basic call A to B cell phone	Party B is a cell phone subscriber B calls A = check display on A side A answers = check display on B side Verify speech path in both directions A clears call = both parties idle again	PASSED	
01-07 BC AB International	Basic call	Basic call A to B International	Party B is an international subscriber (located in another country) A calls B = check ringback tone on A B answers = check display on A side Verify speech path in both directions B clears call = both parties idle again	PASSED	
01-08 BC AB International CELL	Basic call	Basic call A to B International Cell	Party B is an international cell phone subscriber (located in another country) A calls B = check ringback tone on A B answers = check display on A side Verify speech path in both directions	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
			B clears call = both parties idle again		
01-09 BC AB Long Duration Call	Basic call	Long duration call A to B	A calls B = check display on B side B answers = check display on A side Verify speech path in both directions Wait 4 Hours, check speech path A clears call = both parties idle again	FAILED	COX disconnects the call approximately after 1 hour 50 minutes by sending BYE
01-10 BC BA Long Duration Call	Basic call	Long duration call B to A	B calls A = check display on A side A answers = check display on B side Verify speech path in both directions Wait 4 Hours, check speech path B clears call = both parties idle again	PASSED	
01-11 BC AB Mute	Basic call	Basic call A to B Mute	A calls B = check display on B side B answers = check display on A side Verify speech path in both directions Mute call on both ends Wait 30 minutes Verify speech path in both directions again A clears call = both parties idle again	PASSED	
01-12 BC A1A2	Basic call	Basic call via SIP Trunk A1 to A2	Party A1 calls A2 via SIP Trunk A1 calls A2 = check displays and ringback tone B answers = check displays again Verify speech path in both	PASSED	FROM and TO header contains "Unassigned Unassigned" and the display on the phone is "Unassigned Unassigned <DID>"

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
			directions B clears call = both parties idle again		
01-13 BC Emergency Call	Basic call	Emergency call	A calls emergency number (I.E. 911 for the US or 112 for EU) Call center answers, speech path in both directions Call center agent has correct location of A (provided by Carrier) A clears call	PASSED	Calls is made from 402-932-4563 but the CLID displayed to the operator is 402-915-4177 without any address of the Caller.
02-01 BC AB no reply	Basic Call Extended	No Reply A to B	A calls B = check display on B side B does not answer = wait for timeout by provider Verify the call is properly cleared on both sides	PASSED	Timeout is triggered from OpenScape SBC
02-02 BC BA no reply	Basic Call Extended	No Reply B to A	B calls A = check display on B side A does not answer = wait for timeout by provider Verify the call is properly cleared on both sides	PASSED	
02-03 BC AB busy	Basic Call Extended	Busy A to B	A calls busy B = check busy tone/display on A Verify the call is properly cleared	PASSED	
02-04 BC BA busy	Basic Call Extended	Busy B to A	B calls busy A = check busy tone/display on B Verify the call is properly cleared	PASSED	
02-05 BC AB reject	Basic Call Extended	Reject A to B	A calls B = check display on B side B does reject call Verify the call is properly cleared on both sides	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
02-06 BC BA reject	Basic Call Extended	Reject B to A	B calls A = check display on A side A does reject call Verify the call is properly cleared on both sides	PASSED	Executed using DND on PBX. PBX sent announcement.
02-07 BC AB CLIR	Basic Call Extended	CLIR A to B	A with CLIR calls B = check display on B side B answers = check display on A side Verify speech path in both directions A clears call = both parties idle again	PASSED	
02-08 BC BA CLIR	Basic Call Extended	CLIR B to A	B with CLIR calls A = check display on A side A answers = check display on B side Verify speech path in both directions A clears call = both parties idle again	PASSED	
02-09 BC AB invalid CLI	Basic Call Extended	Invalid CLI A to B	A has invalid CLI (Incomplete digits/ Wrong digits) A calls B = check display on B side (displays default CLI) is call goes through B answers = check display on A side Verify speech path in both directions A clears call = both parties idle again	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
03-01 BC AB codec negotiation	Special Basic Call	Codec Negotiation A to B	A has low bandwidth preferred (g.729-0723 high priority, g.711-0722 low priority) B has high quality preferred (g.711-0722 high priority, g.729-0723 low priority) A calls B = check codec proposal B answers = check codec selected A clears call	PASSED	
03-02 BC BA codec negotiation	Special Basic Call	Codec Negotiation B to A	A has low bandwidth preferred (g.729-0723 high priority, g.711-0722 low priority) B has high quality preferred (g.711-0722 high priority, g.729-0723 low priority) B calls A = check codec proposal A answers = check codec selected B clears call	PASSED	
03-03 BC AB G.722	Special Basic Call	G.722 A to B	Enable G.722 and set it as high priority on A and B A calls B = check codec proposal B answers = check codec selected B clears call	NOT SUPPORTED	G722 not supported by DUT
03-04 BC BA G.722	Special Basic Call	G.722 B to A	Enable G.722 and set it as high priority on A and B B calls A = check codec proposal A answers = check codec selected A clears call	NOT SUPPORTED	G722 not supported by DUT

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
03-05 BC AB incompatible codec	Special Basic Call	Incompatible Codec A to B	A has low bandwidth only (g.729-0723) B has high quality only (g.711-0722) A calls B = check codec proposal If the call is now released, the provider does not interwork codec's, check for proper call clearing. If B is ringing: B answers = check codec selected Verify speech path in both directions	NOT SUPPORTED	G729 not supported by DUT
03-06 BC BA incompatible codec	Special Basic Call	Incompatible Codec B to A	A has low bandwidth only (g.729-0723) B has high quality only (g.711-0722) B calls A = check codec proposal If the call is now released, the provider does not interwork codec's, check for proper call clearing If A is ringing: A answers = check codec selected Verify speech path in both directions	NOT SUPPORTED	G729 not supported by DUT

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
03-07 BC AB session timer	Special Basic Call	Session Timer A to B	Enable session timer on both sides (OpenScape Voice and Provider) A calls B B answers Let the call active until the session timer was two times refreshed by both sides B clears call	PASSED	Header "Session-Expires: 1800;refresher=uas " is removed when it is sent out from EdgeMarc to COX. Session refresh happens based on Re-INVITE from OpenScape PBX and SBC. UPDATE as a part of Session Refresh is sent from COX to EdgeMarc but EdgeMarc do not forward it to OpenScape SBC/PBX but acknowledges the UPDATE with 200OK.
03-08 BC BA session timer	Special Basic Call	Session Timer B to A	Enable session timer on both sides (OpenScape Voice and Provider) B calls A B answers Let the call active until the session timer was two times refreshed by both sides A clears call	PASSED	Header "Session-Expires: 1800;refresher=uas " is removed when it is sent out from EdgeMarc to COX. Session refresh happens based on Re-INVITE from OpenScape PBX and SBC. UPDATE as a part of Session Refresh is sent from COX to EdgeMarc but EdgeMarc do not forward it to OpenScape SBC/PBX but acknowledges the UPDATE with 200OK.
03-09 BC A to invalid B	Special Basic Call	Invalid	A calls invalid number = verify proper announcement (or SIP cause) Verify that the call is released properly -If you hear an announcement/tone, check if the	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
			payload is sent before connect (183 progress)		
03-10 BC announcement after connect	Special Basic Call	Announcement after connect	A calls switched off cell phone = A hears announcement before connect Clear call	PASSED	
03-11 BC announcement before connect	Special Basic Call	Announcement before connect	A calls conference bridge = A hears announcement before connect Clear call	PASSED	
03-12 BC Provider Voicemail	Special Basic Call	Provider Voicemail	**This test case assumes that a provider voicemail (VM) service is available** A has VM box on the provider VM server A calls VM server A1hears VM announcement – depending on functionality, A1 should be automatically forwarded to its voicemail box and a PIN is requested then. A enters PIN – A1 is logged into VM box root menu A browses VM menu using keypad A clears call	NOT SUPPORTED	Feature not supported by DUT
04-01 Hold A	Hold/Toggle	Hold A to B - Music on Hold	Establish call A-B A put B in hold = verify MoH (hold indication if possible) on B A retrieve B = verify speech path A-B (display back to normal call if possible) Clear call	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
04-02 Hold B	Hold/Toggle	Hold B to A - Music on Hold	Establish call A-B B put A in hold = verify MoH (hold indication if possible) on A B retrieve A = verify speech path A-B (display back to normal call if possible) Clear call	PASSED	
04-03 Toggle A	Hold/Toggle	Toggle A to B1 to B2 - Music on Hold	Establish call A-B1 A put B1 in hold = verify MoH (hold indication if possible) on B1 A calls B2 B2 answers = verify speech path A toggles between B1 and B2 several times = verify MoH (hold indication if possible) on held party and speech path (display) on active party. Clear call	PASSED	
04-04 Toggle B	Hold/Toggle	Toggle A1 to B to A2 - Music on Hold	Establish call A1-B B put A1 in hold = verify MoH (hold indication if possible) on A1 B calls A2 A2 answers = verify speech path B toggles between A1 and A2 several times = verify MoH (hold indication if possible) on held party and speech path (display) on active party. Clear call	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
04-05 Toggle A1	Hold/Toggle	Toggle A1 to B to A2 - Music on Hold	Establish call A-B A1 put B in hold = verify MoH (hold indication if possible) on B A1 calls A2 A2 answers = verify speech path A toggles between B and A2 several times = verify MoH (hold indication if possible) on held party and speech path (display) on active party. Clear call	PASSED	
05-01 CFU A1/A2	Call Forward	Call Forward A1 to A2	A1 has CFU to A2 B calls A1 = verify A2 is ringing A2 answers = check speech path and display on both parties Clear call	PASSED	
05-02 CFU A/B2	Call Forward	Call Forward A to B2	A has CFU to B2 B1 calls A = verify B2 is ringing B2 answers = check speech path and display on both parties Clear call	PASSED	
05-03 CFU A/B2 BUSY	Call Forward	Call Forward A to B2 Busy	A has CFU to busy B2 B1 calls A = B1 receives busy tone	PASSED	
05-04 CFU B1/B2	Call Forward	Call Forward B1 to B2	B1 has CFU to B2 A calls B1 = verify B2 is ringing B2 answers = check speech path and display on both parties Clear call	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
05-05 CFU B/A2	Call Forward	Call Forward B to A2	B has CFU to A2 A1 calls B = verify A2 is ringing A2 answers = check speech path and display on both parties Clear call	PASSED	
05-06 CFU A/B2 Cell	Call Forward	Call Forward A to B2 Cell phone	A has CFU to B2 = B2 is a Cell phone B1 calls A = verify B2 is ringing B2 answers = check speech path and display on both parties Clear call	PASSED	
05-07 CFU B1/B2 Cell	Call Forward	Call Forward B1 to B2 Cell Phone	B1 has CFU to B2 = B2 is a Cell phone A calls B1 = verify B2 is ringing B2 answers = check speech path and display on both parties Clear call	PASSED	
05-08 CFB A1/A2	Call Forward	Call Forward Busy A1 to A2	A1 has CFB to A2 A1 is busy B calls A1 = verify A2 is ringing A2 answers = check speech path and display on both parties Clear call	PASSED	
05-09 CFB B1/B2	Call Forward	Call Forward Busy B1 to B2	B1 has CFB to B2 B1 is busy A calls B1 = verify B2 is ringing B2 answers = check speech path and display on both parties Clear call	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
05-10 CFNR A1/A2	Call Forward	Call Forward No Response A1 to A2	A1 has CFNR to A2 B calls A1 = verify A1 is ringing A1 does not answer = verify call is forwarded to A2 A2 answers = check speech path and display on both parties Clear call	PASSED	
05-11 CFNR B1/B2	Call Forward	Call Forward No Response B1 to B2	B1 has CFNR to B2 A calls B1 = verify B1 is ringing B1 does not answer = verify call is forwarded to B2 B2 answers = check speech path and display on both parties Clear call	PASSED	
05-12 CFNR A/B2 Busy	Call Forward	Call Forward No Response A to B2 busy	A has CFNR to busy B2 B1 calls A = verify A is ringing A does not answer = verify call is forwarded to B2-B1 receives busy tone	PASSED	
05-13 Call deflect A/B2	Call Forward	Call deflect A to B2	B1 calls A = A is ringing A selects call deflect and dials B2 = verify call is forwarded to B2, A stops ringing B2 answers = check speech path and display on both parties Clear call	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
05-14 Call deflect B1/B2	Call Forward	Call deflect B1 to B2	A calls B1 = B1 is ringing B1 selects call deflect and dials B2 = verify call is forwarded to B2, B1 stops ringing B2 answers = check speech path and display on both parties Clear call	PASSED	Call deflect feature is not available on PSTN phones. Executed the test case by looping back the call to OSV via ITSP.
05-15 Provider Voicemail_CF1	Call Forward	Call forward to voicemail	*This test case assumes that a provider voicemail (VM) service is available* A1 has VM box on the provider VM server A1 sets CF to VM server A2 calls A1 A2 is connected to A1's VM box A2 leaves message A2 clears call VM server sends MWI A1 shows MWI in phone display A1 answers MWI – A1 is connected to VM Box A1 enters PIN A1 retrieves A2's voice message A1 deletes A2's voice message – VM server sends MWI A1's phone clears MWI indication A1 clears call	NOT SUPPORTED	Feature not supported by DUT

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
05-16 Provider Voicemail_CFNR1	Call Forward	Call Forward No Response to voicemail	*This test case assumes that a provider voicemail (VM) service is available* A1 has VM box on the provider VM server A1 sets CFNR to VM server A2 calls A1 A1 does not answer - A2 is connected to A1's VM box A2 leaves message A2 clears call VM server sends MWI A1 shows MWI in phone display A1 answers MWI – A1 is connected to VM Box A1 enters PIN A1 retrieves A2's voice message A1 deletes A2's voice message – VM server sends MWI A1's phone clears MWI indication A1 clears call	NOT SUPPORTED	Feature not supported by DUT

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
05-17 Provider Voicemail_CF2	Call Forward	Call forward to voicemail	*This test case assumes that a provider voicemail (VM) service is available A1 has VM box on the provider VM server* A1 sets CF to VM server B calls A1 - B is connected to A1's VM box B leaves message B clears call VM server sends MWI A1 shows MWI in phone display A1 answers MWI – A1 is connected to VM Box A1 enters PIN A1 retrieves B's voice message A1 deletes B's voice message – VM server sends MWI A1's phone clears MWI indication A1 clears call	NOT SUPPORTED	Feature not supported by DUT

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
05-18 Provider Voicemail_CFNR2	Call Forward	Call Forward No Response to voicemail	B is a cell phone subscriber *This test case assumes that a provider voicemail (VM) service is available A has VM box on the provider VM server* A sets CFNR to VM server B calls A A does not answer - B is connected to A's VM box B leaves message B clears call VM server sends MWI A shows MWI in phone display A answers MWI – A is connected to VM Box A enters PIN A retrieves B's voice message A deletes B's voice message – VM server sends MWI A's phone clears MWI indication A clears call	NOT SUPPORTED	Feature not supported by DUT

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
06-01 Attended CT A1/A2	Call Transfer	Attended Call Transfer A1 to A2	Establish call A1-B A1 put B in hold = verify MoH (hold indication if possible) on B A1 calls A2 = A2 is ringing A2 answers A1 transfers call A2 and B connected = check speech path and display on both parties Clear call	PASSED	
06-02 Attended CT A/B2	Call Transfer	Attended Call Transfer A to B2	Establish call A-B1 A put B1 in hold = verify MoH (hold indication if possible) on B1 A calls B2 = B2 is ringing B2 answers A transfers call B1 and B2 connected = check speech path and display on both parties Clear call	PASSED	
06-03 Attended CT B1/B2	Call Transfer	Attended Call Transfer B1 to B2	Establish call A-B1 B1 put A in hold = verify MoH (hold indication if possible) on A B1 calls B2 = B2 is ringing B2 answers B1 transfers call A and B2 connected = check speech path and display on both parties Clear call	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
06-04 Attended CT B/A2	Call Transfer	Attended Call Transfer B to A2	Establish call A1-B B put A1 in hold = verify MoH (hold indication if possible) on A1 B calls A2 = A2 is ringing A2 answers B transfers call A1 and A2 connected = check speech path and display on both parties Clear call	PASSED	
06-05 Semi attended CT A1/A2	Call Transfer	Semi Attended Call Transfer A1 to A2	Establish call A1-B A1 put B in hold = verify MoH (hold indication if possible) on B A1 calls A2 = A2 is ringing A1 transfers call before A2 answers = B hears ringback tone now A2 answers A2 and B connected = check speech path and display on both parties Clear call	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
06-06 Semi attended CT A/B2	Call Transfer	Semi Attended Call Transfer A to B2	Establish call A-B1 A put B1 in hold = verify MoH (hold indication if possible) on B1 A calls B2 = B2 is ringing A transfers call before B2 answers = B1 hears ringback tone now B2 answers B1 and B2 connected = check speech path and display on both parties Clear call	PASSED	
06-07 Semi attended CT B1/B2	Call Transfer	Semi Attended Call Transfer B1 to B2	Establish call A-B1 B1 put A in hold = verify MoH (hold indication if possible) on A B1 calls B2 = B2 is ringing B1 transfers call before B2 answers = A hears ringback tone now B2 answers B2 and A connected = check speech path and display on both parties Clear call	PASSED	
06-08 Blind CT A/B2	Call Transfer	Blind Call Transfer A to B2	Establish call A-B1 A selects blind transfer and dials B2 = B2 is ringing, check displays B2 answers = speech path and display on both parties Clear call	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
06-09 Blind CT B/A2	Call Transfer	Blind Call Transfer B to A2	Establish call A1-B B selects blind transfer and dials A2 = A2 is ringing, check displays A2 answers = speech path and display on both parties Clear call	PASSED	
07-01 Conference - Media Server (MS)	Conference	Conference to a Media Server (MS)	A1 has large conference configured Establish call A1-B1 A1 put B1 in hold A1 dials A2 A2 answers A1 selects conference A1 selects hold, dials B2 B2 answers A1 selects add to conference A1, A2, B1 and B2 in conference = check speech path and display on both parties Clear calls	PASSED	
07-02 Conference - local phone	Conference	Conference to a local phone	A1 has local (phone) conference configured Establish call A1-B1 A1 put B1 in hold A1 dials A2 A2 answers A1 selects conference A1, A2 and B1 in conference = check speech path and display on both parties Clear calls	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
08-01 Fax t.38 AB	Fax and DTMF	Fax t.38 A to B	A and B are represented as Fax machines in this test case A and B use t.38 for fax calls A calls B B answers Codec change to t.38 is initiated Several pages of documents are sent over the connection A releases automatically the call after all pages are sent	PASSED	
08-02 Fax t.38 BA	Fax and DTMF	Fax t.38 B to A	A and B are represented as Fax machines in this test case A and B use t.38 for fax calls B calls A A answers Codec change to t.38 is initiated Several pages of documents are sent over the connection B releases automatically the call after all pages are sent	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
08-03 Fax G.711 AB	Fax and DTMF	Fax G.711 A to B	A and B are represented as Fax machines in this test case A and B have low bandwidth codec as high priority, high quality codec as low priority A calls B B answers = the call is with low bandwidth established (G.729 or G.723) Codec change to G.711 is initiated Several pages of documents are sent over the connection A releases automatically the call after all pages are sent	PASSED	Voice call is negotiated on G711u. Fax Re-Invite with t38 is sent from COX but is rejected by Fax machine as "488 not acceptable here" and the fax is sent on G711u passthrough.
08-04 Fax G.711 BA	Fax and DTMF	Fax G.711 B to A	A and B are represented as Fax machines in this test case A and B have low bandwidth codec as high priority, high quality codec as low priority B calls A A answers = the call is with low bandwidth established (G.729 or G.723) Codec change to G.711 is initiated Several pages of documents are sent over the connection B releases automatically the call after all pages are sent	PASSED	COX supports only G711u law codec

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
08-05 Fax G.711 AB AhiBhi	Fax and DTMF	Fax G.711 High Speed A to High Speed B	A and B are represented as Fax machines in this test case A and B are high-speed (G3+) devices A calls B B answers Several pages of documents are sent over the connection A releases automatically the call after all pages are sent	NOT APPLICABLE	Only G3 speeds are applicable for COX
08-06 Fax G.711 BA BhiAhi	Fax and DTMF	Fax G.711 High Speed B to High Speed C	A and B are represented as Fax machines in this test case A and B are high-speed (G3+) devices B calls A B answers Several pages of documents are sent over the connection B releases automatically the call after all pages are sent	NOT APPLICABLE	Only G3 speeds are applicable for COX
08-07 Fax G.711 AB AloBhi	Fax and DTMF	Fax G.711 Low Speed A to High Speed B	A and B are represented as Fax machines in this test case A is a low speed (G3) and B is a high-speed (G3+) device A calls B B answers Several pages of documents are sent over the connection A releases automatically the call after all pages are sent	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
08-08 Fax G.711 BA BloAhi	Fax and DTMF	Fax G.711 Low Speed B to High Speed A	A and B are represented as Fax machines in this test case A is a low speed (G3) and B is a high-speed (G3+) device B calls A A answers = the call is with low bandwidth established Several pages of documents are sent over the connection B releases automatically the call after all pages are sent	PASSED	
08-09 Fax G.711 AB AhiBlo	Fax and DTMF	Fax G.711 High Speed A to Low Speed B	A and B are represented as Fax machines in this test case A is a high-speed (G3+) and B is a low speed (G3) device A calls B B answers Several pages of documents are sent over the connection A releases automatically the call after all pages are sent	PASSED	
08-10 Fax G.711 BA BhiAlo	Fax and DTMF	Fax G.711 High Speed B to Low Speed A	A and B are represented as Fax machines in this test case A is a high-speed (G3+) and B is a low speed (G3) device B calls A A answers = the call is with low bandwidth established Several pages of documents are sent over the connection	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
			B releases automatically the call after all pages are sent		
08-11 DTMF RFC2833 AB	Fax and DTMF	DTMF RFC2833 A to B	Establish call A-B A dials digits after connect = verify that the digits are sent as own payload type B dials digits after connect = verify that the digits are sent as own payload type Clear call	PASSED	
08-12 DTMF RFC2833 BA	Fax and DTMF	DTMF RFC2833 B to A	Establish call B-A B dials digits after connect = verify that the digits are sent as own payload type A dials digits after connect = verify that the digits are sent as own payload type Clear call	PASSED	
08-13 DTMF in band AB	Fax and DTMF	DTMF in band A to B	Disable RFC2833 on both parties (if possible) If possible replace party B by voicemail or anything else with DTMF recognition Establish call A-B A dials digits after connect = verify that the digits are sent as the same payload like the voice Clear call	NOT APPLICABLE	OSV does not support Inband DTMF provisioning.

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
08-14 DTMF in band BA	Fax and DTMF	DTMF in band B to A	Disable RFC2833 on both parties (if possible) If possible replace party A by voicemail or anything else with DTMF recognition Establish call B-A B dials digits after connect = verify that the digits are sent as the same payload like the voice Clear call	NOT APPLICABLE	OSV does not support Inband DTMF provisioning.
08-15 DTMF RFC2833 before connect	Fax and DTMF	DTMF RFC2833 before connect	Re-Enable RFC2833 on both parties If possible replace party B by voicemail or anything else with DTMF recognition before answer A calls B A gets announcement before connect A dials digits = B recognizes digits (I.E. forwards A to voice mail) B answers = check display on A side Verify speech path in both directions A clears call = both parties idle again	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
09-01 MLHG A3	OpenScape Voice features	MLHG A3	Configure MLHG with parties A1, A2 and A3 B calls MLHG A1 is ringing A2 is ringing = A1 stops ringing A3 is ringing = A2 stops ringing A3 answers = check speech path and display on both parties Clear call	PASSED	
09-02 MLHG A1	OpenScape Voice features	MLHG A1	Configure MLHG with parties A1, A2 and A3 B calls MLHG A1 answers = check speech path and display on both parties Clear call	PASSED	
09-03 Pickup Group A2	OpenScape Voice features	Pickup Group A2	Configure Pickup Group with parties A1, A2 and A3 B calls A1 =A1 is ringing, A2 and A3 have display notification A2 picks up call = check speech path and display on both parties Clear call	PASSED	
09-04 Pickup Group A1	OpenScape Voice features	Pickup Group A1	Configure Pickup Group with parties A1, A2 and A3 B calls A1 =A1 is ringing, A2 and A3 have display notification A1 answers call = check speech path and display on both parties Clear call	PASSED	

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
09-05 ONS A1(A2)B1	OpenScape Voice features	ONS A1(A2)B1	A1 is/has OpenScape UC user A1 OpenScape UC selects A2 as preferred device A1 calls B1 via ODC or OWC A2 rings/auto answers (answer manually when ringing) B1 rings and shows A1 in Display B1 answers = talks to A2, but A1 in Display Wait 20 minutes, check speech path regularly Clear call	NOT TESTED	License not installed to support ONS
09-06 ONS A1(B1)A2	OpenScape Voice features	ONS A1(B1)A2	A1 is/has OpenScape UC user A1 OpenScape UC selects B1 as preferred device A1 calls A2 via ODC or OWC = B1 rings B1 answers = A2 rings and shows A1 in Display A2 answers = A2 and B1 connected, A1 in B1's Display Wait 20 minutes, check speech path regularly Clear call	NOT TESTED	License not installed to support ONS
09-07 ONS A1(B1)B2	OpenScape Voice features	ONS A1(B1)B2	A1 is/has OpenScape UC user A1 OpenScape UC selects B1 as preferred device A1 calls B2 via ODC or OWC = B1 rings B1 answers = B2 rings and shows A1 in Display	NOT TESTED	License not installed to support ONS

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
			B2 answers = talks to B1, but A1 in Display Clear call		
09-08 ONS A1(B1)B2 CELL1	OpenScape Voice features	ONS A1(B1)B2	A1 is/has OpenScape UC user B1 is of type "cell phone" A1 OpenScape UC selects B1 as preferred device A1 calls B2 via ODC or OWC = B1 rings B1 answers = B2 rings, shows A1 in Display B2 answers = B2 and B1 connected, A1 in B2-Display Wait 20 minutes, check speech path regularly Clear call	NOT TESTED	License not installed to support ONS
09-09 ONS A1(B1)B2 CELL2	OpenScape Voice features	ONS A1(B1)B2	A1 is/has OpenScape UC user B2 is of type "cell phone" A1 OpenScape UC selects B1 as preferred device A1 calls B2 via ODC or OWC = B1 rings B1 answers = B2 starts ringing, shows A1 in Display B2 answers = B2 connected to B1, but A1 in B2's Display Clear call	NOT TESTED	License not installed to support ONS

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
10-01 BC AB1	Branch Subscriber	BC A to B1	A is a branch subscriber A calls B = check display on B side B answers = check display on A side Verify speech path in both directions A clears call = verify on both parties a successful the end of the call	NOT TESTED	Branch Office is not part of the setup
10-02 BC AB2	Branch Subscriber	BC A to B2	A is a branch subscriber A calls B = check display on B side B answers = check display on A side Verify speech path in both directions B clears call = verify on both parties a successful the end of the call	NOT TESTED	Branch Office is not part of the setup
10-03 CFU A1/A2	Branch Subscriber	CFU A1 to A2	A1 (in the headquarter) has CFU to A2 (branch subscriber) B calls A1 = verify A2 is ringing, check display both parties A2 answers = check speech path and display on both parties Clear call	NOT TESTED	Branch Office is not part of the setup
10-04 CFU A2-B	Branch Subscriber	CFU A2 to B	A2 (branch subscriber) has CFU to B A1 (in the headquarter) calls A2 = verify B is ringing Check B’s Display for calling and diverting numbers A2 answers = check speech path and display on both parties Clear call	NOT TESTED	Branch Office is not part of the setup

Test Case ID	Test Case	Description	Procedure	Status (Passed or Failed etc)	Observations
10-05 Toggle A1	Branch Subscriber	Toggle A1	A2 (branch subscriber, A1 (in the headquarter) Establish connection A2-B A2 put B in hold = verify MoH (hold indication if possible) on B A2 calls A1 A1 answers = verify speech path A2 toggles between B and A1 several times = verify MoH (hold indication if possible) on held party and speech path (display) on active party. Clear call	NOT TESTED	Branch Office is not part of the setup
10-06 Attended CT A1/A2	Branch Subscriber	Attended CT A1 to A2	A1 is a branch subscriber, A2 is located in the headquarter Establish call A1-B A1 put B in hold = verify MoH (hold indication if possible) on B A1 calls A2 = A2 is ringing A2 answers A1 transfers call A2 and B connected = check speech path and display on both parties Clear call	NOT TESTED	Branch Office is not part of the setup