



OpenScape Business V3

HowTo

Skype for Business Interworking
SIP Trunking and Gateway

Version 1.0

Definitions

HowTo

A HowTo describes the configuration of an feature within the administration of the system. It addresses primarily trained administrators.

Tutorial

Within the tutorials procedures for installation, administration and operation of specific devices, applications or 3rd party systems, which are connected to the system, are described. The tutorial addresses primarily trained administrators.

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Table of History

Date	Version	Changes
2020-06-23	1.0	initial version

Disclaimer:

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The Skype for Business examples in this document give a rough overview of needed components in a basic setup. For detailed information and needed Software and Hardware requirements for Microsoft Skype for Business, licenses resp. license bundles and administration of Microsoft Skype for Business please contact Microsoft or your Microsoft Integration Partner.

Please note:

Unify offers voice interworking capabilities with Microsoft Skype for Business with a technical description of how to configure the OpenScape Business. Microsoft Skype for Business is a 3rd party product. Therefore UNIFY doesn't deliver any administration for Skype for Business. This is up to the responsibility of the Microsoft Integration Partner.

References

- [1] Deploy Skype for Business Server 2015
<https://technet.microsoft.com/en-us/library/dn933893.aspx>
<https://docs.microsoft.com/en-us/skypeforbusiness/deploy-1/deploy-overview>

- [2] Microsoft Infrastructure Interoperability Program
<https://technet.microsoft.com/en-us/office/dn947483>







- [3] OpenScape Business, Installation Guide

- [4] OpenScape Business, Administrator Documentation

- [5] OpenScape Business, Tutorial VoIP Interfaces
http://wiki.unify.com/images/8/8c/How_To_Configure_LAN_WAN_Interface_for_VoIP.pdf

1. Introduction

OpenScape Business V3 following systems support “Skype for Business Interworking” and are in scope of this document:

OpenScape Business X1	OpenScape Business X3	OpenScape Business X5	OpenScape Business X8	UC Booster Server	OpenScape Business S
Rack Version	Rack Version	Rack Version	Rack Version	Rack Version	Rack Version
n/a				n/a	n/a
Wall Version	Wall Version	Wall Version	Wall Version	Wall Version	Wall Version
n/a			n/a	n/a	n/a
Server-based solution	Server-based solution	Server-based solution	Server-based solution	Server-based solution	Server-based solution
n/a	n/a	n/a	n/a	n/a	

To provide non-secure media Interworking (Microsoft Interoperability cases: SIP Trunking and Gateway) between OpenScape Business (OSBiz) and Microsoft Skype for Business (SfB) several components are used and need to be configured.

The minimal solution consists of the Skype for Business Standard Edition server, a Skype for Business Mediation server and the OpenScape Business communication system. If there is a need for a Skype for Business Enterprise Edition, then two servers are needed; a SfB Frontend and a SfB Backend server. Additionally the Mediation server can be colocated with the Standard Edition server or the Frontend server.

- Interworking with Microsoft Skype for Business requires a Project Specific Release

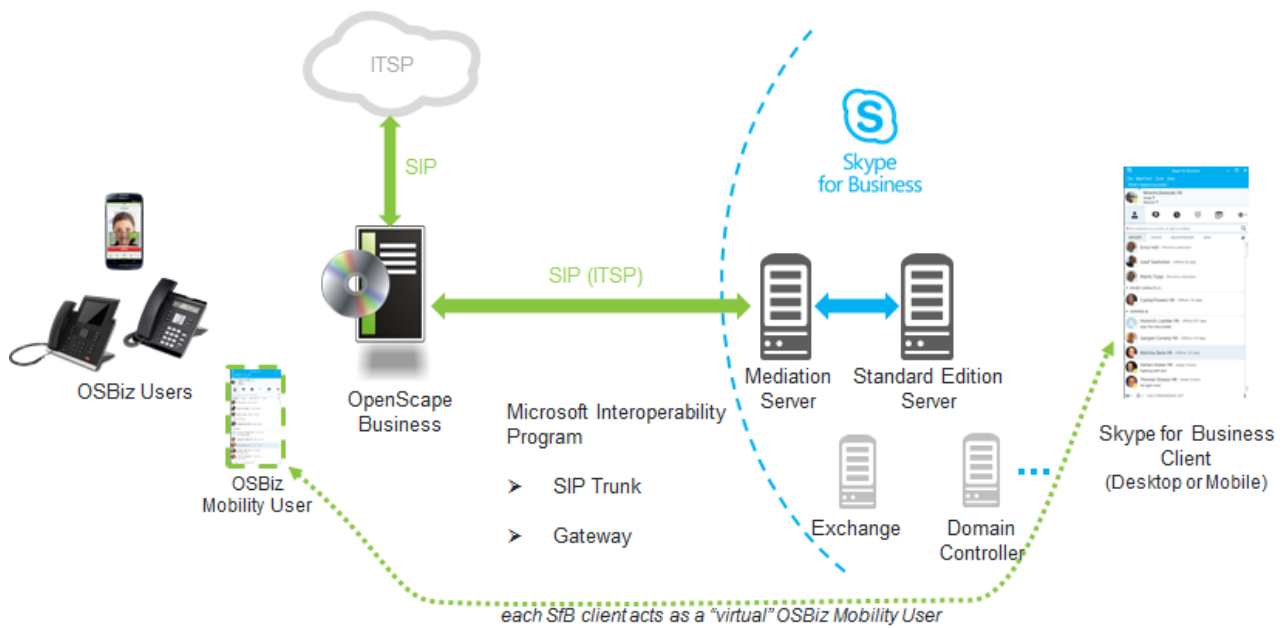
- Skype for Business Interworking is only provided for OpenScape Business systems which are under valid SW Support

- Skype for Business Interworking is handled like an ITSP connection
 - new „Skype for Business” ITSP template for easy configuration
 - ITSP trunk channel licenses are required in OpenScape Business
 - multiple networked OpenScape Business systems can interwork with a Skype for Business installation
 - access to „Skype for Business” ITSP Profile via the LAN interface only
 - the OpenScape Business „Skype for Business” ITSP Profile supports:
 - Skype for Business Server via Mediation Server
 - Office 365 via Skype for Business Cloud Connector Edition

- Skype for Business clients are configured as virtual OpenScape Business users of new type „Skype for Business“
 - Mobility / IP User license required per Skype for Business user
 - the virtual Skype for Business users can operate standalone or can be added to a Mobility group / MULAP (One Number Service)
 - outgoing calls from Skype for Business client use OpenScape Business ONS number
 - Class of Service / traffic restrictions are checked by OpenScape Business (Basic Call and Consultation Call)
 - parallel ringing to desk phone and Skype for Business client for inbound calls
 - internal calls: just dial short numbers in both directions
 - Skype for Business client can perform call forwarding, transfers and conferences in such cases the traffic restriction of the mobile User applies
 - Skype for Business Conferences are accessible via OpenScape Business
 - OpenScape Business supports Email forwarding to Skype for Business of missed calls and voicemail calls

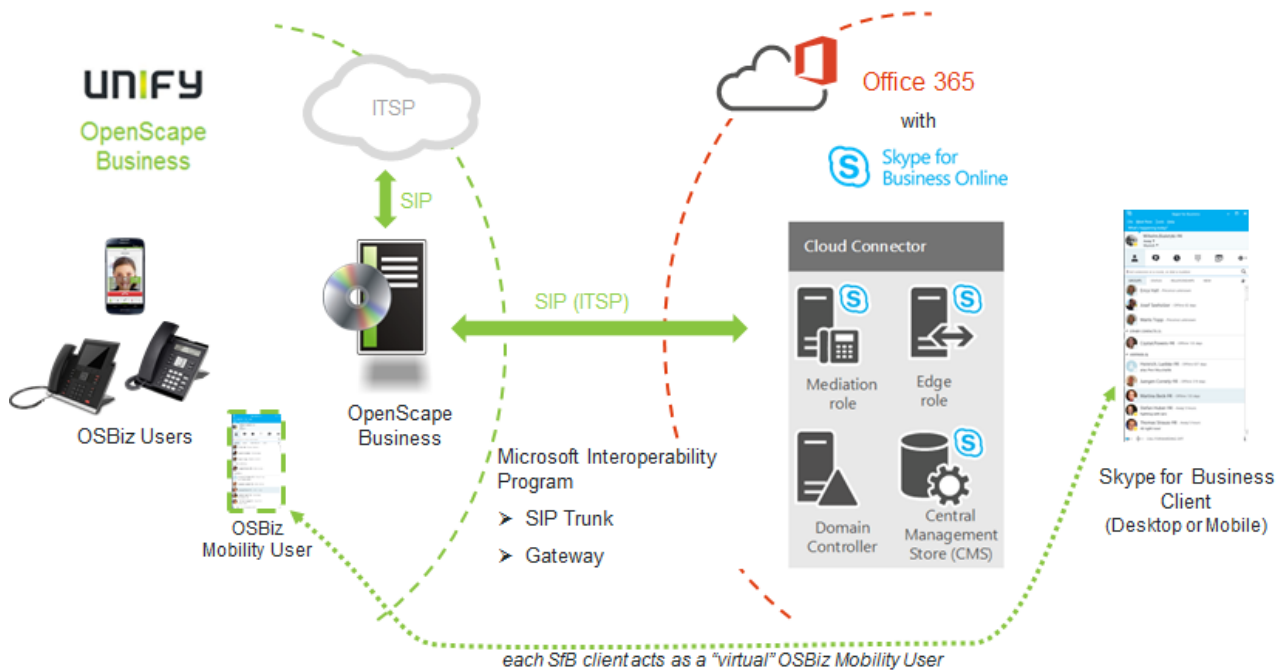
➤ **Skype for Business Server 2015 / 2019**
(Mediation Server + Standard Edition Server „on-premise“)

- Any Skype for Business deployment needs a domain controller server (with DNS). There is also the need for a certificate authority to issue the certificates during SfB deployment setup phase.
- For Microsoft Skype for Business, please check required Licenses (Server and CAL-Plus Licenses) together with the Skype for Business Integration Partner of the Customer.
<https://products.office.com/en-us/skype-for-business/it-pros>



Basic Scenario: Interworking with Skype for Business 2015 / 2019

➤ Office 365 (Skype for Business Cloud Connector Edition „on-premise“)



Basic Scenario: Interworking with Office 365

- from OpenScape Business perspective the SIP connection to the Cloud Connector Edition is the same as for the Skype for Business Mediation Server
- if OpenScape Business acts as Voice Gateway the Add-on services Office 365 Calling Plan and PSTN Conferencing might not be needed
- according to Microsoft recommendations the Cloud Connector Edition must be deployed on dedicated hardware running Windows Server 2012 R2 Datacenter edition (English) with the Hyper-V role enabled. For example to deploy a small version of Cloud Connector Edition that supports up to 50 simultaneous calls the following hardware is required:
 - Intel i7 4790 quad core with Intel 4600 Graphics

please check actual details under:

<https://technet.microsoft.com/en-us/library/mt605227.aspx>

Please note: these recommendations are based on Microsoft requirements and needs to be verified before starting with the project.

- the User needs the Phone System subscription

which is included in Office Enterprise E5

<https://products.office.com/en-us/business/office-365-enterprise-e5-business-software>

or can be obtained as add-on for Office Enterprise E3

<https://products.office.com/en-us/business/office-365-enterprise-e3-business-software>

or can be obtained as as add-on for Office Enterprise E1

<https://products.office.com/en-us/business/office-365-enterprise-e1-business-software>

- Further general details to Microsoft information are available here:

<https://technet.microsoft.com/en-us/library/skype-for-business-online-service-description.aspx>

- Overview of **Office 365 License** which can be obtained to connect Cloud Connector Edition with OpenScape Business (status June 2018 – source: Microsoft)

License	Add-on
Office 365 Enterprise E5	
Office 365 Enterprise E3	Phone System
Office 365 Enterprise E1	Phone System

or

or

2. Skype for Business Server

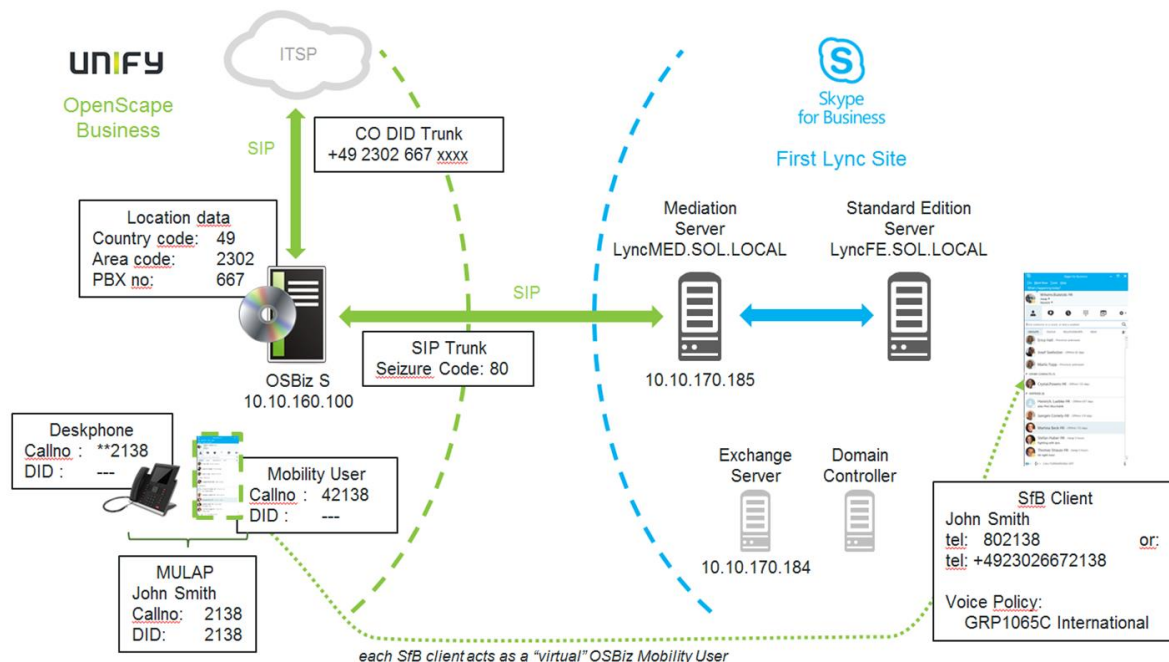
This section outlines how to configure Microsoft Skype for Business Server 2015 to operate with OpenScape Business.



All configuration examples mentioned in this document are valid for the Skype for Business Cloud Connector Edition (Office 365) as well.

Skype for Business deployment configuration activities to support enterprise voice are also necessary but beyond the scope of this document (Ref. [1],[2]).

The following picture gives an overview of the example topology.



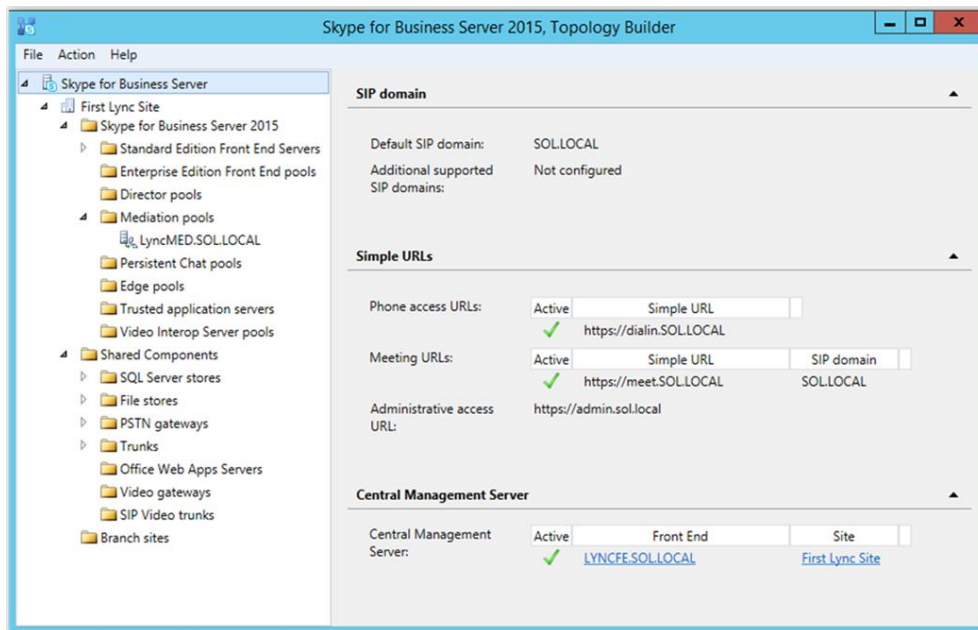
Example Topology

Ensure with the **“Set-CsTrunkConfiguration”** command at the Skype for Business Server Management Shell that the following parameters are set accordingly for the Skype for Business Server trunk connection to the OpenScape Business:

- “EnableReferSupport”** set **“False”**
- “RTCPActiveCalls”** set **“False”**
- “EnableSessionTimer”** set **“True”**

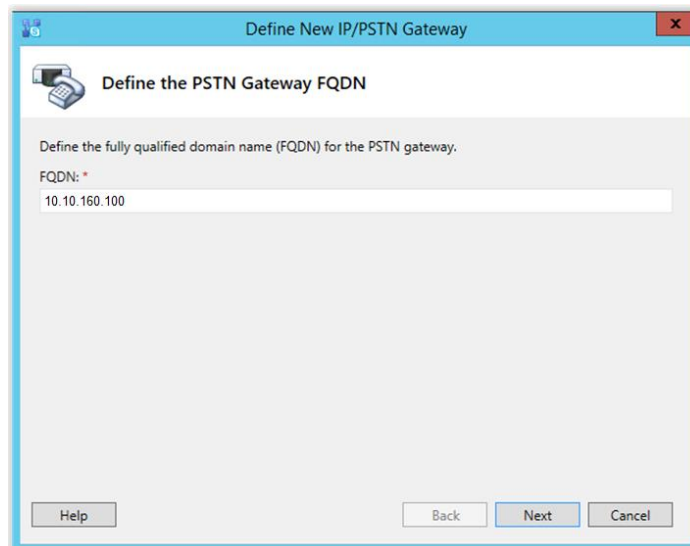
2.1. Configuring Open Scape Business as an IP/PSTN Gateway

1. Go to Skype for Business server and open [Skype for Business Topology Builder](#).
2. Select the [Download Topology from existing deployment](#) option and click on [OK]; you're prompted to save the downloaded topology.
3. Enter a name for the topology file and click on [Save]. This step enables you to roll back from any changes you make during the installation. The [Topology Builder](#) screen with the downloaded topology is displayed:



Topology Builder Displaying Downloaded Topology

4. In the tree, expand Skype for Business Server 2015 > your site name > Shared Components.
5. Right-click the [PSTN Gateways](#) folder and select [New IP/PSTN Gateway](#) from the popup menu. This dialog opens:



Define New IP/PSTN Gateway

6. Enter the IP address of OSBiz (e.g. 10.10.160.100). The user could use FQDN but this should be resolvable, so a corresponding DNS record should be created. Click on [Next].
7. Define the listening mode IPv4 of the IP address of your new PSTN gateway and click on [Next].
8. On the next window click on [Next].
9. Define a **root trunk** for the PSTN gateway. A trunk is a logical connection between a Mediation Server and a gateway, uniquely identified by the combination {Mediation Server FQDN, Mediation Server listening port (TLS or TCP): gateway IP and FQDN, gateway listening port}
 - When defining a PSTN gateway in **Topology Builder**, you must define a **root trunk** to successfully add the PSTN gateway to your topology.
 - The **root trunk** cannot be removed until the associated PSTN gateway is removed.
 - Mediation Server's default port for TCP is 5068 and for TLS is 5067. These can be changed if the user edits the Skype for Business Mediation pool and edits the PSTN gateways properties.

Define New IP/PSTN Gateway

Define the root trunk

Trunk name: *
10.10.160.100

Listening port for IP/PSTN gateway: *
5060

SIP Transport Protocol:
TCP

Associated Mediation Server:
LyncMED.SOLLOCAL First Lync Site

Associated Mediation Server port: *
5068

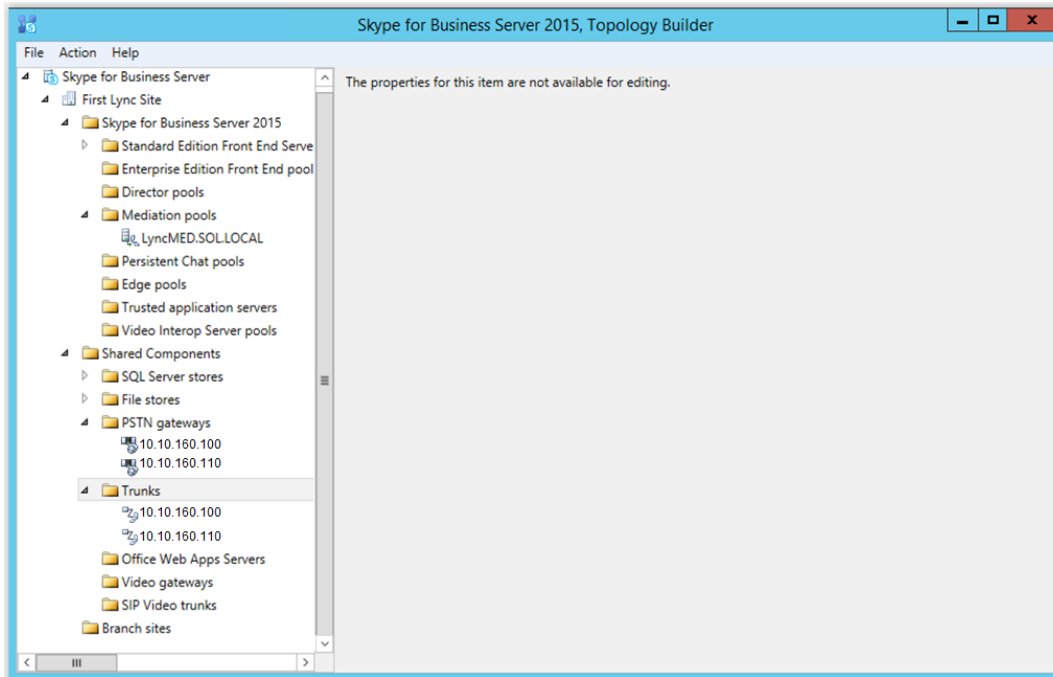
Help Back Finish Cancel

Define the Root Trunk

- In the '[Listening Port for IP/PSTN Gateway](#)' field, type the listening port that the OpenScape Business will use for SIP messages from the Mediation Server that will be associated with the root trunk of the PSTN gateway (i.e., 5060).
- In the '[SIP Transport Protocol](#)' field, click the transport type TCP that the trunk uses.
- In the '[Associated Mediation Server](#)' field, select the Mediation Server pool to associate with the root trunk of this PSTN Gateway.
- In the '[Associated Mediation Server port](#)' field, type the listening port that the Mediation Server will use for SIP messages from the Open Scape Business (i.e., 5068).

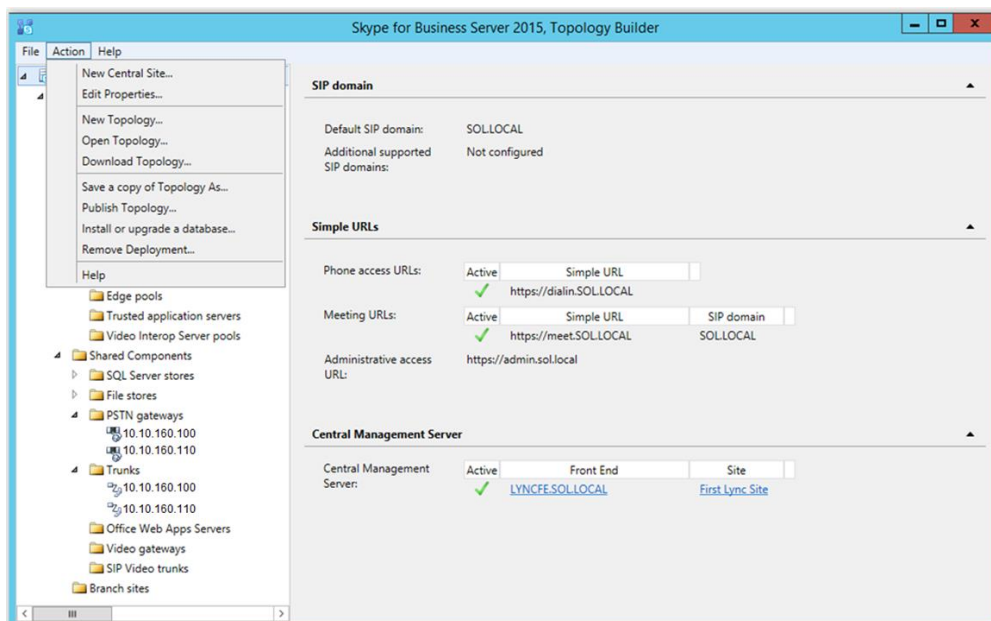
Click on [Finish];

10. Repeat the steps 1-9 to add another Open Scape Business node (e.g. 10.10.160.110). The both Open Scape Business nodes are added as a PSTN Gateways and the corresponding trunks are created:



Open Scape Business nodes added as an IP/PSTN Gateways and Trunks Created

11. Publish the Topology: In the main tree, select the root item **Skype for Business Server** and from the **Action** menu, select **Publish Topology**:

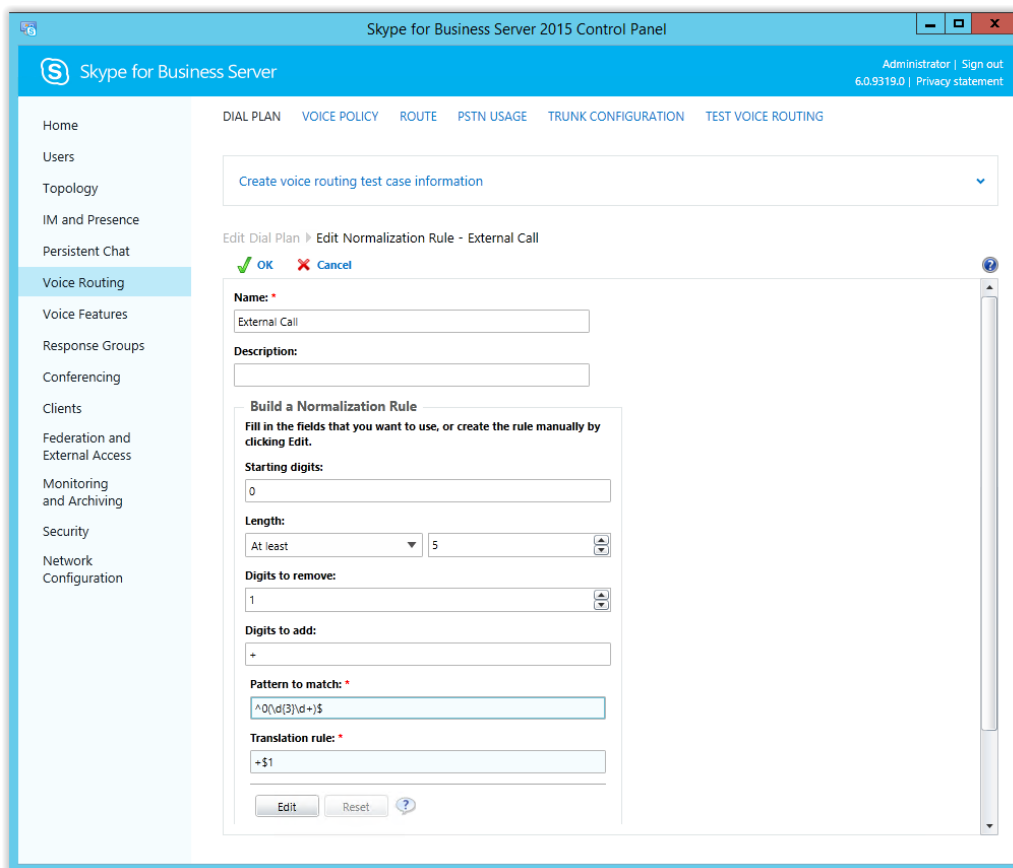


Selecting 'Publish Topology' from the 'Action' Menu

12. Click on [Next] on the [Publish Topology](#) screen; the [Topology Builder](#) starts publishing your topology.
13. Wait for the publishing topology process to successfully complete and click on [Finish].

2.2. Configuring Voice Routing on Skype for Business Server 2015

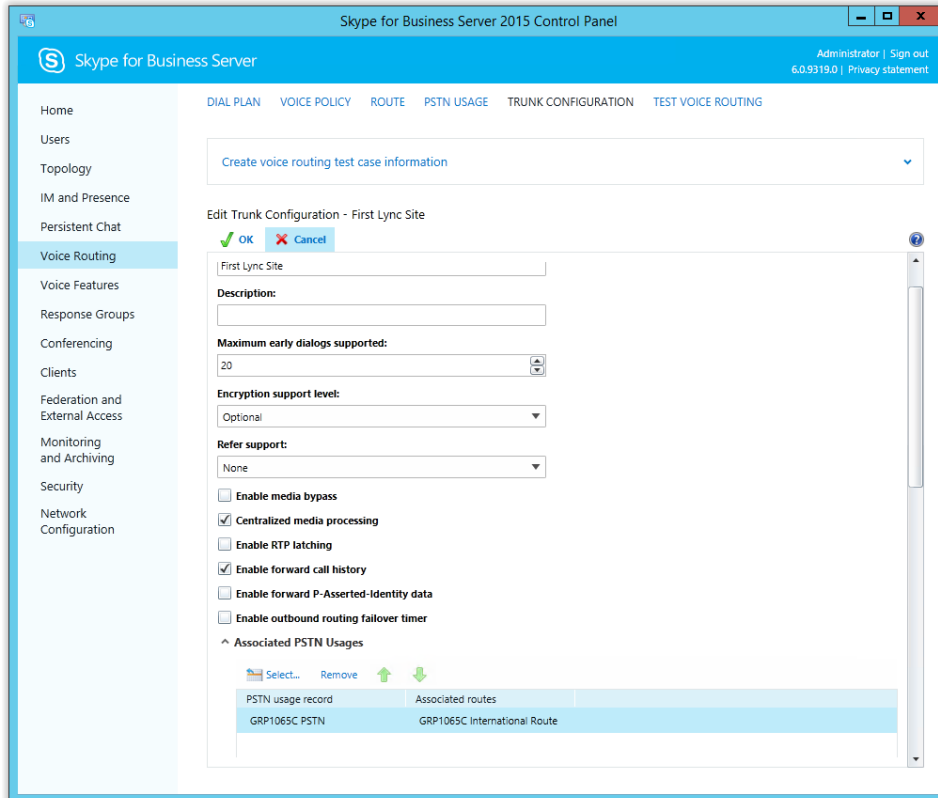
1. Run [Skype for Business Server Control Panel](#) and select [Voice Routing](#).
2. A dial plan could be created that handles the calls from users in Skype for Business domain to users to Open Scape Business. A dial plan takes user-dialed numbers and formats (“normalizes”) them for Skype for Business.



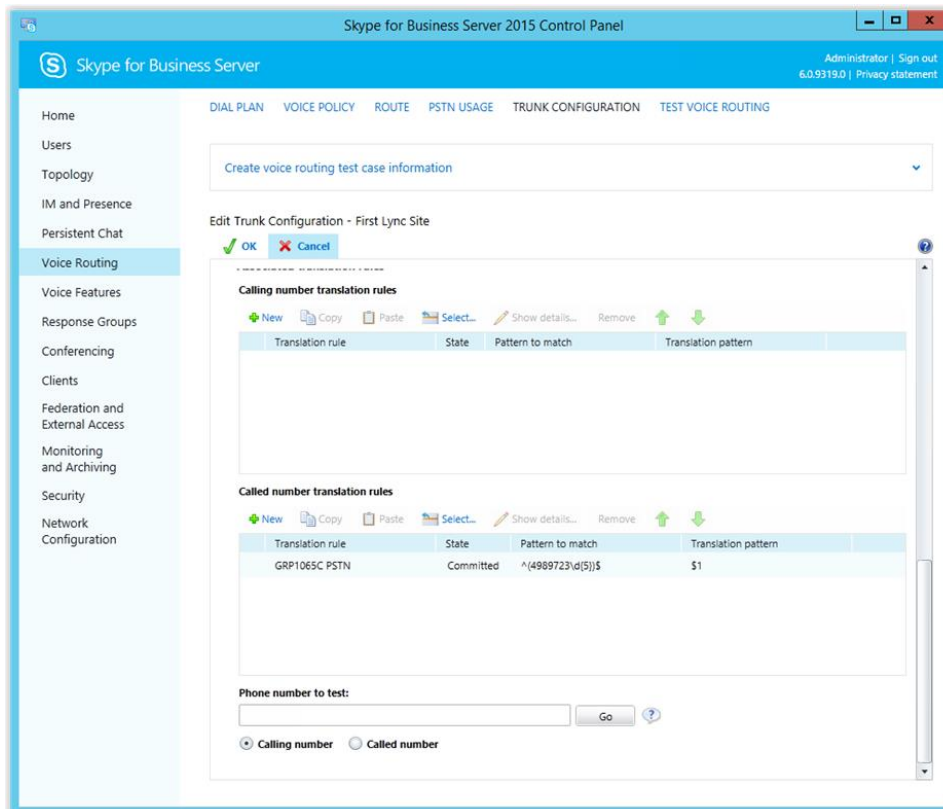
Dial plan for calls to OpenScape Business

In order Skype for Business clients to call a number in OpenScape Business (external line), they should dial the number “0” first.

3. In the Voice Routing page, click the [Trunk Configuration](#) tab.
4. Click on [New] to create a new trunk to OpenScape Business.



Trunk to OpenScape Business configuration page (1)

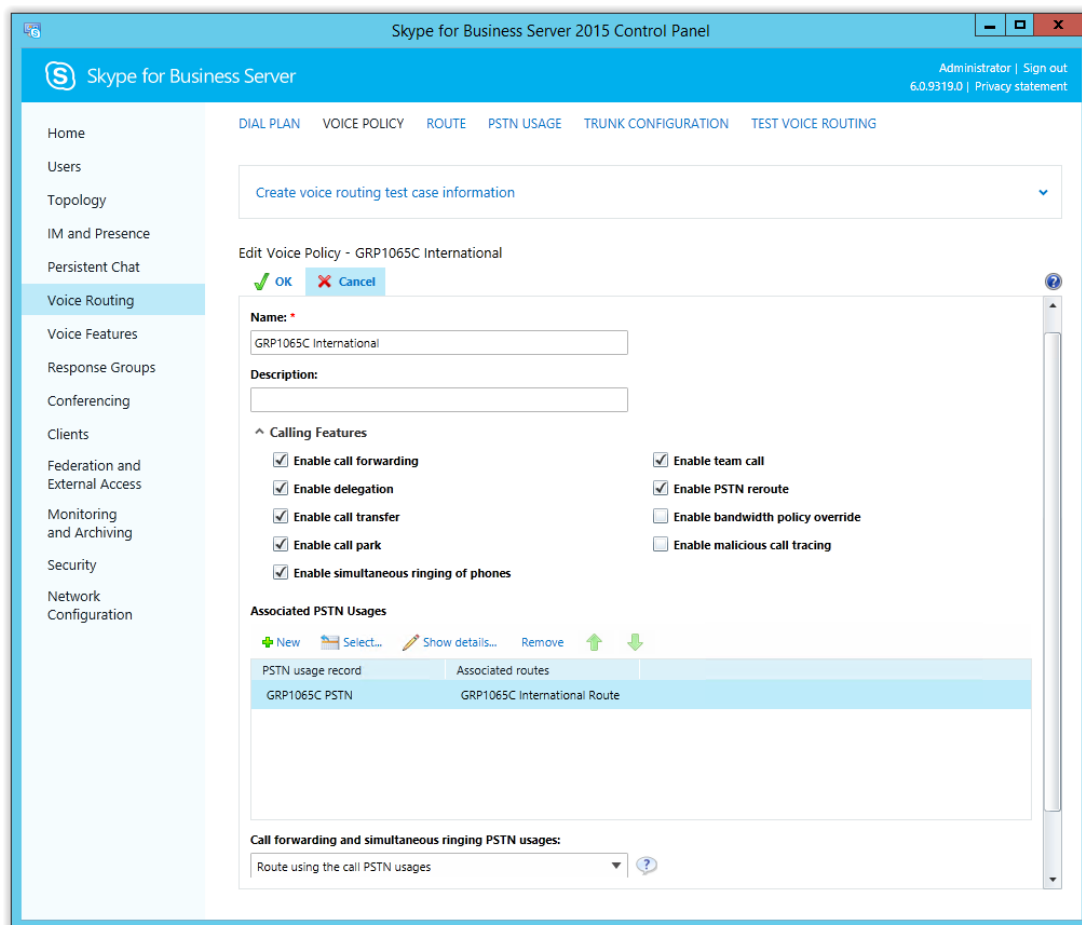


Trunk to OpenScape Business configuration page (2)

- a) General settings:
- **Encryption support level:** “Optional” (depending on the secure/non-secure interconnection to OpenScape Business requirements).
 - **Enable refer support:** unchecked / “disabled” (OpenScape Business doesn’t support REFER messages from Skype for Business)
 - **Enable media bypass:** unchecked / “disabled” (unless RTP messages from Skype for Business domain should not go through Skype for Business Mediation Server).
 - **Centralized media processing:** checked / “Activated”.
 - **Enable forward call history:** checked / “Activated”.
- b) Associate a **PSTN usage** record (explained later on). A PSTN Usage record is the link between voice policies and voice routes.
- c) Associate **Translation Rules** for the OpenScape Business; final phone number manipulation before exiting Skype for Business.

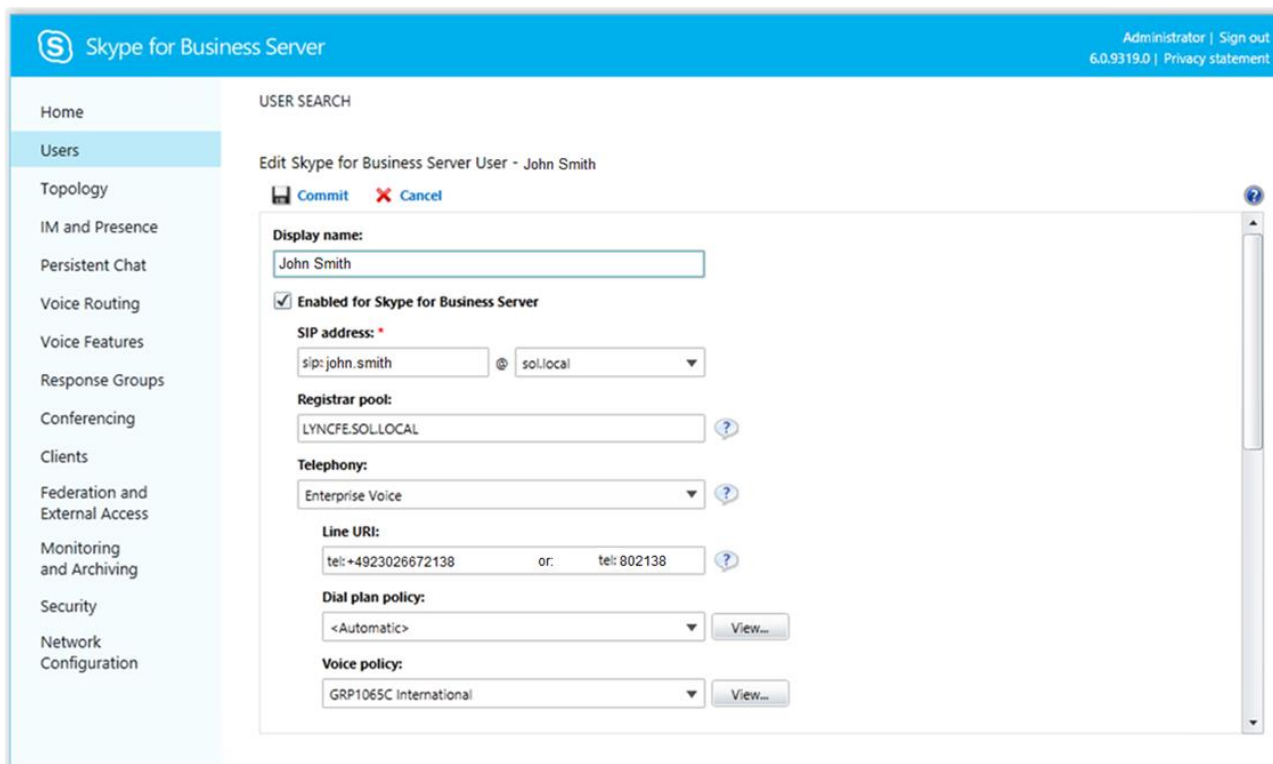
5. In the **Voice Routing** page, click the **Voice Policy** tab. A Voice Policy controls what kind of calls users are able to make.

6. Click on [New] to create a new voice policy for calls to OpenScape Business.



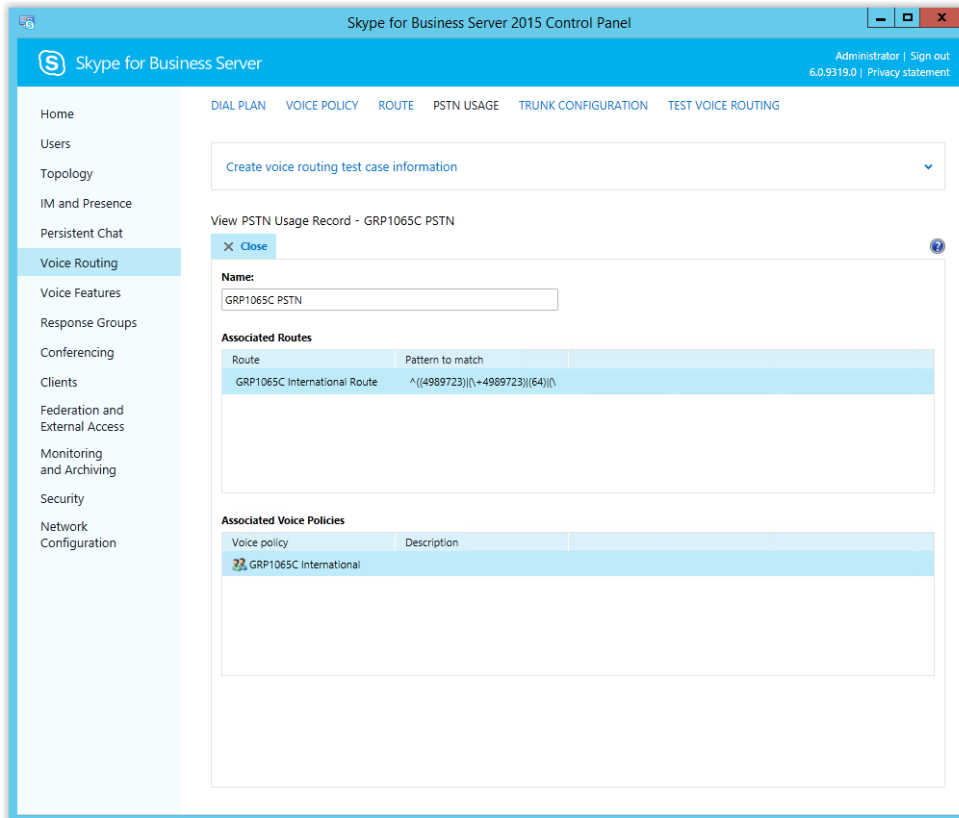
Voice Policy for the route to OpenScape Business

A Skype for Business user should have a voice policy that allows calls through the trunk connection to OpenScape Business. The Telephony option: Enterprise Voice reflects the Skype for Business user license (here: CAL Plus) which enables the user to perform PSTN calls (here: via the OpenScape Business). The according Line URI must fit into the OpenScape Business numbering plan and DID range provided by the Central Office Carrier.

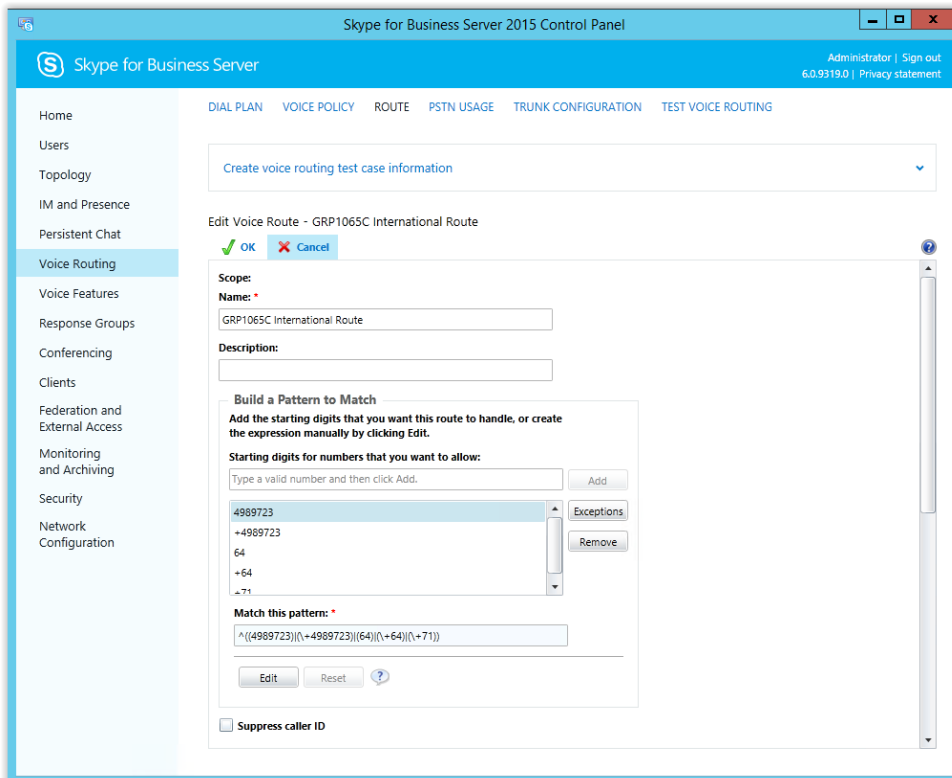


OpenScape Business trunk voice policy for Skype for Business user

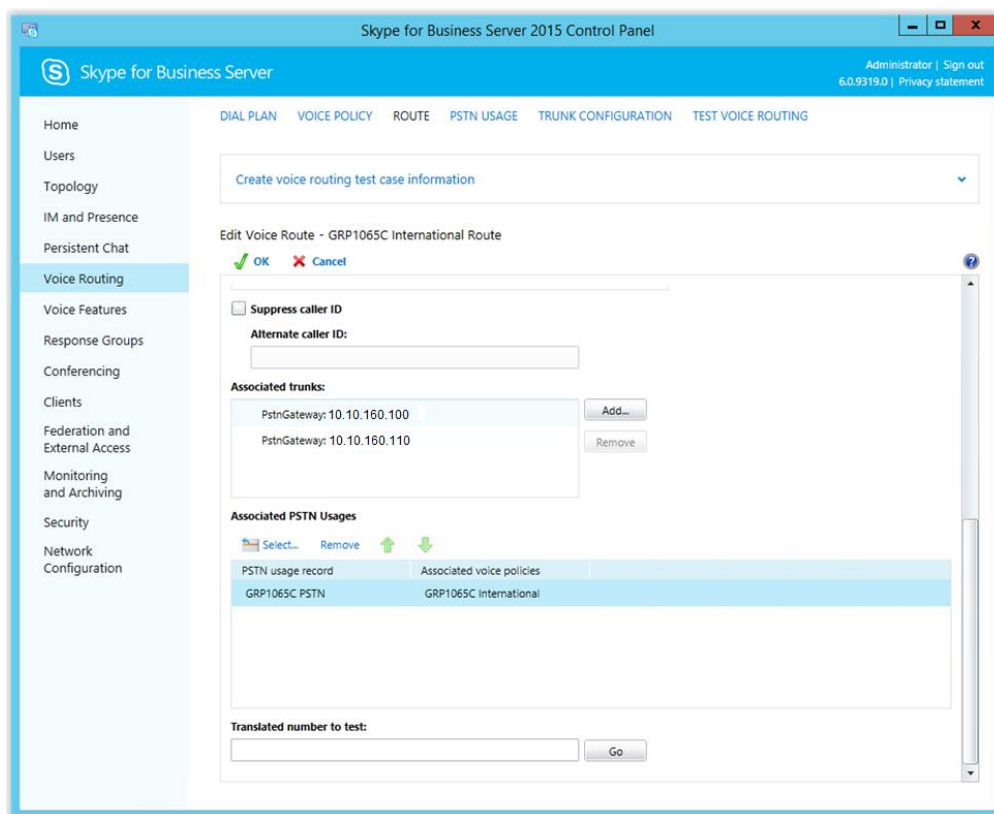
7. Associate a route at Associated [PSTN Usages](#) by clicking on [New]; a voice route specifies how Skype for Business Server handles outbound calls (to OpenScape Business).



PSTN Usage for outbound calls to OpenScope Business



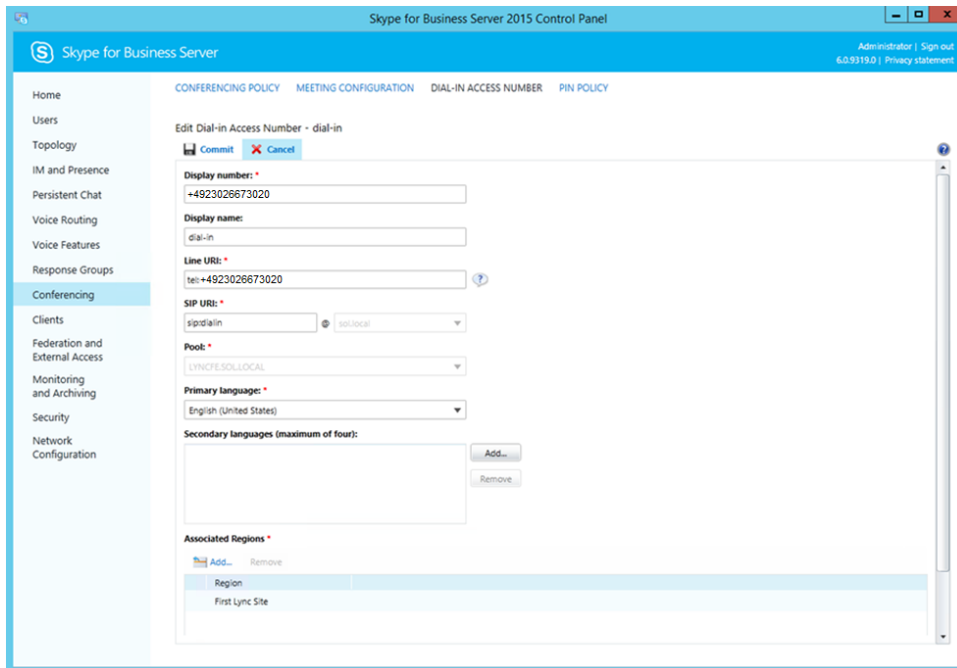
Voice Route for outbound calls to OpenScope Business (1)



Voice Route for outbound calls to OpenScape Business (2)

- a) In the **'Build a Pattern to Match'** field, enter the starting digits you want this route to handle (e.g., * matches all numbers).
- b) In the **Associated Trunks** panel, a list of all the deployed gateways is displayed (e.g., 10.10.160.100 and 10.10.160.110).

8. The setup of the conference call numbers can be found here and must be in line with the Skype for Business dial plan and OpenScape Business LCR dial plan.



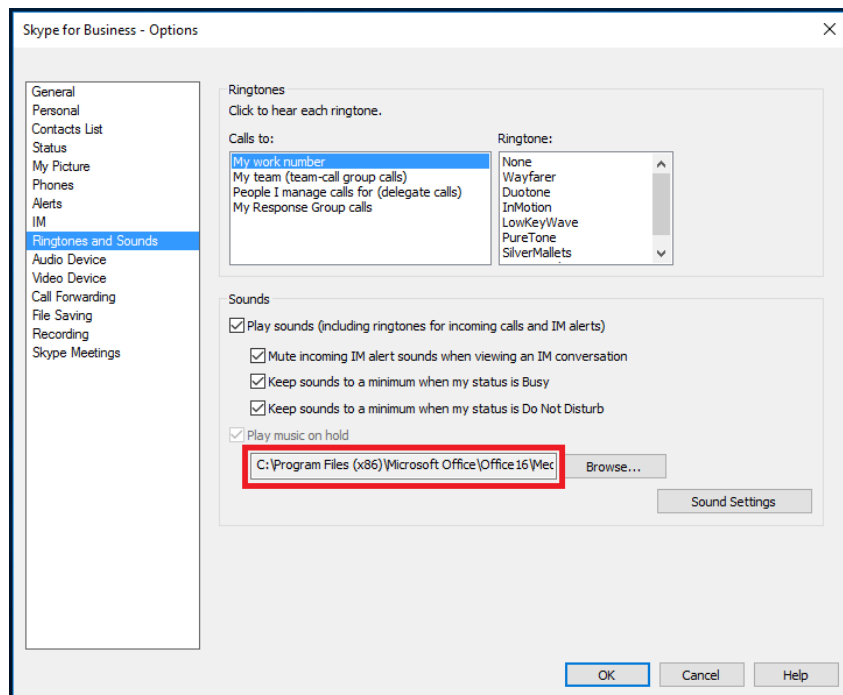
Conference Call number

3. Skype for Business Client

In order to have the SfB client’s full Calling Features (see: Voice Policy), install MS Office 2016 Pro on a PC (the client is included in the package).

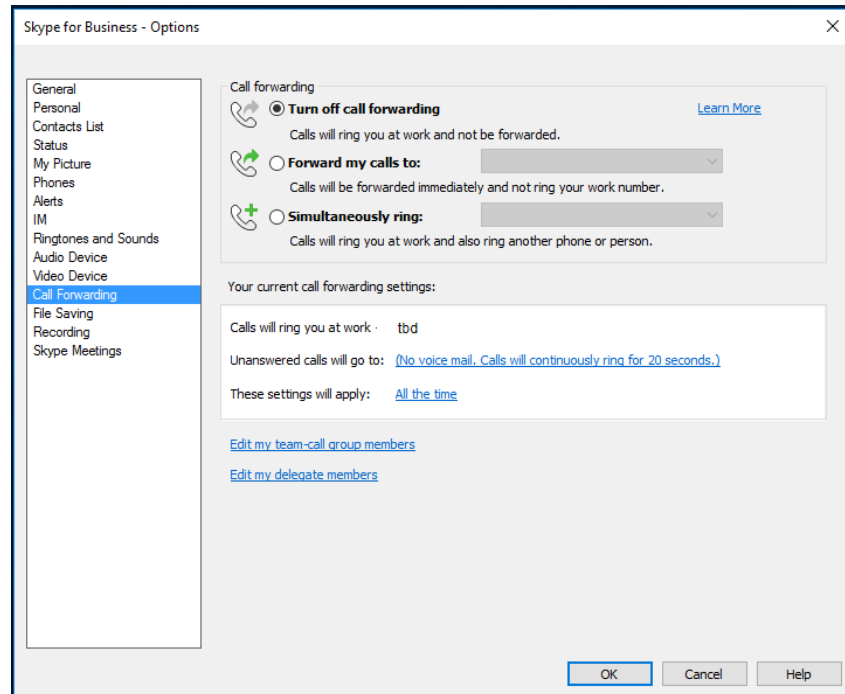
Please, refer to <https://technet.microsoft.com/en-us/library/dn933896.aspx>

The SfB Client provides the Music on Hold settings:



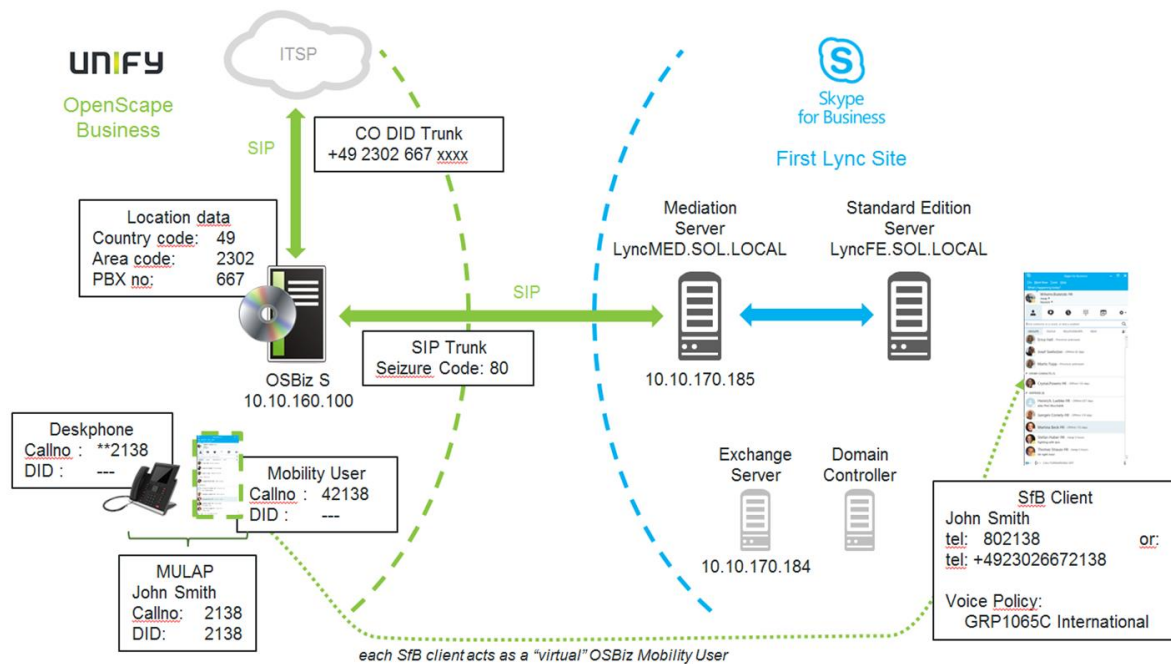
During the SfB Calling Features “Hold” and “Consultation” the payload connection is still handled via the SfB Mediation Server and the call receives Music on Hold (MoH) by the SfB Client.

The SfB Client provides the Call Forwarding settings. These Call Forwarding settings are independent to the call forwarding settings of the OpenScape Business. The different settings might overrule each other.



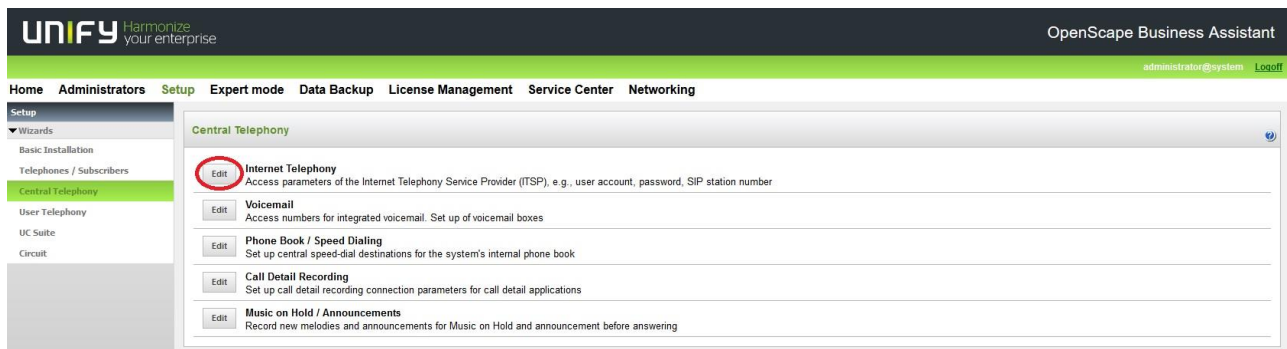
4. Configuration OpenScape Business

The following picture gives an overview of the example configuration. The SIP connection between OpenScape Business and the Skype for Business Mediation Server is realized with an ITSP Trunk.



4.1. Configuration Wizard – Internet Telephony

Go to Central Telephony – “Internet Telephony“



The overview page appears for entering the location data.

Setup - Wizards - Central Telephony - Internet Telephony

Overview

Note: changes done in expert mode must be reviewed/repeated after running through the wizard.
Note: At least the configuration of the 'Country code' is needed for features such as 'Internet telephony' and 'MeetMe conference'.

PABX number

Country code: 00 49 (mandatory)
Local area code: 0 2302 (optional)
PABX number: 667 (optional)

Click [OK & Next].

Provider configuration and activation for Internet Telephony:

No call via Internet -> uncheck

Setup - Wizards - Central Telephony - Internet Telephony

Provider configuration and activation for Internet Telephony

No call via Internet:

Country specific view: Germany

Note: changes done in expert mode must be reviewed/repeated after running through the wizard.

	Activate Provider	Internet Telephony Service Provider
Edit	<input type="checkbox"/>	Sipgate
Edit	<input type="checkbox"/>	Sipgate Trunking
Edit	<input type="checkbox"/>	Skype Connect
Edit	<input checked="" type="checkbox"/>	Skype for Business
Edit	<input type="checkbox"/>	Telekom DeutschlandLAN SIP-Trunk Registered Mode

Activate “Skype for Business” and click on [Edit].

Hint: depending on the “history” of the OpenScape Business the field [Edit] might be grayed out and an initializing of the LCR with default data might be requested before executing the wizard for “Internet Telephony”.

Please collect needed LCR data before initializing.

Expert mode - Telephony Server

LCR

LCR Flags

Edit LCR Flags

Reset LCR data

Activate LCR

Delete the configured LCR data and initialize the LCR with default data

Apply Undo Help

4.1.1. IP Address setup “Skype for Business” ITSP

Enter the IP Address (or Host Name) and Port of the Skype for Business Mediation Server:

Setup - Wizards - Central Telephony - Internet Telephony

Internet Telephony Service Provider

Provider Name: Skype for Business
Enable Provider:
Secure Trunk:
Domain Name: 10.10.170.185

Provider Registrar

Use Registrar:
IP Address / Host name:
Port: 5060
Reregistration Interval at Provider (sec): 600

Provider Proxy

IP Address / Host name: 10.10.170.185
Port: 5068

Provider Outbound Proxy

Use Outbound Proxy:
IP Address / Host name: 0.0.0.0
Port: 0

Click [OK & Next]

In the next dialog the Skype for Business User data will be configured.

Setup - Wizards - Central Telephony - Internet Telephony

Internet Telephony Stations for Skype for Business

Name of Internet Telephony Station
New Internet Telephony Station

Add

Click on [Add].

Setup - Wizards - Central Telephony - Internet Telephony

Internet Telephony Station for Skype for Business

Internet telephony station: SfB
Authorization name:
Password:
Confirm Password:

Call number assignment

Use public number (DID)

If using 'configurable clip' you have to change the configuration to 'Use public number (DID)' here!
Changing trunk parameters in case of internal subscriber no. is not allowed!

ITSP-multiple route:
Default Number: +4923026670

Default Number
ITSP as primary CO access
Enter one of the call numbers supplied by your network provider here. This will be used in outgoing calls as the calling party number in case no other number is available for the respective call.
All call numbers supplied by your network provider are to be entered within the trunk and telephones configuration (DID field) primary CO access.

Enter a name (e.g. “SfB”) and the default number (e.g. intercept position) for the User.

Click [OK & Next].

Setup - Wizards - Central Telephony - Internet Telephony

Call Number Assignment for Skype for Business

Name of Internet Telephony Station	Internet Telephony Phone Number	Direct inward dialing	Use as PABX number for outgoing calls
In order to complete the configuration please verify that the relevant user DIDs are set in stations.(Telephones / Subscribers configuration)			

Click [OK & Next] (no input needed)

4.1.2. Define bandwidth (# Trunks)

On this page the amount of simultaneous calls to Skype for Business is configured. The maximum number of Active Calls is defined by the available bandwidth.

Setup - Wizards - Central Telephony - Internet Telephony

Settings for Internet Telephony

Simultaneous Internet Calls

Available Lines for ITSP: 184

Please enter in field 'Upstream up to (Kbit/sec)' the Upstream of your Internet connection communicated by your Provider. You have typed in **Upstream up to (Kbps) = 2056**

In the 'Change Feature -> Internet Telephony' Assistant. This upstream allows you to conduct up to 16 Internet phone calls simultaneously. If the call quality deteriorates due to the network load, you will need to reduce this number of simultaneous calls. The number of simultaneous Internet Calls also depends on the licensing.

Upstream up to (Kbps):

Number of Simultaneous Internet Calls:

Line assignment

Internet Telephony Service Provider	Configured Lines	Assigned Lines
Skype for Business	8	<input type="text" value="8"/>
	0	<input type="text" value="8"/>

Click [OK & Next]

4.1.3. Special phone numbers

In this dialog it is possible to route special phone numbers (e.g. emergency calls). Those numbers should not be assigned to the Skype for Business Server.

Click [OK & Next]

4.1.4. ITSP status

On next page status of ITSP is displayed.

Setup - Wizards - Central Telephony - Internet Telephony

Status for the Internet Telephony Service Provider (ITSP)

Provider	Enabled	User
Skype for Business	Enabled	SB1 registered

Click [Next]

Hint: if status is orange please check for correct IP address / port (the Diagnose button will provide further info).

4.1.5. Exchange Line Seizure

Select which trunk will use access code 0. Enter the local area code without prefix digits (needed only when local area code was not entered in first step PBX number).

This should not be assigned to the Skype for Business Server.

Setup - Wizards - Central Telephony - Internet Telephony

Exchange Line Seizure

Exchange Line Seizure

Trunk Access Code 0

Dial over Provider Central Office ITSP

Click [OK & Next]

Hint: You select which trunk will use access code 0. The list may have more than one option because you may have configured more than one ITSP trunks or because there is also ISDN access at your system.

Area Code field is visible under specific circumstances and depends on country's specific data (e.g: when national prefix exists). If available, you have to set the Local Area Code your system belongs to (but without national prefix in front of it). Please mind the case of multiple provider configurations (Multi Site)

Overview with all configured „Outside line Seizure“ are displayed.

Setup - Wizards - Central Telephony - Internet Telephony

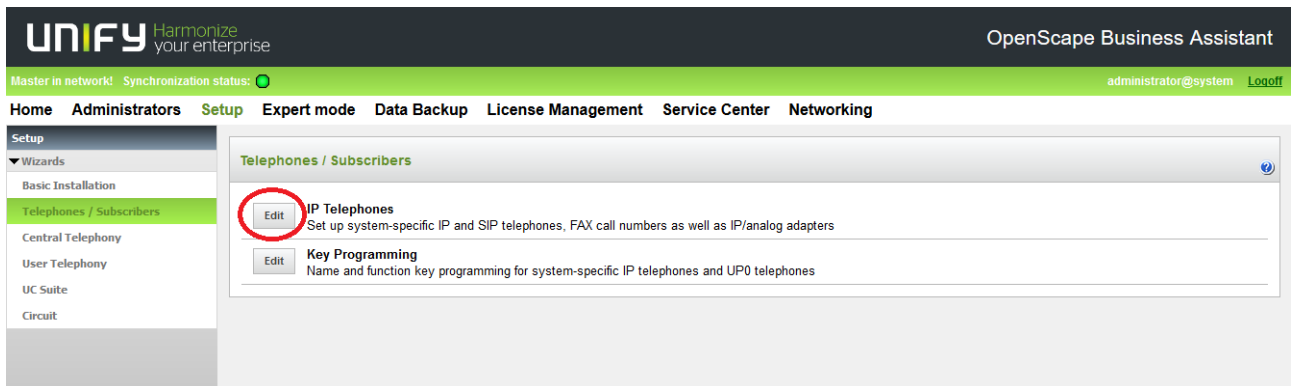
Seizure Code for the 'Outside line Seizure'

	Seizure code for 'Outside line Seizure'	
Central Office ITSP	0	
Skype for Business	80	

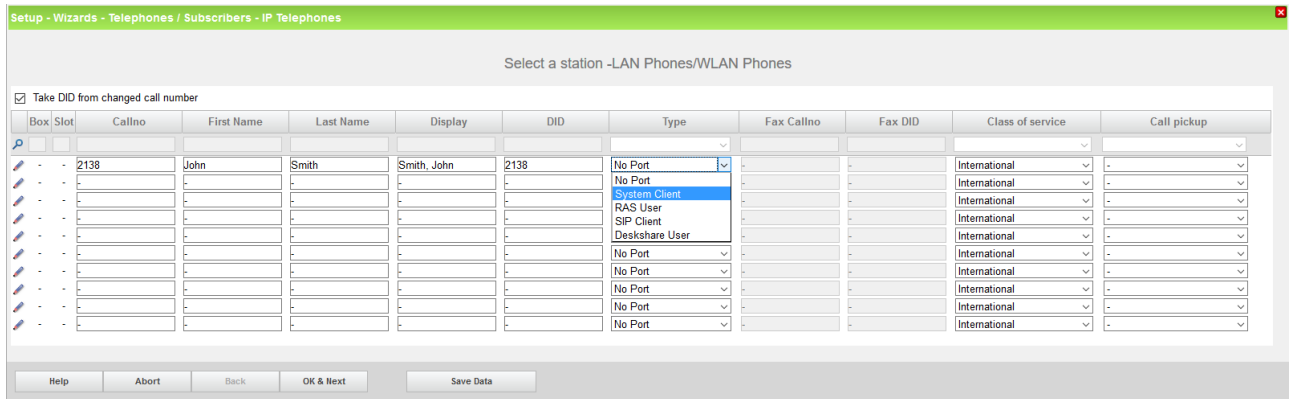
Click [OK & Next] and on the next page [Finish]

4.2. Configuration Wizard – IP Telephones

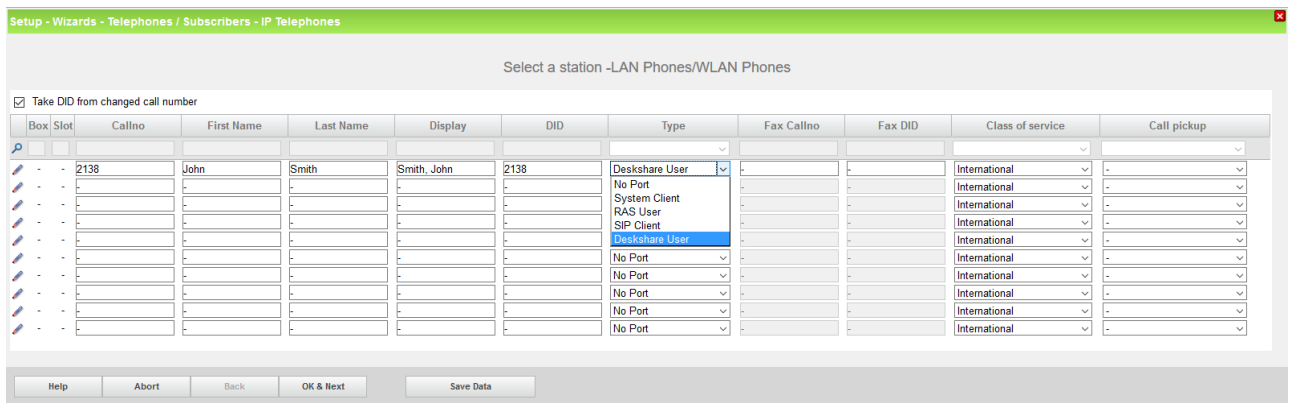
Create a System Client or Deskshare User by entering the “IP Telephones” wizard



Configure Names and Numbers and select Type: System Client



or Deskshare User



4.3. Configuration Wizard – Mobile Phone Integration

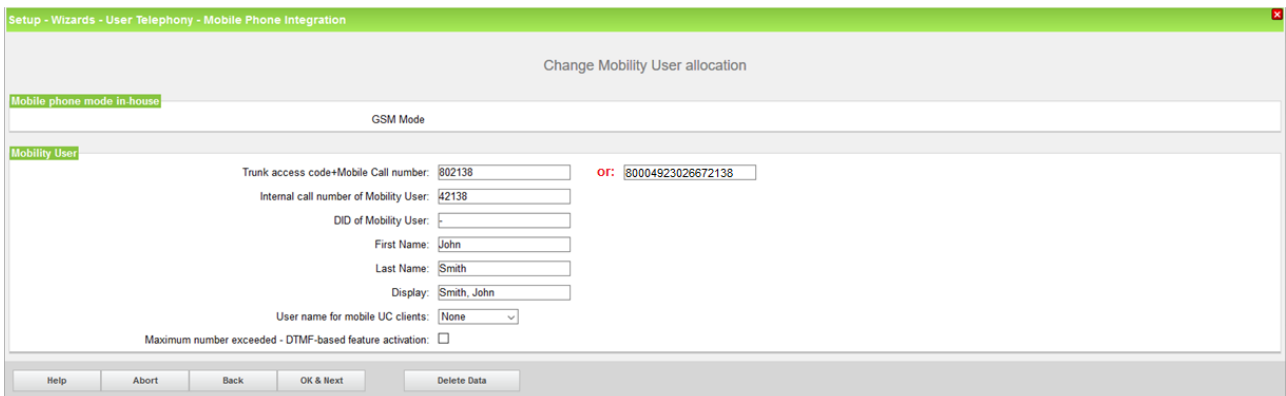
Create a Mobility User by entering the „Mobility Phone Integration“ wizard

The screenshot shows the OpenScope Business Assistant interface. At the top, there is a navigation bar with the UNIFY logo and the text "Harmonize your enterprise". On the right, it says "OpenScope Business Assistant" and "administrator@system" with a "Logout" link. Below the navigation bar, there are tabs for "Home", "Administrators", "Setup", "Expert mode", "Data Backup", "License Management", "Service Center", and "Networking". The "Setup" tab is active, and a sidebar on the left shows a tree view of configuration options: "Wizards", "Basic Installation", "Network / Internet", "Telephones / Subscribers", "Central Telephony", "User Telephony" (highlighted), "Security", "UC Suite", and "Circuit". The main content area is titled "User Telephony" and contains several configuration sections, each with an "Edit" button. The "Mobile Phone Integration" section is circled in red. Its description reads: "Set up a link between a mobile phone and an internal station with the goal of enabling incoming and outgoing availability under one station number (One Number Service)".

Press “Add” to create a new Mobility User

The screenshot shows a dialog box titled "Setup - Wizards - User Telephony - Mobile Phone Integration". The main text inside the dialog says "Select station for Mobility". Below this, there is a "DISA port" label and a "Direct inward dialing:" label followed by an empty text input field. At the bottom left, there is a button labeled "Add" and the text "New Mobility User".

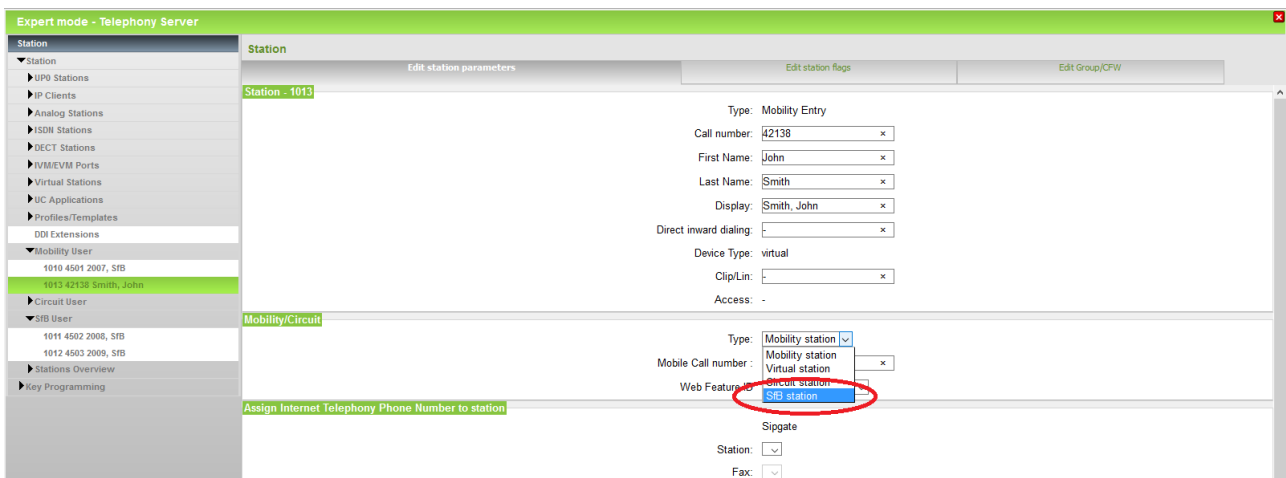
Skype for Business Server numbering plan is in standard E.164 format – other formats are optional and might not support the entire Skype for Business Server feature set. OpenScape Business supports both types:



Click [OK & Next] and on the next page [Finish]

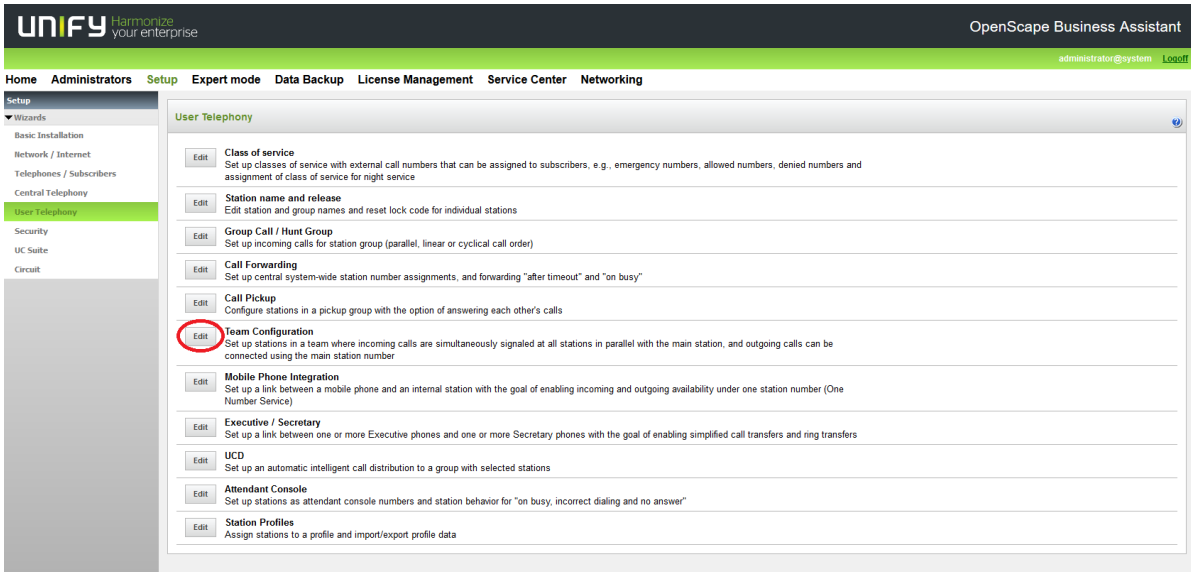
Hint: Depending on the use case standalone Mobility User or Mobility MULAP the Mobility User will have a DID. The station flag: “DTMF-based feature activation” – available in OSBiz X - is ignored for Mobility User type “SfB Station”. A Mobility User of type “SfB Station” sends DTMF transparently through the system.

Change in Expert Mode the Virtual Station Type to: “SfB station”:

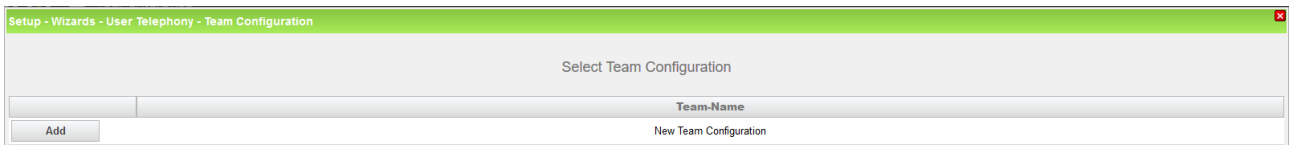


4.4. Configuration Wizard – Team Configuration

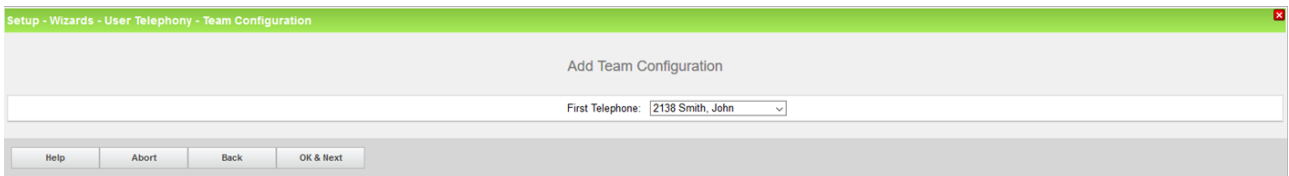
Create a MULAP by entering the „Team Configuration“ wizard



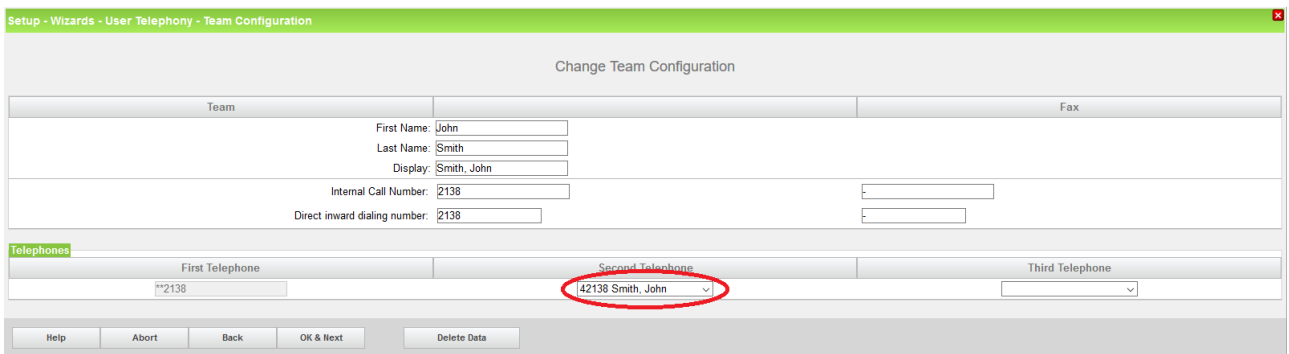
Press “Add” to create a new Team MULAP



Select the associated desk phone



Assign Name to MULAP and select the associated User callno



Click [OK & Next] and on the next page [Finish]

4.5. Additional Configuration

4.5.1. Route configuration

The route configuration must be changed manually. In this example the free of choice route name is “SfB ITSP”.

Other route parameters must be changed like this:

Expert mode - Telephony Server

Trunks/Routing

Route

Change Route | Change Routing Parameters | Special Parameter change

Route Name: SfB ITSP

Seizure code: 90

CO code (2nd trunk code):

Gateway Location

Country code: 49

Local area code: 2302

PABX number: 667

PABX number-incoming

Country code:

Local area code:

PABX number:

Location number:

PABX number-outgoing

Country code:

Local area code:

PABX number:

Suppress station number:

Overflow route

Overflow route: None

Digit transmission

Digit transmission: en-bloc sending

Mobile Extension Number (MEX)

MEX Number:

Apply | Undo | Help

Expert mode - Telephony Server

Trunks/Routing

Route

Change Route | Change Routing Parameters | Special Parameter change

Routing flags

Digit repetition:

Analysis of second dial tone / Trunk monitoring:

Intercept per direction:

Over service 3-1-ME:

Add direction prefix incoming:

Add direction prefix outgoing:

Call No. with international / national prefix:

Ringback tone to CO:

Name in CO:

Segmentation: yes

deactivate UUS per route:

Always use DSP:

Analog trunk seizure: no pause

Trunk call pause: Pause 6 s

Type of seizure: linear

Route type: PABX

No. and type, outgoing: Country code

Call number type: Internal / DID

Rerouting

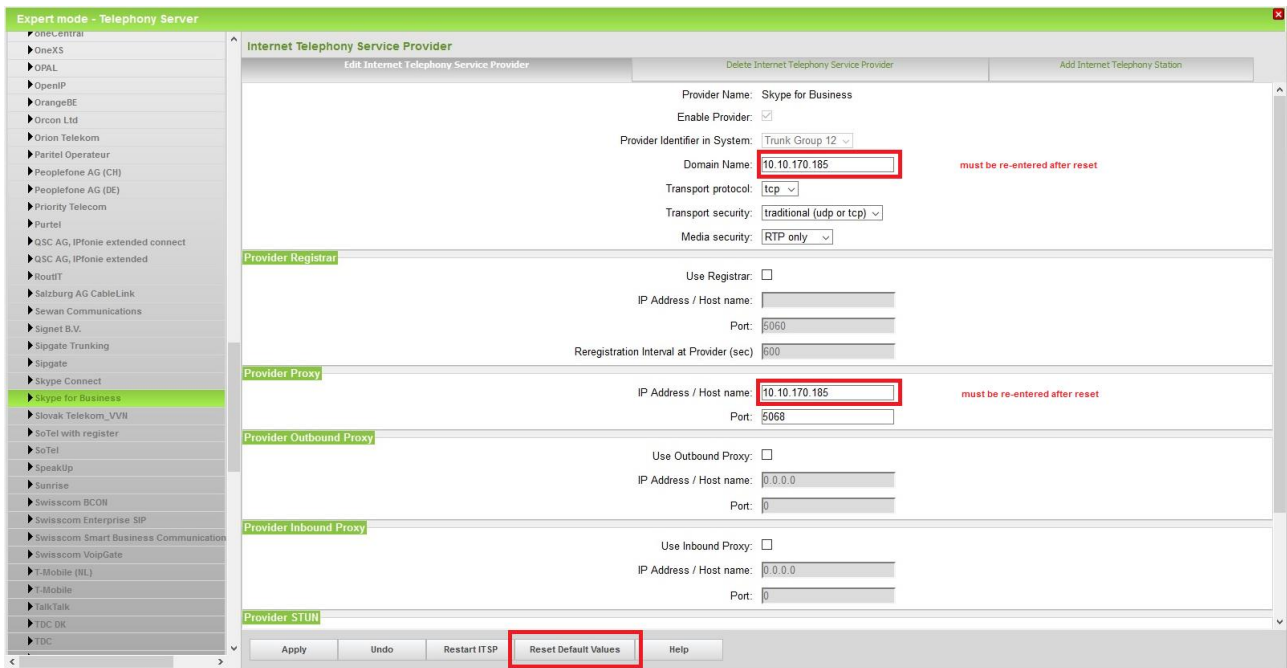
Change route allowed:

Route optimize active: No

Apply | Undo | Help

Within V3 overriding of these values is prohibited by the “Skype for Business” ITSP template. The capabilities of the “Skype for Business” ITSP template can be obtained

- if first configuration of “Skype for Business” ITSP is performed
- if “Skype for Business” ITSP is reseted to default values once:



After Click [Reset Default Values] please re-enter Domain Name and IP Address / Host Name (see chapter 4.1.1).

4.5.2. LCR changes

Additional manual configuration in the LCR settings is required in order to call the SfB-Clients and the SfB-Conference.

1. Changes in Dialplan

Create a Dial Plan entry for the SfB-Clients and the SfB-Conference call numbers. The SfB-Clients and the SfB-Conference Dialed digits depend on the Skype for Business Server numbering plan. They have to cover the Trunk Access Code, the Mobile Call number of the Mobility User and the SfB-Conference dial-in number. The SfB-Conference dial-in number needs to be defined (tbd) on the customer's need and might not fit into the SfB-User / Mobile Call numbering schema. Link these entries to independent Routing Tables.

If the Skype for Business Server numbering plan is in E.164 format:

Dial Plan	Name	Dialed digits	Routing Table	Acc. code	Classes of service	Emergency
101	SfB-Clients	80-00-492302667XXX	60	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
102	SfB-Conference	to be defined on customer's need (tbd)	61	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
103			-	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
104			-	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>

or (in all other cases):

Dial Plan	Name	Dialed digits	Routing Table	Acc. code	Classes of service	Emergency
101	SfB-Clients	80-XXXX	60	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
102	SfB-Conference	tbd	61	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
103			-	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
104			-	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
105			-	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
106			-	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
107			-	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>

2. Configure Routing table

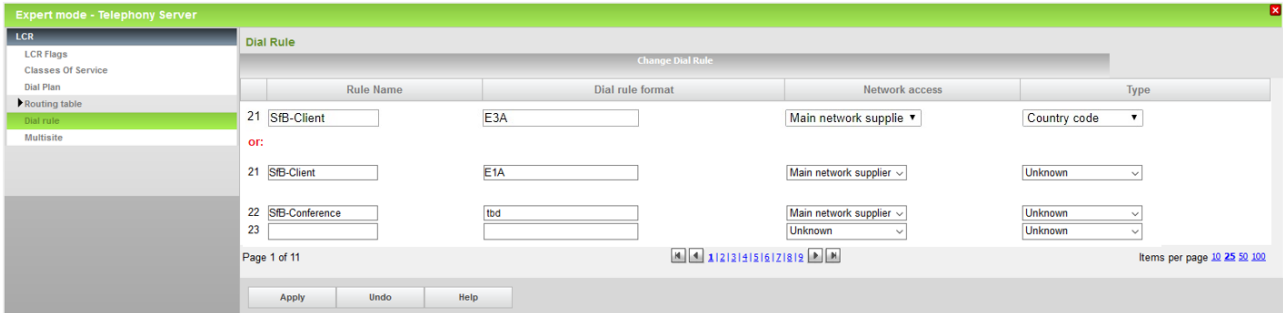
The SfB-Clients and the SfB-Conference Routing tables should use the Route which is linked to the SfB ITSP and use independent Dial Rules.

Index	Dedicated Route	Route	Dial Rule	min. COS	Warning	Dedicated Gateway	GW Node ID
1	<input type="checkbox"/>	SfB ITSP	SfB-Client →	1	None	No	
2	<input type="checkbox"/>	None	None	15	None	No	
3	<input type="checkbox"/>	None	None	15	None	No	
4	<input type="checkbox"/>	None	None	15	None	No	

Index	Dedicated Route	Route	Dial Rule	min. COS	Warning	Dedicated Gateway	GW Node ID
1	<input type="checkbox"/>	SfB ITSP	SfB-Conference →	1	None	No	
2	<input type="checkbox"/>	None	None	15	None	No	
3	<input type="checkbox"/>	None	None	15	None	No	
4	<input type="checkbox"/>	None	None	15	None	No	

3. Create Dialing Rules

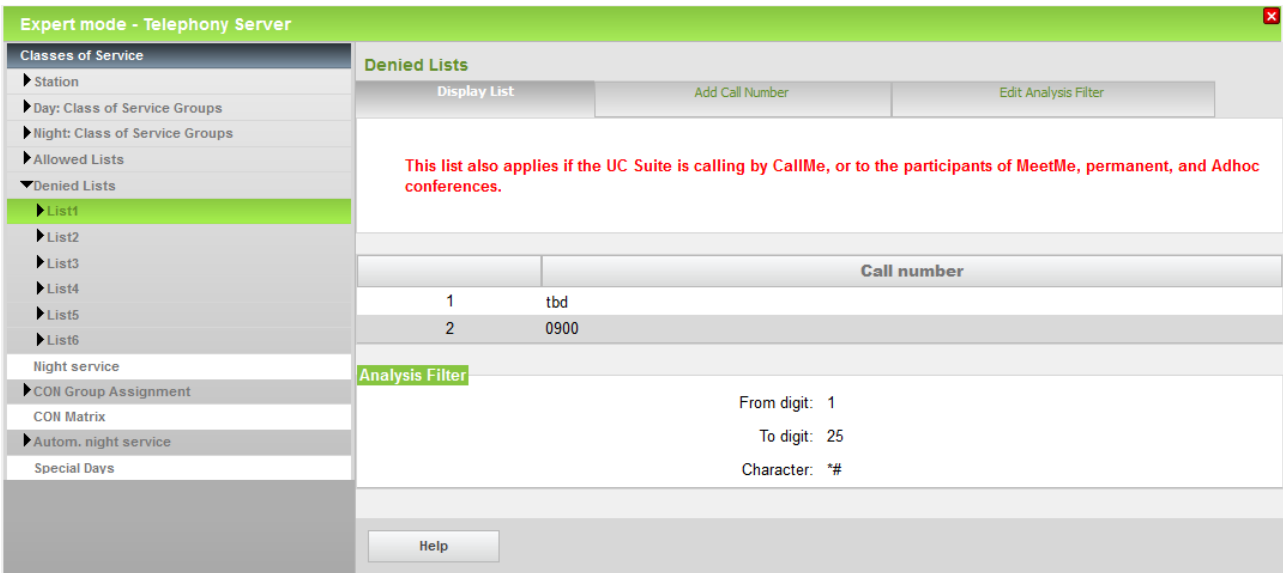
The SfB-Clients and the SfB-Conference Dial Rule format setting depends on the Skype for Business Server numbering plan. In all cases the Network access should be “Main network supplier”. If the Skype for Business Server numbering plan is in E.164 format the type is of “Country code” – see 1st example. In all other cases the type is set to “Unknown” – following example:



Hint: the intention of dial rule format E1A is to repeat the trunk access code “80” as part of the mobile call number.

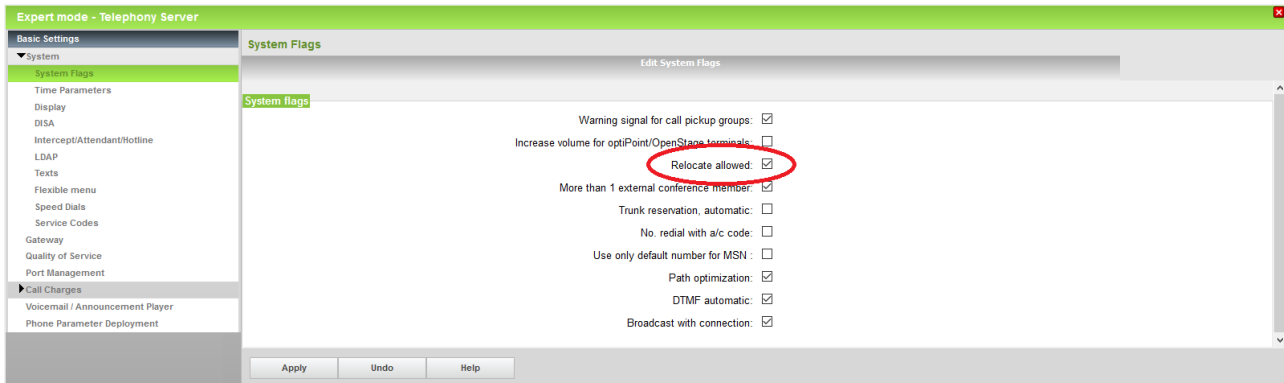
4.5.3. Class of Service

During SfB Client features Call Forwarding and Blind Transfer (w/o Consultation) the resulting second call is not assigned to the Mobility MULAP resp. the Mobility User but is restricted by the COS of the Mobility User.

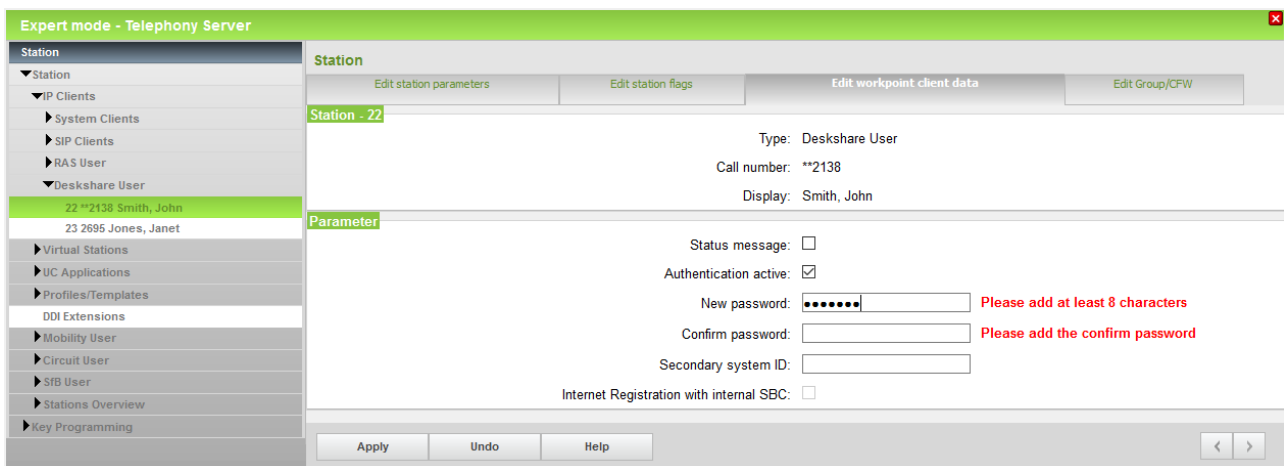


4.5.4. Deskshare User

Activate the deskshare feature



Assign a password to the Deskshare User



Hints:

- in a network scenario all involved nodes need to activate this feature
- when activating desksharing, the selected station number must be the MULAP member (here: **2138) and the according PIN
- in a network the user has to enter in addition the GK IP-Address of the OSBiz node (here: 10.10.160.100) where the origin user (here: **2138) is located

4.5.5. E-mail Forwarding

OpenScape Business Email forwarding to Skype for Business for missed calls, voicemail calls requires Voice Mail and UC licenses in addition.

The screenshot displays the OpenScape Business Assistant web interface. The top navigation bar includes the UNIFY logo with the tagline "Harmonize your enterprise" and the text "OpenScape Business Assistant" on the right. Below the navigation bar, a menu contains "Home", "Administrators", "Setup", "Expert mode", "Data Backup", "License Management", "Service Center", and "Networking". The "Service Center" section is expanded, showing a sidebar with "Documents", "Software", "Inventory", "SW Update", "E-mail forwarding" (highlighted), "Remote Access", "Restart / Reload", and "Diagnostics".

The main content area is titled "E-mail Forwarding" and contains the following configuration sections:

- Server Information:** Includes fields for "Outgoing Mail Server (SMTP)" (10.10.170.184), "Outgoing mail server port" (2525), and a checked checkbox for "This server requires an encrypted connection (TLS/SSL)".
- Login Information:** Includes fields for "User Name" (OSBizS@sol.local), "Password" (masked with dots), and "Confirm Password" (masked with dots).
- User Information (Sender):** Includes a field for "E-Mail Address" (OSBizS@sol.local).
- API change notification recipients:** Includes fields for "E-Mail Address 1" and "E-Mail Address 2".

At the bottom of the form, there are three buttons: "Abort", "OK & Next", and "Check e-mail forwarding".

Hint: hereby the usage of an Exchange server for the mail services and Unified Messaging is necessary for Interworking between OpenScape Business and Microsoft Skype for Business.

4.5.6. License

Add the “S2M/SIP Trunk” license to the SIP-Trunk:

The screenshot shows the 'CO Trunks' configuration page in the UNIFY OpenScope Business Assistant. The page title is 'CO Trunks'. Below the title, it states: 'Access to the Central Office via Internet telephony is licensed by CO trunk licenses'. To the right, it says 'Available licenses for SIP trunks: 0'. Below this, there is a section for 'SIP trunks' with the following information: 'The configured number of simultaneous Internet calls for each Internet Telephony Service Provider is: 16', 'License number of simultaneous Internet calls in this mode: 2', and 'License demand for number of simultaneous Internet calls in this mode: 2'. There is a dropdown menu next to the demand value, currently set to '2'.

Add the needed licenses to the IP User (desk phone) or Deskshare User:

The screenshot shows the 'IP User' configuration page in the UNIFY OpenScope Business Assistant. The page title is 'IP User'. Below the title, there is a search bar with '2138' and a 'Display' button. Below the search bar is a table with columns: 'Access', 'Call number', 'Display', and 'Remaining licenses'. The table has one row: 'LAN 0-SYS-10', '**2138', 'Smith, John', and '8 5 5 5 5* 5 5 0 0 1 2'. Below the table, there is a legend for the license status: 'Successfully licensed' (green check), 'Not licensed' (red X), 'Unsaved license release' (green check), 'Unsaved license demand release' (red X), 'License demand configurable' (checkbox), and 'License demand not configurable' (checkbox). There are also buttons for 'Abort' and 'OK & Next'.

Add the needed licenses to the Mobility User:

The screenshot shows the 'Mobility User' configuration page in the UNIFY OpenScope Business Assistant. The page title is 'Mobility User'. Below the title, there is a search bar with '2138' and a 'Display' button. Below the search bar is a table with columns: 'Access', 'Call number', 'Display', and 'Remaining licenses'. The table has one row: '42138', 'Smith, John', and '8* 5 5 5 5* 5 5 0 0 1 2'. Below the table, there is a legend for the license status: 'Successfully licensed' (green check), 'Not licensed' (red X), 'Unsaved license release' (green check), 'Unsaved license demand release' (red X), 'License demand configurable' (checkbox), and 'License demand not configurable' (checkbox). There are also buttons for 'Abort' and 'OK & Next'.

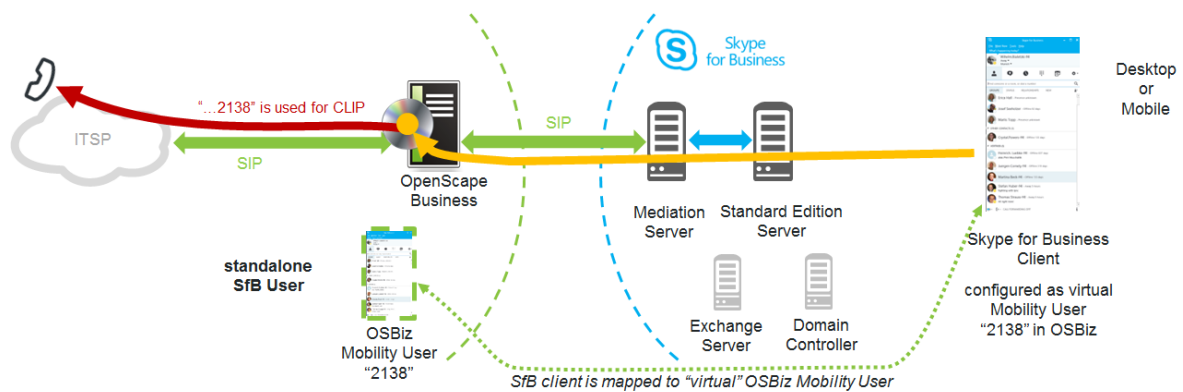
5. Example Call Flows

The following call flows example are based on the numbering plan defined in previous chapter.

5.1. via Skype GUI: Outbound Call from Standalone SfB Client

In this example an outbound call is launched from the SfB Client via the OpenScape Business. Within the OpenScape Business the SfB Client is assigned to a Mobility User (standalone configuration - differs from the configuration example of chapter 4.2). The inbound call from the SfB route claims the assigned Mobility User and signals the outbound call with the according CLIP.

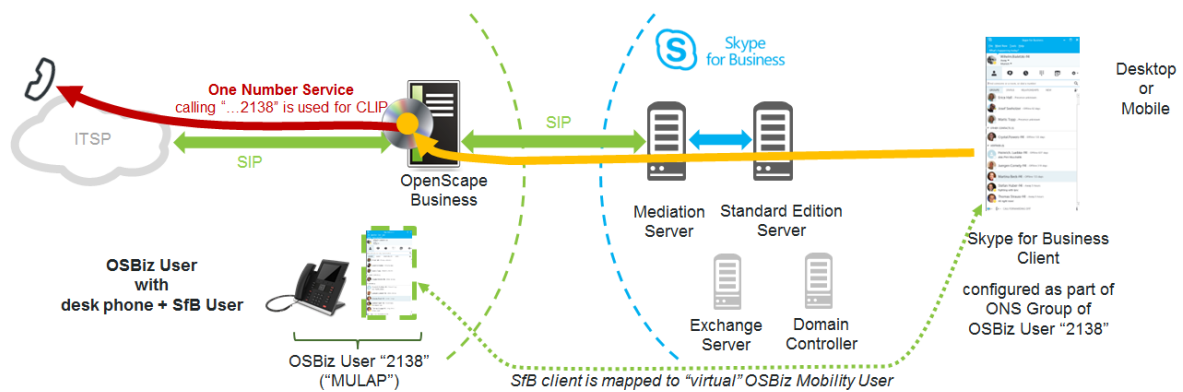
A trunk license is required per simultaneously used SIP channel which are shared for ITSP, SfB and others. The SfB client needs a virtual user representation in OpenScape Business. Therefore a Mobility / IP User license is required.



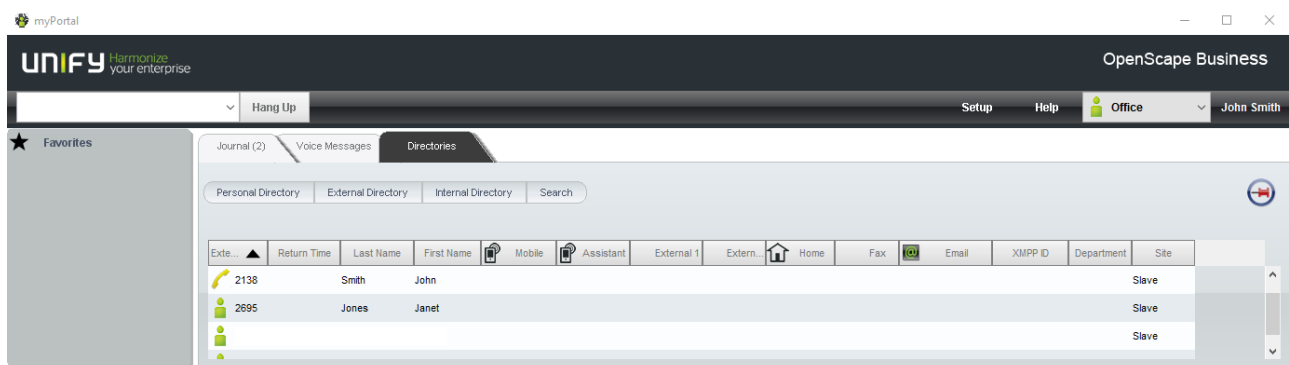
5.2. via Skype GUI: Outbound Call with One Number Service

In this example an outbound call is launched from the SfB Client via the OpenScape Business using One Number Service. That means within the OpenScape Business the SfB Client is assigned to a Mobility User and this Mobility User is part of a MULAP where the User has a desk phone in addition. The inbound call from the SfB route claims the assigned Mobility MULAP and signals the outbound call with the according CLIP.

A trunk license is required per simultaneously used SIP channel which are shared for ITSP, SfB and others. The SfB client needs a virtual user representation in OpenScape Business. Therefore a Mobility / IP User license and in addition a IP User license for the deskphone are required.

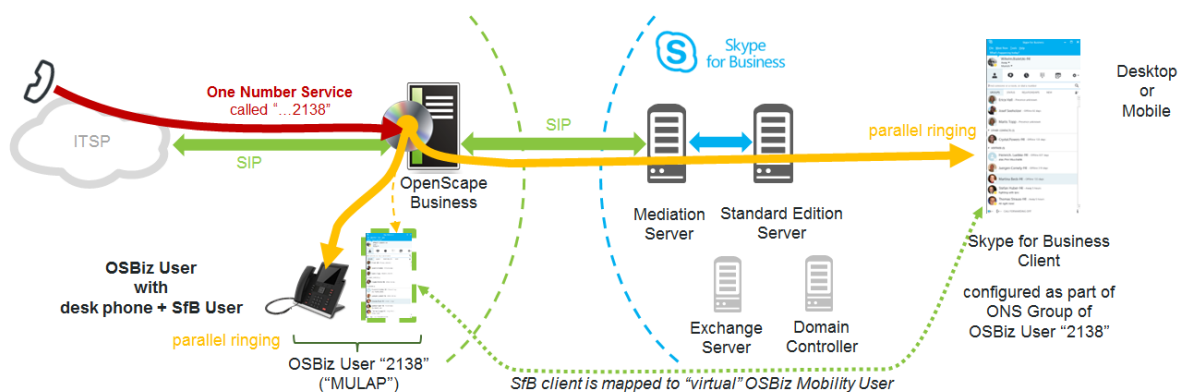


UC Suite example - according Hook State signaling: User is in a call

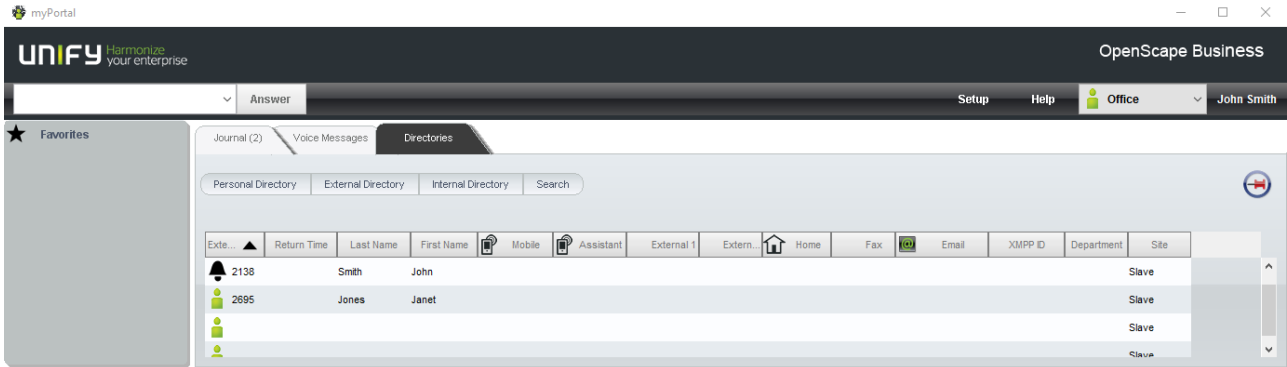


5.3. Inbound Call with parallel ringing

In this example with the same configuration as in chapter 5.2 there is an inbound call to the Mobility MULAP with parallel ringing on the desk phone and the SfB Client. Hereby is the Mobility User claimed by the inbound call.



UC Suite example - according Hook State signaling: User is alerting

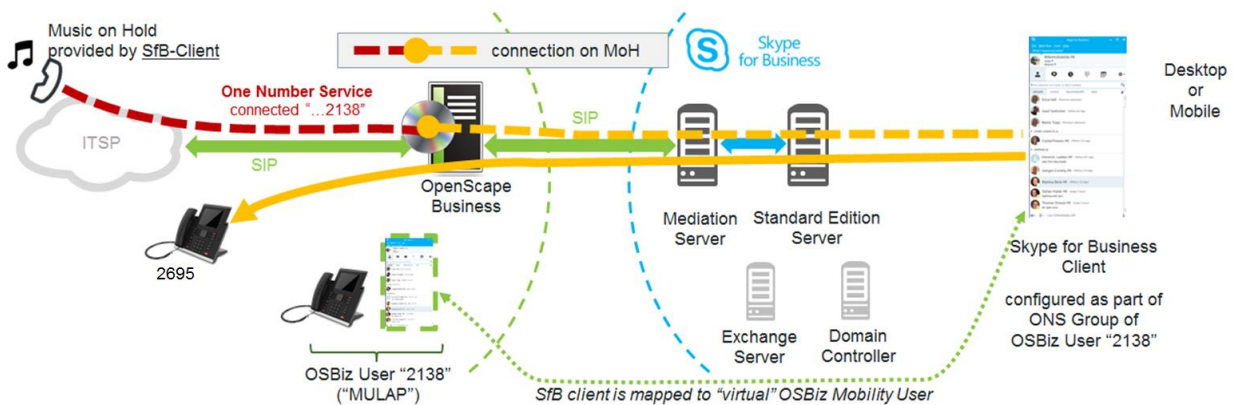


5.4. via Skype GUI: Consultation Call

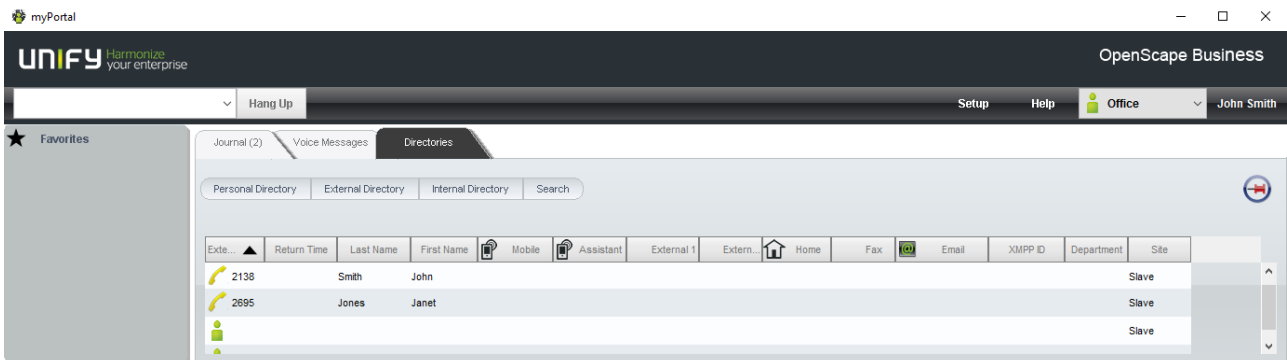
In this example with the same configuration as in chapter 5.2 the SfB User is in an existing first call as described in one of the previous chapters. Now the User puts the first call on hold and starts a consultation call. The payload connection is still handled via SfB Mediation Server and the first call receives Music on Hold (MoH) by the SfB Client. The second call requires another SIP channel in addition.

Although the second call inherits Calling Number and Calling Name and according Class of Service of the Mobility User, the second call is not assigned to the Mobility User.

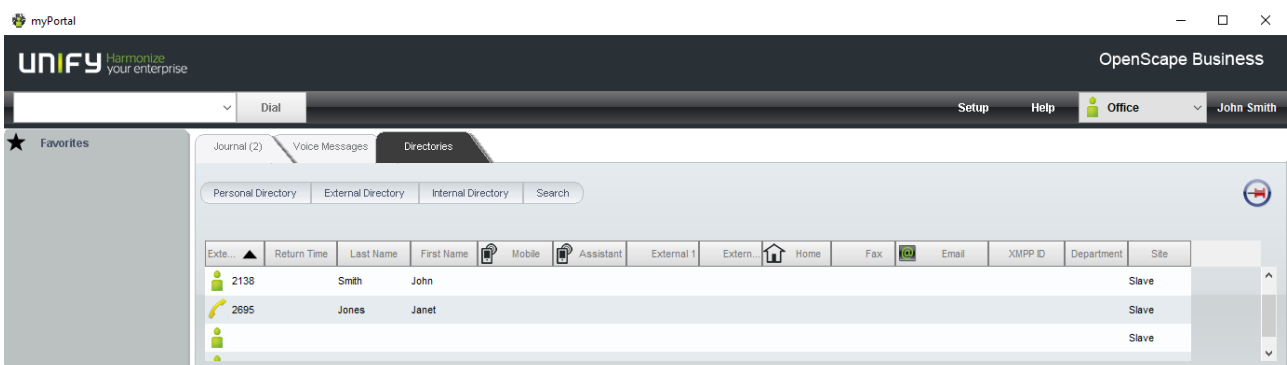
This means that only the first connection between OSBiz and an SfB client is monitored by OSBiz in terms of MULAP busy handling. If the external party of this example disconnects, the MULAP of User will be shown as idle to other OSBiz Users even if the consultation connection (second call) remains active.



UC Suite example - consultation call (1st call still on hold)

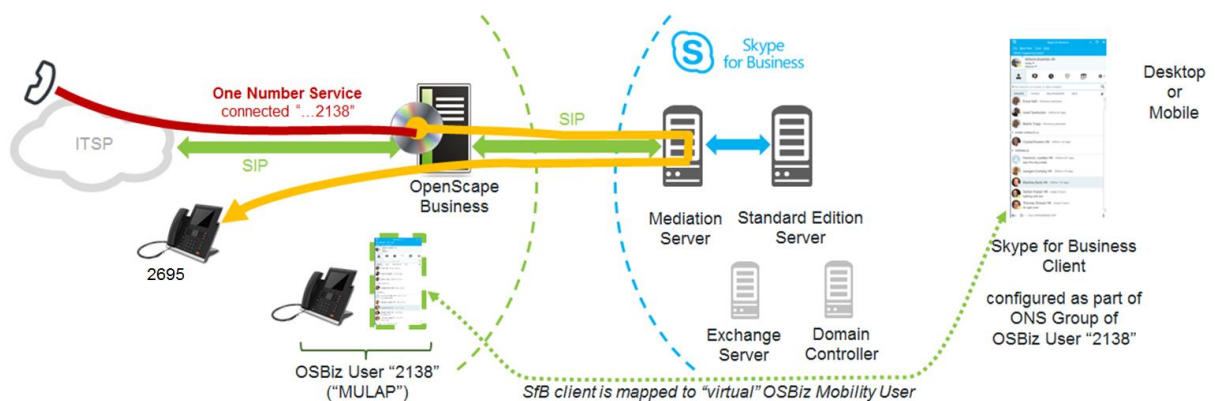


UC Suite example - consultation call (1st call is released – User appears idle)

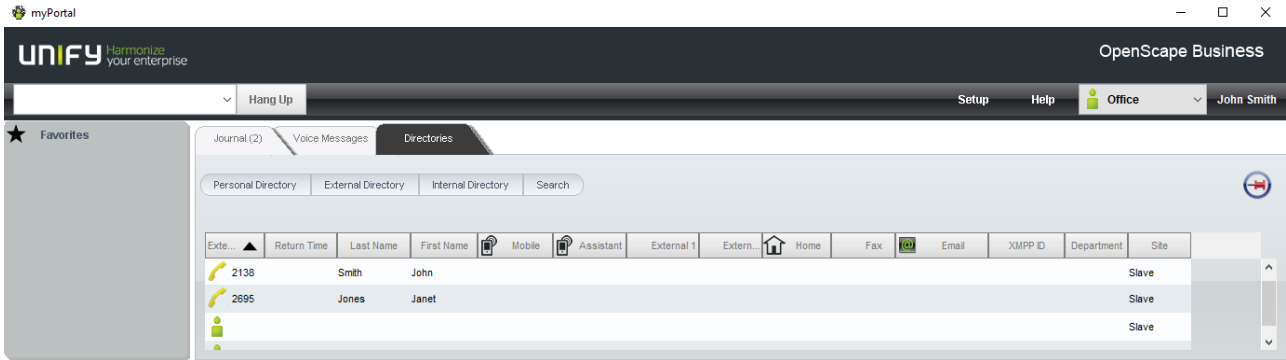


5.5. via Skype GUI: Supervised Transfer (after Consultation)

In this example based on chapter 5.4 the SfB User transfers a first call places on hold to a connected second call. The payload connection is still handled via SfB Mediation Server and because Path Replacement is not supported both calls stay active in a trombone connections until the transferred call is released. Meanwhile the Mobility MULAP resp. assigned Mobility User stays busy.



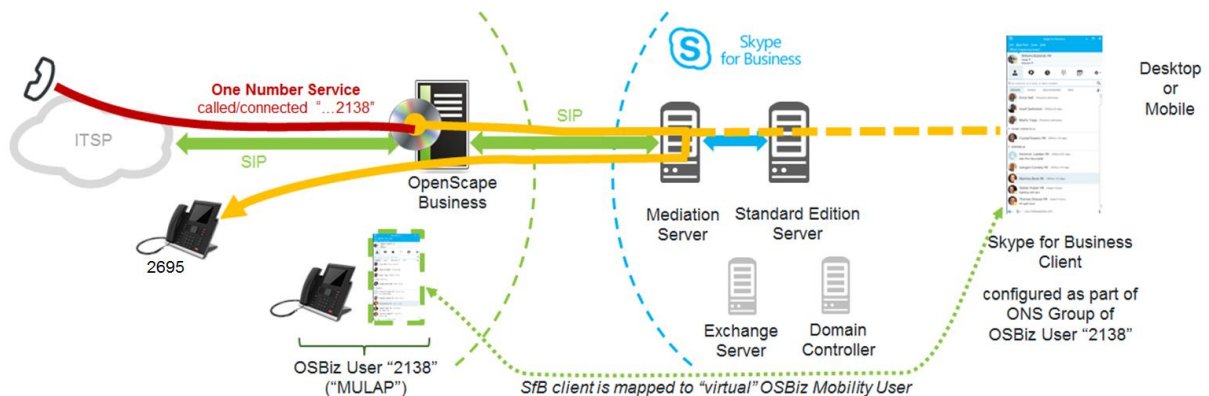
UC Suite example – call has been transferred: User stays busy (trombone connection)



5.6. via Skype GUI: Call Forwarding and Blind Transfer (w/o Consultation)

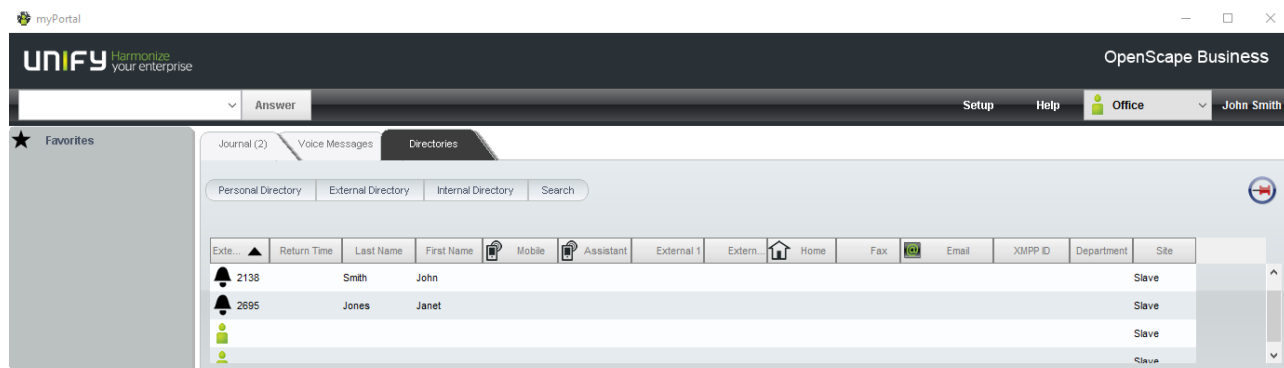
In this example the SfB User transfers or forwards a first call to a alerting second call. The payload connection is handled via SfB Mediation Server and because Path Replacement is not supported both calls stay active in a trombone connections until the transferred resp. forwarded call is released. Meanwhile the Mobility MULAP resp. assigned Mobility User stays busy.

The resulting second call is not assigned to the Mobility MULAP resp. the Mobility User.

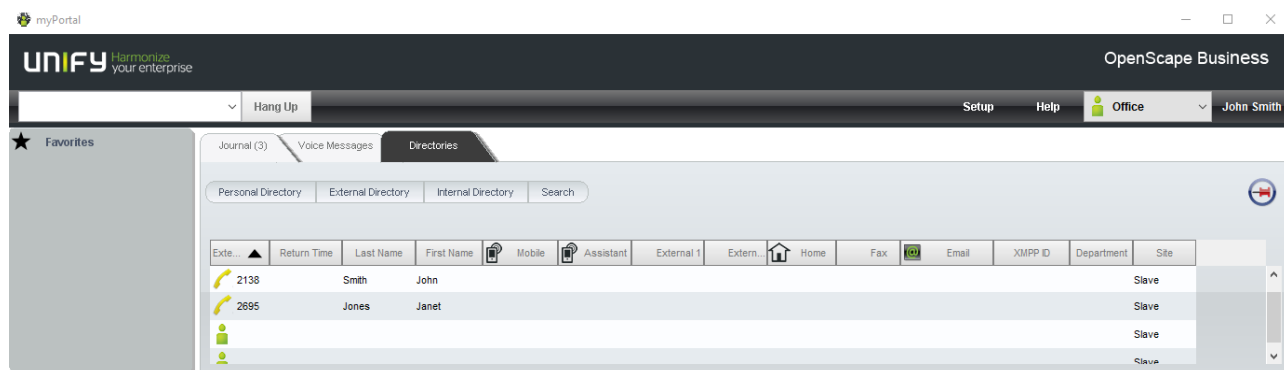


Call Forwarding settings in OpenScope Business and Skype for Business Server are independent from each other. A forwarding setting of OpenScope Business might overrule a forwarding setting of Skype for Business Server and vice versa.

UC Suite example – call is forwarded by Skype for Business Server while OpenScope Business Call Management rules apply:



UC Suite example – SfB User has transferred or forwarded a call: User stays busy



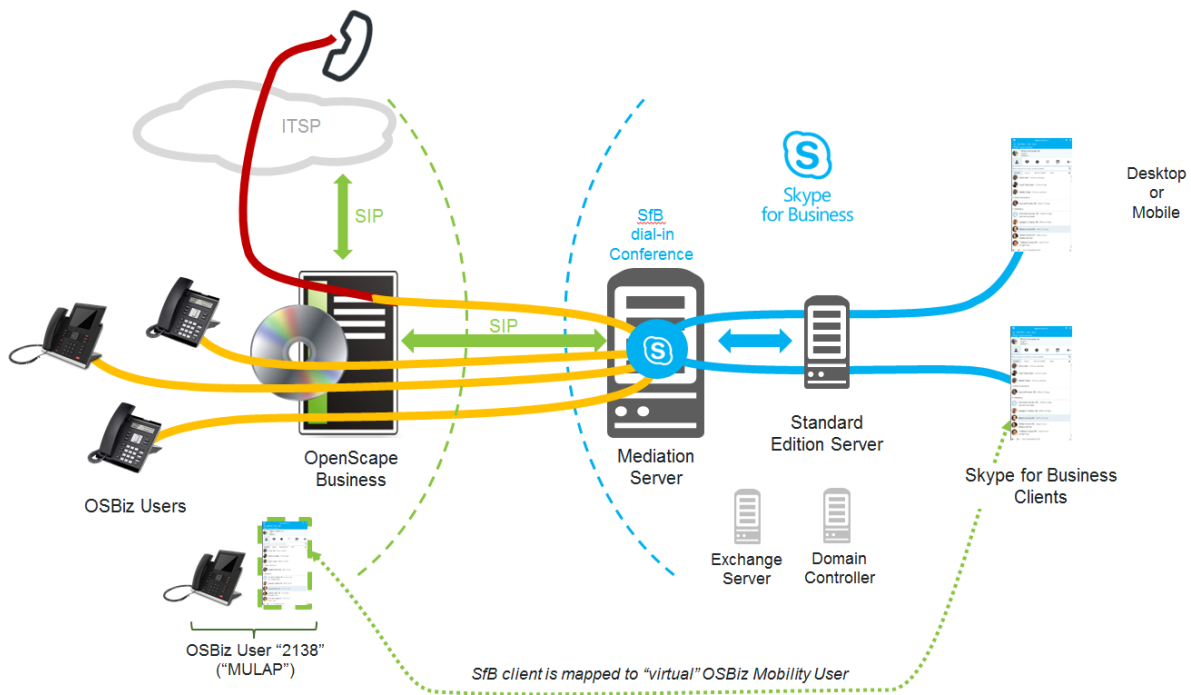
5.7. Conferencing via Skype for Business

In this example the Sfb dial plan is configured in a way that Sfb User dial into a Sfb dial-in conference directly w/o using OSBiz resources (SIP Trunk channels). Only other OSBiz endpoints need a SIP trunk channel (here 3 internal and 1 external).

In this scenario the assigned Mobility User of the Sfb Clients is not claimed. This means that the according OSBiz User is idle.

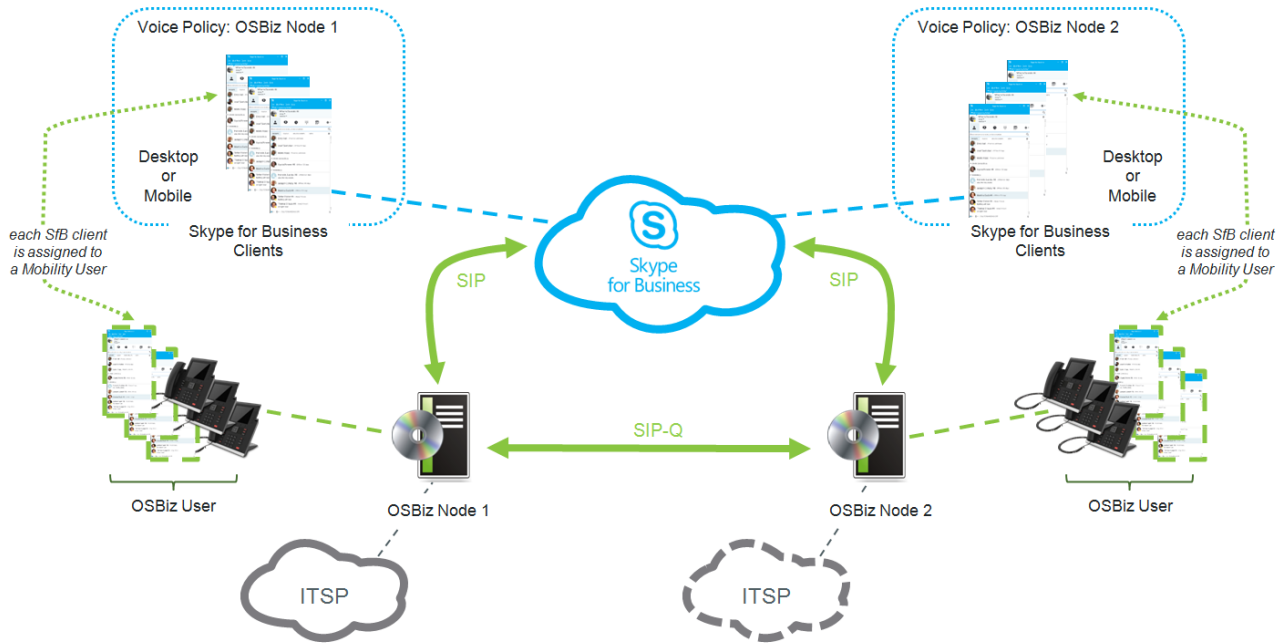
The same conditions apply if the User starts an ad-hoc conference via the “inviting more people” option of the Sfb-Client.

Hint: if an OSBiz User is part of a Sfb conference and the OSBiz User starts a consultation there is no other way to suppress the OSBiz MoH than by muting this endpoint in the Sfb conversation. The OSBiz User can unmute himself once his consultation has ended.



5.8. Networking - Multiple OpenScape Business per SfB

Customers with several OpenScape Business Systems in a network can be addressed to a single Skype for Business Server. This requires a definition of a “IP/PSTN Gateway” per link for each OpenScape Business System. The assignment of the SfB Clients to the OSBiz User is achieved via a “VoicePolicy” per OSBiz node (for details please refer to chapter 2 or according Microsoft documentation). OpenScape Business networking requires a Network License in addition.



6. Capacities & Feature Interaction

Amount of ITSP channels

A trunk license is required per simultaneously used SfB ITSP SIP channel. OSBiz supports up to 60/180 ITSP channels (OSBiz X / OSBiz S) which are shared for ITSP, SfB and others.

Amount of Mobility Users

Each SfB client needs a virtual user representation in OSBiz = Mobility user → Mobility / IP User license required. OSBiz supports up to 150/250 Mobility Users (OSBiz X / OSBiz S)

Codec support

OpenStage/DP IP phones or other calling devices must be configured to offer at least a G.711 codec. The Skype for Business Mediation Server supports only G.711 codec (μ -law and A-law).

Basic Call

The Skype for Business Server doesn't send SIP header P-Asserted-Identity in 180 or 200 messages to convey connected party information. Display names are converted by OpenScape Business.

Call Hold/Retrieve

When a Skype for Business user has placed a call with an OpenScape Business user on hold (and music on hold has been enabled) "RTCPActiveCalls" must be changed from its default setting of "true" to a setting of "false" to prevent held calls from being dropped after 30 sec by the Skype for Business Mediation server.

The OpenScape Business feature held call is not to displayed on Skype for Business and vice versa.

Consultation

Although the consultation call inherits Calling Number and Calling Name and according Class of Service of the Mobility User, the consultation call is not assigned to the Mobility User. This means that only the first connection between OSBiz and an SfB client is monitored by OSBiz in terms of MULAP busy handling. If the external party of this example disconnects, the MULAP of User will be shown as idle to other OSBiz Users even if the consultation connection (second call) remains active.

Example for UC-Suite and DSS-Server: *User B is via his SfB-Client of the OSBiz Mobility User is in a first call with extension A. As long as this first call stays active UC-Suite will show the according Mobility User B busy and the DSS-Server will have the according LED switched on. If the SfB-Client now consults and starts a second call to extension C via the OSBiz this second call bypasses the Mobility User and is handled like an ordinary trunk call and neither UC-Suite nor DSS-Server are aware of linking this second call to the Mobility User B. This 2nd call just inherits call number, name and COS of User B. If now the first call is released UC-Suite will show the*

according Mobility User B idle and the DSS-Server will have the according LED switched off. From then on the Mobility MULAP or the Mobility User B is idle and ready to receive further calls (see chapter 5.4).

Call Forward

Call Forwarding settings in OpenScape Business and Skype for Business Server are independent from each other. A forwarding setting of OpenScape Business might overrule a forwarding setting of Skype for Business Server and vice versa.

The forwarded-to party's display won't show that the call had been forwarded, when the call is forwarded from the OpenScape Business to the Skype for Business domain and vice versa.

The payload connection is handled via the SfB Mediation Server without Path Replacement. Therefore the forwarded call stays active in a trombone connection until the forwarded call is released. Meanwhile the assigned Mobility MULAP resp. the Mobility User is busy due to this call.

The forwarded call is not assigned to the Mobility MULAP resp. the Mobility User and restricted by COS Denied List 1.

Call Transfer

The OpenScape Business does not support Microsoft Skype for Business Server's REFER message.

In order for call transfer scenarios to work properly, REFER support must be disabled for the Skype for Business Server trunk. SIP re-INVITEs will then be used to achieve call transfer scenarios.

In certain call transfer (Attended/Blind) scenarios, user devices (OpenScape Business/Skype for Business) may display the original connected party and not the transferred-to party (see Skype for Business PAI restriction in "Basic Call" scenarios).

The payload connection is handled via the SfB Mediation Server without Path Replacement. Therefore the transferred call stays active in a trombone connection until the transferred call is released. Meanwhile the assigned Mobility MULAP resp. the Mobility User is busy due to this call.

The blind transfer call is not assigned to the Mobility MULAP resp. the Mobility User and restricted by COS Denied List 1.

Conference

There is no conference display indication on OpenScape Business user's phone who has been invited to a Skype for Business conference. On the other hand, at the Skype for Business client there will be no conference indication display when participating in a conference started in OpenScape Business.

When an OpenScape Business subscriber invokes call hold, while being a member of a Skype for Business 2015 conference, MOH is played into the conference by the OpenScape Business.

Depending on SfB configuration the SfB User might dial into conference directly w/o OSBiz SIP Trunk channels. In this scenario the assigned Mobility User is not claimed. This means that the according OSBiz User is idle.

Encryption

OpenScape Business does not support secure media interworking with Skype for Business Server.

LAN/WAN Interface

As Skype for Business Interworking is possible via the LAN interface only, no WAN interface is available as a TCP/IP connection for another ITSP.

Details are available in [5]: Tutorial VoIP Interfaces.

7. Support & Serviceability

7.1. Assistance to resolve OSBiz or SfB-Client related issues

ITSP Status of Skype for Business	<ul style="list-style-type: none">not registeredPlease check for correct IP-Adress and Port. The default Port of Skype for Business Mediation Server is 5068.
ITSP on WAN	<ul style="list-style-type: none">together with SfB Interworking no WAN interface is available as a TCP/IP connection
no calls with SfB-Client possible	<ul style="list-style-type: none">OSBiz is not under Software supportno SfB ITSP lines are configured or all SfB ITSP lines are busyassigned Mobility User does not have a licenseMobility User is not of type “SfB station”the Mobile call number does not match the SfB numbering plan
no outbound calls to SfB-Client possible	<ul style="list-style-type: none">depending on SfB numbering plan the called party number in E.164 requires dialing rule type “Country code”, else the type is “Unknown”
<ul style="list-style-type: none">short call to SfB-Client is aborted	<ul style="list-style-type: none">see “no calls with SfB-Client possible”
no inbound calls from SfB-Client possible	<ul style="list-style-type: none">see “no calls with SfB-Client possible”
Central Office ITSP call are not signalled at SfB-Client	<ul style="list-style-type: none">please check for G.711 on Carrier side – G.729 is not supported by SfB
desk phone calls are not signalled at SfB-Client	<ul style="list-style-type: none">please check for G.711 on phone side – G.729 is not supported by SfB

<p>SfB-Client Hold/Park Call</p> <ul style="list-style-type: none"> • MoH is different to OSBiz MoH or no MoH is provided at all • no “on hold” indication on hold party 	<ul style="list-style-type: none"> • SfB-Client calling feature Park Call must be enabled via SfB Voce Policy • MoH is provided by the SfB-Client • “on hold” indication for Display is not supported
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<p>SfB-Client Consultation Call</p> <ul style="list-style-type: none"> • MoH is different to OSBiz MoH or no MoH is provided at all • no “on hold” indication on hold party • no busy indication for assigned Mobility User or Mobility Mulap if hold call is released (e.g. another incoming call does not follow CFB rules, DSS-LED is off, Hook state is Idle in UCSuite Journal, ...) 	<ul style="list-style-type: none"> • MoH is provided by the SfB-Client • “on hold” indication for Display is not supported • the consultation call does not claim the assigned Mobility User or Mobility Mulap
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<p>SfB-Client Supervised Transfer (via consultation)</p> <ul style="list-style-type: none"> • no update on Display • assigned Mobility User or Mobility Mulap stays busy • transfer call is not successful – call is aborted 	<ul style="list-style-type: none"> • SfB-Client calling feature Transfer Call must be enabled via SfB Voce Policy • SfB Mediation Server does not support update of transferred party information • SfB Server handles Supervised Transfer via a Trombone connection and Path Replacement is not supported • please disable Refer support of SfB Server
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<p>SfB-Client Call Forwarding</p> <ul style="list-style-type: none"> • forwarded call does not follow COS of assigned Mobility User • assigned Mobility User or Mobility Mulap stays busy • Call Forwarding destination is not signalled with original calling party information • display of the forwarded to party does not show the name • display of the transferred to party show invalid number 	<ul style="list-style-type: none"> • SfB-Client calling feature Call Forwarding must be enabled via SfB Voce Policy • Trunk Configuration is enabled to forward call history • V2R4 ITSP template is activated • Call Forwarding of SfB Clients is controlled via COS Denied List 1 (V2R3 only) • SfB Server handles Call Forwarding via a Trombone connection and Path Replacement is not supported • please check CLIP no screening settings in OSBiz and on CO Carrier • name provision is not supported • please check for SfB route type “PABX”
<hr/>	
<p>SfB-Client Blind Transfer (w/o consultation)</p> <ul style="list-style-type: none"> • transferred call does not follow COS of assigned Mobility User • assigned Mobility User or Mobility Mulap stays busy • display of the transferred to party does not show the name • display of the transferred to party show invalid number • transfer call is not successful – call is aborted • after blind transfer the caller hears MoH and ringback tone 	<ul style="list-style-type: none"> • SfB-Client calling feature Transfer Call must be enabled via SfB Voce Policy • Trunk Configuration is enabled to forward call history • V2R4 ITSP template is activated • Blind Transfer of SfB Clients is controlled via COS Denied List 1 (V2R3 only) • SfB Server handles Blind Transfer via a Trombone connection and Path Replacement is not supported • name provision is not supported • please check for SfB route type “PABX” • please disable Refer support of SfB Server • this issue is under investigation
<hr/>	
<p>Cut through Delay after call is transferred</p>	

<ul style="list-style-type: none"> • Cut through Delay of SfB Client calls which are transferred to an Call Center Agent or via Autoattendant might be observed in some configuration 	<ul style="list-style-type: none"> • this issue is under investigation
SfB Conference (dial-in, ad-hoc)	
<ul style="list-style-type: none"> • OSBiz MoH disturbs the conference call 	<ul style="list-style-type: none"> • mute the according OSBiz User in the SfB conversation - the OSBiz User can unmute himself
User is signalled busy (incoming calls receive busy signalling or are forwarded to Voice Mail) although SfB-Client and desk phone are idle	<ul style="list-style-type: none"> • SfB-Client has performed a Supervised Transfer or Blind Transfer and the transferred call is still active => due to missing Path Replacement OSBiz User stays busy until transferred call is released • SfB-Client has activated Call Forwarding and a forwarded call is still active => due to missing Path Replacement OSBiz User stays busy until forwarded call is released
UC Suite or UC Smart clients do not support consultation call while User is on SfB-Client	<ul style="list-style-type: none"> • consultation call of UC Suite or UC Smart clients is not supported by OSBiz while User is on a call with his SfB-Client
DTMF-Feature activation does not work	<ul style="list-style-type: none"> • DTMF-Feature activation within OSBiz is not supported for SfB-Clients • DTMF-based feature activation” – available in OSBiz X - is ignored for Mobility User type “SfB Station” • DMTF transmission of SfB-Client is send transparently through the OSBiz
calls are not reported on CDR	<ul style="list-style-type: none"> • SfB interworking does not support the “incoming call” feature on CDR output

7.2. Required trace configuration options for error reporting

OpenScape Business Trace Profiles:

1. Basic
2. Voice Fax Connections
3. SIP_Interconnection_Subscriber_ITSP

In case of registration issue please activate the OpenScape Business Trace Profile in addition:

4. SIP_Registration

OpenScape Business Trace Components:

1. FP_CP-Port-User: level 9
2. FP_DH-SIP: level 9 (only for OpenScape Business X variant)

7.3. Required trace files for error analysis

OpenScape Business Diagnosis Logs, Wireshark traces

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