



OpenScape Business V3

How to Configure SIP Trunk for Universe SIP Connect





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Table of History

Date	Version	Changes
05.05.2020	1.0	First version



Configuration Wizard

Internet Telephony

Go to Setup -> Central Telephony -> "Internet Telephony"

The screenshot shows the Unify OpenScape Business Assistant interface. The top navigation bar includes 'Home', 'Administrators', 'Setup', 'Expert mode', 'Data Backup', 'License Management', and 'Service Center'. The left sidebar lists various setup categories, with 'Central Telephony' selected. The main content area displays the 'Central Telephony' configuration page, where the 'Internet Telephony' option is circled in red. Below this, there are several other configuration options, each with an 'Edit' button and a brief description.

The overview page appears for entering the location data. The most flexible type of configuration is to enter the Country code only.

The screenshot shows the 'Overview' page for the 'Internet Telephony' configuration. The breadcrumb trail is 'Setup - Wizards - Central Telephony - Internet Telephony'. The page title is 'Overview'. There are two notes: 'Note: changes done in expert mode must be reviewed/repeated after running through the wizard.' and 'Note: At least the configuration of the 'Country code' is needed for features such as 'Internet telephony' and 'MeetMe conference''. The 'PABX number' section contains three input fields: 'Country code: 00 49 (mandatory)', 'Local area code: 0 (optional)', and 'PABX number: (optional)'.

Click [OK & Next].



Provider configuration and activation for Internet Telephony -> No call via Internet -> uncheck

Use County specific view: Germany and select "Universe SIP Connect".

Setup - Wizards - Central Telephony - Internet Telephony

Provider configuration and activation for Internet Telephony

No call via Internet:

Country specific view:

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

	Activate Provider	Internet Telephony Service Provider
<input type="button" value="Add"/>		Other Provider
<input type="button" value="Edit"/>	<input type="checkbox"/>	Unitymedia
<input type="button" value="Edit"/>	<input checked="" type="checkbox"/>	Universe SIP Connect
<input type="button" value="Edit"/>	<input type="checkbox"/>	Verizon
<input type="button" value="Edit"/>	<input type="checkbox"/>	Vodafone Anlagenanschluss
<input type="button" value="Edit"/>	<input type="checkbox"/>	Vodafone Anlagenanschluss R3
<input type="button" value="Edit"/>	<input type="checkbox"/>	Vodafone Anlagenanschluss R4
<input type="button" value="Edit"/>	<input type="checkbox"/>	VoIPXS

Activate Provider and click on [Edit].



On this page the behaviour of the features call forwarding can be controlled:

- "Rerouting active" deactivated (default) -> a call forwarding establishes a second connection and control of the call remains in the system
- "Rerouting active" activated -> Rerouting is carried out in the office during a call forwarding. The system loses further control over the call

Setup - Wizards - Central Telephony - Internet Telephony

Internet Telephony Service Provider

Provider Name: Universe SIP Connect
Enable Provider:
Secure Trunk:
Domain Name:

Provider Registrar

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

Provider Proxy

IP Address / Host name:
Port:

Provider Outbound Proxy

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

Provider Feature

Route optimize active:

Help Abort Back OK & Next Delete Data

Click [OK & Next]



In the next dialog the specific customer SIP Userdata will be configured.

Internet Telephony Stations for BroadCloud	
	Name of Internet Telephony Station
Add	New Internet Telephony Station

Click on [Add].

Internet telephony station: **Username** is inserted here (e.g: pbx_abcdefghij)

Authorization name: **Username** is inserted here (e.g: pbx_abcdefghij)

Password: **Password** provided by Universe

Default number: Main number of connection. The default number is used as outgoing number when no DDI number is assigned to a station. (e.g: +49602144360). Usually the **Lead Number** is entered here.

Internet Telephony Station for Universe SIP Connect

Internet telephony station: pbx_....
Authorization name: pbx_....
Password:
Confirm Password:

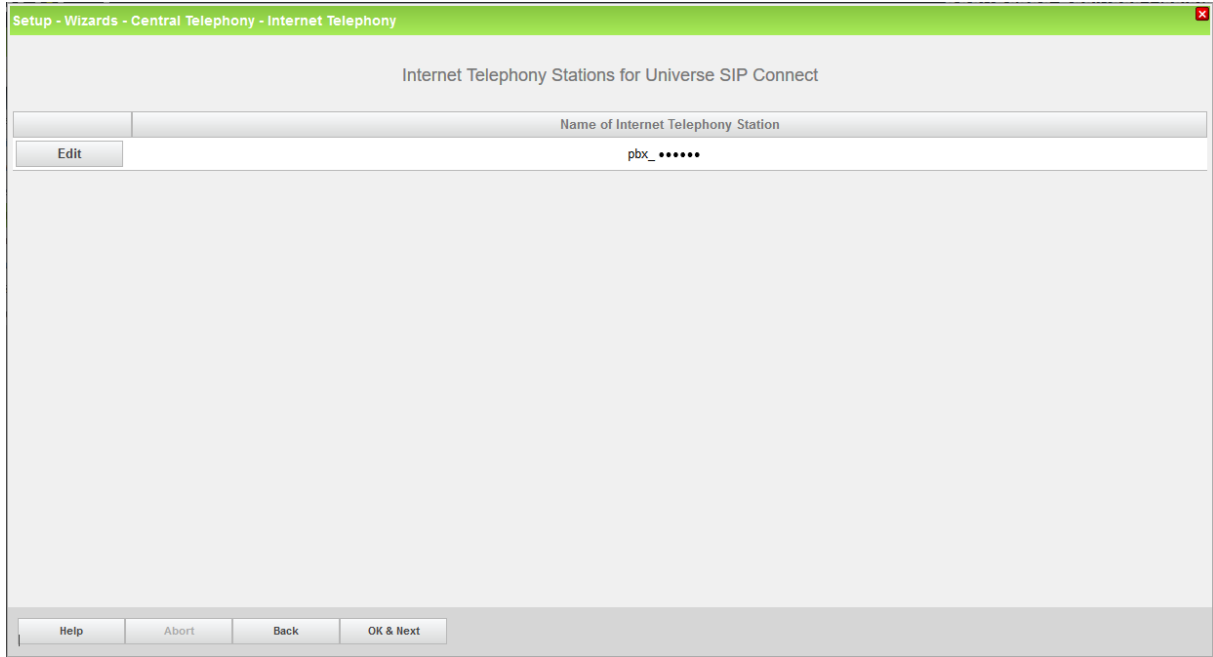
Call number assignment
Use public number (DID)

ITSP-multiple route:
Default Number: +49602144360

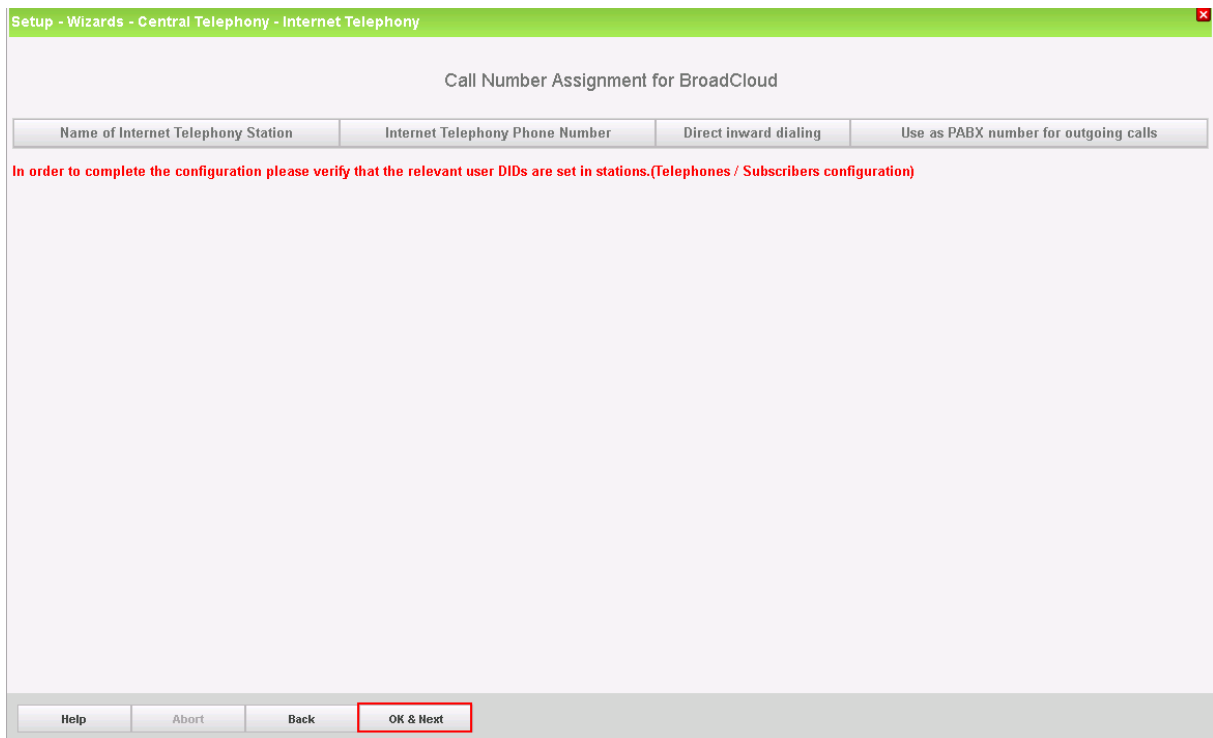
Default Number
ITSP as primary CO access
Enter one of the call numbers supplied by your network provider here. This will be used in outgoing calls as the calling party number in case no other number is available for the respective call. All call numbers supplied by your network provider are to be entered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

Help Abort Back OK & Next Delete Data

Enter the relevant data and click [OK & Next].



Click [OK & Next]



Click [OK & Next] (no input needed)



Setup - Wizards - Central Telephony - Internet Telephony

Provider configuration and activation for Internet Telephony

No call via Internet:

Country specific view:

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

	Activate Provider	Internet Telephony Service Provider
<input type="button" value="Add"/>		Other Provider
<input type="button" value="Edit"/>	<input type="checkbox"/>	Unitymedia
<input type="button" value="Edit"/>	<input checked="" type="checkbox"/>	Universe SIP Connect
<input type="button" value="Edit"/>	<input type="checkbox"/>	Verizon
<input type="button" value="Edit"/>	<input type="checkbox"/>	Vodafone Anlagenanschluss
<input type="button" value="Edit"/>	<input type="checkbox"/>	Vodafone Anlagenanschluss R3
<input type="button" value="Edit"/>	<input type="checkbox"/>	Vodafone Anlagenanschluss R4
<input type="button" value="Edit"/>	<input type="checkbox"/>	VoIPXS

Click [OK & Next]



Define bandwidth (# Trunks)

Insert the Upload speed of your Internet Access.

The amount of simultaneous Internet (**Assigned Lines**) calls must be aligned with the **Maximum Active Calls** you have ordered.

Setup - Wizards - Central Telephony - Internet Telephony

Settings for Internet Telephony

Simultaneous Internet Calls

Available Lines for ITSP: 200

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) = 2000**

In the 'Change Feature -> Internet Telephony' Assistant. This upstream allows you to conduct up to 15 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
Universe SIP Connect	0	<input type="text" value="4"/>

Help Abort Back OK & Next

Click [OK & Next]



Special phone numbers

In this dialog it is possible to route special phone numbers.

The dialog box is titled "Special phone numbers" and contains a note: "Please make sure that all special call numbers are supported by the selected provider without fail." Below the note is a table with three columns: "Special phone number", "Dialed digits", and "Dial over Provider".

Special phone number	Dialed digits	Dial over Provider
1	0C112	Universe SIP Connect
2	0C110	Universe SIP Connect
3	0C0137Z	Universe SIP Connect
4	0C0138Z	Universe SIP Connect
5	0C0900Z	Universe SIP Connect
6	0C118Z	Universe SIP Connect
7	0C116Z	Universe SIP Connect
8	0C115	Universe SIP Connect
9	0C010Z	Universe SIP Connect
10		Universe SIP Connect
11		Universe SIP Connect
12		Universe SIP Connect
13		Universe SIP Connect
14		Universe SIP Connect
15		Universe SIP Connect

At the bottom of the dialog are buttons for "Help", "Abort", "Back", and "OK & Next".

Click [OK & Next]

On next page status of ITSP is displayed.

The dialog box is titled "Status for the Internet Telephony Service Provider (ITSP)". It shows a table with columns for "Provider" and "User".

Provider	User
Universe SIP Connect	pbx_***** registered

Buttons for "Restart" and "Diagnose" are visible. A green status indicator is shown on the left.

Click [Next]

„Exchange Line Seizure“:

Select which trunk will 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)

The dialog box is titled "Exchange Line Seizure". It has two sections: "Exchange Line Seizure" and "Area Code".

In the "Exchange Line Seizure" section, there is a field for "Trunk Access Code" with the value "0". Below it is a dropdown menu for "Dial over Provider" set to "Universe SIP Connect".

In the "Area Code" section, there is a text field for "Local area code" with the value "0 6021".

Click [OK & Next]



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'
ISDN	88
BroadCloud	0

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



DID configuration

In the DID Section, the full DID will need to be entered without the country code.
(e.g.: 60214436100)

UP0 Stations						
Edit Subscriber		UP0 Master/Slave		Device Info		
Callno	DID	First Name	Last Name	Display	Clip/Lin	Activ
Search:						
100	60214436100	-	-	TDM - 100	-	✓
101	60214436101	-	-	TDM - 101	-	✓
102	60214436102	-	-	TDM - 102	-	✓
-	-	-	-	-	-	-



Additional Configuration

License

Add the "S2M/SIP Trunk" license to the SIP-Trunk

Home Administrators Setup Expert mode Data Backup License Management Service Center

License Management

- License information
- Additional Products
 - OpenScape Personal Edition
- Local User licenses
 - Overview
 - IP User
 - TDM User
 - Mobility User
 - Deskshare User
- CO Trunks**
- System Licenses

CO Trunks

The access to central office via PRI(S2m/T1) trunks or via Internet telephony is licensed by CO trunk licenses
Available licenses for SIP and PRI(S2m/T1) trunks: 0

SIP trunks

The configured number of simultaneous Internet calls for each Internet Telephony Service Provider is: 4
License number of simultaneous Internet calls in this node: 4
License demand for number of simultaneous Internet calls in this node:

PRI (S2M/T1)

Type Slot	Port	Feature	Demands	used licenses
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Mandatory configuration in Expert Mode

Codec Parameters

Go to Expert Mode → Telephony Server → Voice Gateway → Codec Parameters

To comply with the requirements of the Universe SIP Connect the following codec parameters **MUST** be changed:

1. G.729A and G.729AB is **NOT** supported by Universe SIP Connect and **SHOULD** be disabled.
2. T38 fax protocol is not supported by Universe SIP Connect. Fax is supported via G.711 only. For this reason you **MUST** disable T38 protocol.

Expert mode - Telephony Server

Voice Gateway

- SIP Parameters
- ITSP Loc-ID Settings
- Codec Parameters**
- Destination Codec Parameters
- Internet Telephony Service Provider
- Networking
- SIPQ-Interconnection
- Native SIP Server Trunk

Codec Parameters

Edit Codec Parameters

Codec	Priority	Voice Activity Detection	Frame Size
G.711 A-law	Priority 1	VAD: <input type="checkbox"/>	20 msec
G.711 μ-law	Priority 2	VAD: <input type="checkbox"/>	20 msec
G.729A	not used	VAD: <input type="checkbox"/>	20 msec
G.729AB	not used	VAD: <input checked="" type="checkbox"/>	20 msec

Enhanced DSP Channels

Use G.711 only

T.38 Fax

T.38 Fax:

Use FillBitRemoval:

Max. UDP Datagram Size for T.38 Fax (bytes): 1472

Error Correction Used for T.38 Fax (UDP): t38UDPRedundancy

T.30 Fax

Enable ECM:

Misc.

ClearChannel: Frame Size: 20 msec

RFC2833

Transmission of Fax/Modem Tones according to RFC2833:

Transmission of DTMF Tones according to RFC2833:

Payload Type for RFC2833: 98

Redundant Transmission of RFC2833 Tones according to RFC2198:

Apply Undo Help