

# OpenScape Business V3



## How to Configure SIP Trunk for: - AAPT SIP Connect

### Australia

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## About this document

This configuration guide describes an example of how to set up the SIP trunk AAPT SIP Connect as an ITSP connection to the OpenScape Business.

**Note:** The basis for this document is the current OpenScape Business V3 R2. Since OpenScape Business is constantly developed, input masks and interfaces as well as requirements may change in the future. The settings and entries described here then apply accordingly.

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## Table of History

Date	Version	Changes
2022-01-13	1.0	released version for OpenScape Business V3R2

## Information

The AAPT SIP Connect SIP-Trunk will be released for the first time with OpenScope Business V3R2.

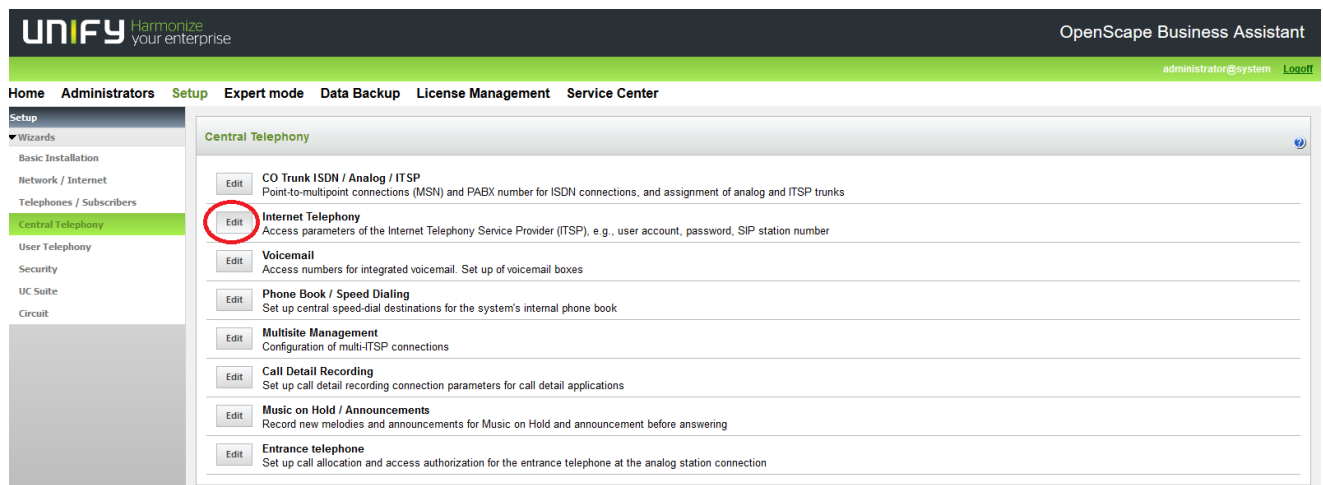
## Trunk Configuration Data provided by AAPT SIP Connect

Via email

## Configuration Wizard

### Internet Telephony

Go to Central Telephony – “Internet Telephony”



The overview page appears for entering the location data. The Country Code, Area code and PABX number is entered (minus the DID which is entered in the station 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:	0011	61	(mandatory)
Local area code:	0	7	(optional)
PABX number:	30510		(optional)

Click [OK & Next].

Provider configuration and activation for Internet Telephony -> No call via Internet -> uncheck  
 Use County specific view: Australia and select AAPT SIP Connect.

Add	Activate Provider	Internet Telephony Service Provider
<input type="button" value="Add"/>		Other Provider
<input type="button" value="Edit"/>	<input checked="" type="checkbox"/>	AAPT SIP Connect
<input type="button" value="Edit"/>	<input type="checkbox"/>	Broadcloud
<input type="button" value="Edit"/>	<input type="checkbox"/>	COLT UK & Europe
<input type="button" value="Edit"/>	<input type="checkbox"/>	COLT VPN
<input type="button" value="Edit"/>	<input type="checkbox"/>	Commander Primus
<input type="button" value="Edit"/>	<input type="checkbox"/>	Engin
<input type="button" value="Edit"/>	<input type="checkbox"/>	gnTel
<input type="button" value="Edit"/>	<input type="checkbox"/>	Internode
<input type="button" value="Edit"/>	<input type="checkbox"/>	Skype Connect
<input type="button" value="Edit"/>	<input type="checkbox"/>	Skype for Business

Activate Provider and click on [Edit].

On the next page enter the following information:

- **Domain Name**

The **SIP Domain Name** and **Registrar Host name** can be found on the paperwork/email provided by AAPT SIP Connect. The SIP Domain Name and Registrar host name are valid for the State that the system resides. Choose the appropriate one from the list below or take the one directly from the paperwork: –

- vic.sip-t.aaptbc.com.au
- nsw.sip-t.aaptbc.com.au
- qld.sip-t.aaptbc.com.au
- tas.sip-t.aaptbc.com.au
- sa.sip-t.aaptbc.com.au
- nt.sip-t.aaptbc.com.au
- wa.sip-t.aaptbc.com.au

The Port is set to “0” in each case to allow for DNS-SRV.

The **Provider Outbound Proxy** is not used and therefore left deactivated, **Route optimize** is not used and should be left unchecked.

Setup - Wizards - Central Telephony - Internet Telephony

### Internet Telephony Service Provider

Provider Name:

Enable Provider: ☒

Secure Trunk: ☐

Domain Name:

Transport protocol:

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**Provider Registrar**

Use Registrar: ☒

IP Address / Host name:

Port:

Reregistration Interval at Provider (sec):

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**Provider Proxy**

IP Address / Host name:

Port:

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**Provider Outbound Proxy**

Use Outbound Proxy: ☐

IP Address / Host name:

Port:

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**Provider Inbound Proxy**

Use Inbound Proxy: ☐

IP Address / Host name:

Port:

Click [OK & Next].

In the next dialog the specific customer SIP Credentials are configured.

Internet Telephony Stations for AAPT SIP Connect	
<b>Add</b>	Name of Internet Telephony Station
	New Internet Telephony Station

Click on [Add].

Data and Credentials are provided on the AAPT SIP Connect paperwork/email

**Internet telephony station:** Username is inserted here (e.g: **0735323821**)  
**Authorization name:** Username is inserted here (e.g: **0735323821**)  
**Password:** Password provided by AAPT SIP Connect  
**Default number:** The Main number of ITSP connection. The default number is used as outgoing number when no DDI number is assigned to a station.  
(e.g: 0730510813).  
Usually the **Main Number** is entered here.

Internet telephony station: 0735323821  
Authorization name: 0735323821  
Password: \*\*\*\*\*  
Confirm Password: \*\*\*\*\*

**Call number assignment**  
Use public number (DID) [v]  
ITSP-multiple route: ☐  
Default Number: 0730510813

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



Setup - Wizards - Central Telephony - Internet Telephony

Internet Telephony Stations for AAPT SIP Connect

	Name of Internet Telephony Station
Edit	0735323821

Help Abort Back OK & Next

Click [OK & Next]

Setup - Wizards - Central Telephony - Internet Telephony

Call Number Assignment for AAPT SIP Connect

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)			

Help Abort Back 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: Australia

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

	Activate Provider	Internet Telephony
Add		Other Provider
Edit	<input checked="" type="checkbox"/>	AAPT SIP Connect
Edit	<input type="checkbox"/>	Broadcloud
Edit	<input type="checkbox"/>	COLT UK & Europe
Edit	<input type="checkbox"/>	COLT VPN
Edit	<input type="checkbox"/>	Commander Primus
Edit	<input type="checkbox"/>	Engin
Edit	<input type="checkbox"/>	gnTel
Edit	<input type="checkbox"/>	Internode
Edit	<input type="checkbox"/>	Skype Connect
Edit	<input type="checkbox"/>	Skype for Business
Edit	<input type="checkbox"/>	Telstra Australia
Edit	<input type="checkbox"/>	Verizon
Edit	<input type="checkbox"/>	VoIPXS

Help
Abort
Back
OK & Next
Display Status

Click [OK & Next]

## Define bandwidth (# Trunks)

The amount of simultaneous Internet (**Assigned Lines**) calls must be aligned with the **Maximum Active Calls** assigned to the Trunk Group on the AAPT SIP Connect paperwork/email.

Setup - Wizards - Central Telephony - Internet Telephony

Settings for Internet Telephony

**Simultaneous Internet Calls**  
Available Lines for ITSP: 174  
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) = 10000**  
In the 'Change Feature -> Internet Telephony' Assistant. This upstream allows you to conduct up to 78 Internet phone calls simultaneously. If the call quality deteriorates due to the network load, you will need to reduce this number of simultaneous Internet Calls. The number of simultaneous Internet Calls also depends on the licensing.

Upstream up to (Kbps): 10000  
Number of Simultaneous Internet Calls: 4 Distribute Lines

**Line assignment**

Internet Telephony Service Provider	Configured Lines
AAPT SIP Connect	0

4

Help Abort Back **OK & Next**

Click [OK & Next]

## Special phone numbers

In this dialog it is possible to route special phone numbers. Use 000 to contact Emergency Services in Australia.

Special phone number	Dialed digits	Dial over Provider
1	0000	AAPT SIP Connect
2		AAPT SIP Connect
3		AAPT SIP Connect
4		AAPT SIP Connect
5		AAPT SIP Connect
6		AAPT SIP Connect
7		AAPT SIP Connect
8		AAPT SIP Connect
9		AAPT SIP Connect
10		AAPT SIP Connect
11		AAPT SIP Connect
12		AAPT SIP Connect
13		AAPT SIP Connect
14		AAPT SIP Connect
15		AAPT SIP Connect

Click [OK & Next]

On next page status of ITSP is displayed.

Provider	Status	Access Code	User
AAPT SIP Connect	Enabled	0735323821	registered

Click [Next]

„Exchange Line Seizure“:

Select which trunk will access code 0.

Trunk Access Code	Dial over Provider
889	AAPT SIP Connect

Click [OK & Next]

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

Seizure Code for the 'Outside line Seizure'	
AAPT SIP Connect	0

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

## DID configuration

In the DID Section, only the last 3 digits of the national number is required.

Calling	DID	First Name	Last Name	Display	Clip/Lin	Active	Device Type	Fax Calln
100	812	UP0e 1	-	UP0e 1	-	✓	OpenStage 40	-
101	823	UP0e 2	-	UP0e 2	-	✓	OpenScape Desk Phone CP 400T	-
102	824	UP0e 3	-	UP0e 3	-	✓	OpenScape Desk Phone CP 400T	-
103	825	UP0e 4	-	UP0e 4	-	✓	OpenScape Desk Phone CP 400T	-

Configure the DID numbers for the IP Clients as well

## Additional Configuration

### License

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

The screenshot shows the 'License Management' section of a software interface. The left sidebar contains a tree view with 'License Management' expanded, showing 'License information', 'Additional Products', 'OpenScape Personal Edition', 'Local User licenses', 'Overview', 'IP User', 'TDM User', 'Mobility User', 'Deskshare User', 'CO Trunks', 'System Licenses', 'License Profiles', and 'Create Profiles'. The main area is titled 'CO Trunks' and contains the following text:

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: 246

**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
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### Route Configuration (Best practice)

We have included these settings as most of the ITSP services in Australia will not work without these settings and they are not default.

Change Route: -

Seizure code: 0

Location number: *ticked*

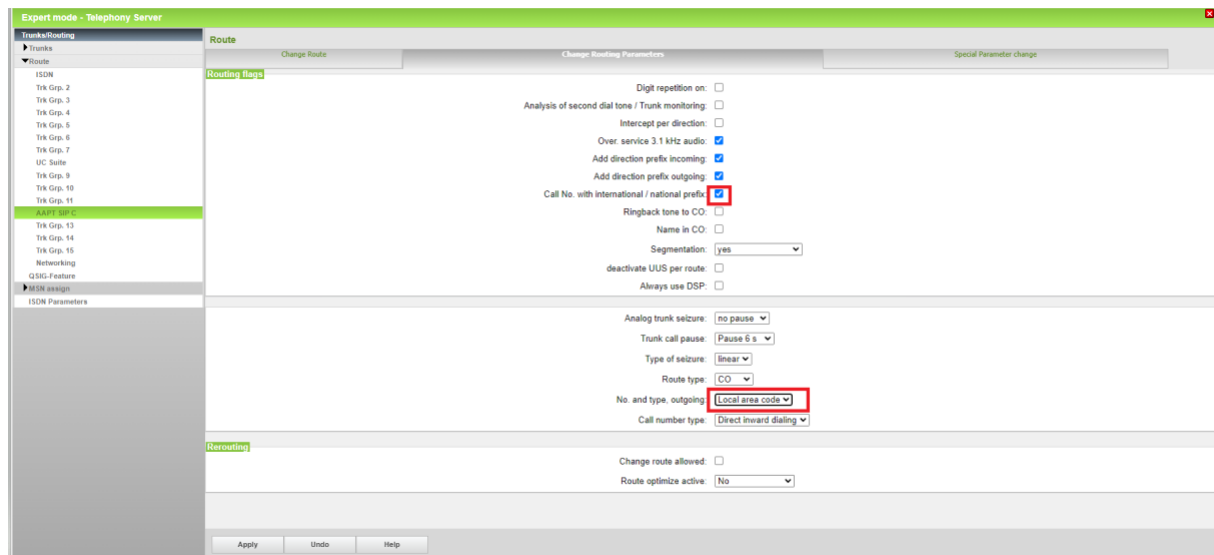
Change Routing Parameters: -

Call No. with international / national prefix: *ticked*

No. and type, outgoing: *Local area code*

The screenshot shows the 'Route' configuration dialog box. The left sidebar contains a tree view with 'Trunks' expanded, showing 'ISDN', 'Trk Grp. 2', 'Trk Grp. 3', 'Trk Grp. 4', 'Trk Grp. 5', 'Trk Grp. 6', 'Trk Grp. 7', 'Trk Grp. 8', 'Trk Grp. 9', 'Trk Grp. 10', 'Trk Grp. 11', 'AAPT SIP C', 'Trk Grp. 13', 'Trk Grp. 14', 'Trk Grp. 15', 'Networking', 'QSIG Feature', 'ISDN Parameters', and 'ISDN Parameters'. The main area is titled 'Route' and contains the following fields:

Route Name: AAPT SIP C  
Seizure code: 0  
CO code (2nd trunk code):  
Gateway Location: Country code: 61, Local area code: 7, PABX number: 30510  
PABX number (incoming): Country code: 61, Local area code: 7, PABX number: 30510, Location number: ☒  
PABX number (outgoing): Country code: 61, Local area code: 7, PABX number: 30510, Suppress station number: ☐  
Overflow route: Overflow route: None  
Digit transmission: Digit transmission: en-bloc sending  
Mobile Extension Number (MEX): MEX Number:  
Trusted External Users: Trusted External Users: ☐



## Known limitations and restrictions:

- Faxing is based on G.711, T.38 is not supported by AAPT SIP Connect
- Route Optimization is not certified for AAPT SIP Connect
- TLS was not tested.

## Mandatory configuration in Expert Mode

### Port management – no change

Go to Expert Mode → Telephony Server → Basic Settings → Port Management

Default port configuration was used during the certification, *no changes required*.

Expert mode - Telephony Server			
Basic Settings			
Port Management			
Edit Global Port Management Settings			
Protocol Name	Port Number		
CSP	8800		single
HFA	4060		single
HFA_EXT	4062		single
HFA_TLS	4061		single
HFA_TLS_EXT	4063		single
MEB_SIP	15060		single
RTP_MIN	29100		min. (ext. RTP-port range 30274-30529)
SIP	5060		single
SIP_EXT	5070		single
SIP_TLS_SUB	5062		single
SIP_TLS_SUB_EXT	5071		single
SIPS	5061		single
VSL_MULTISITE	8778		single

## Codec Parameters, deactivate T.38 Fax

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

T38 fax protocol is not supported by AAPT SIP Connect.

Fax is supported via G.711 only.

Untick the T.38 Fax box to deactivate.

The screenshot shows the 'Expert mode - Telephony Server' configuration window. The left sidebar lists various configuration categories, with 'Codec Parameters' selected. The main area displays the 'Edit Codec Parameters' configuration. A table lists codecs and their priorities. Below this, the 'Enhanced DSP Channels' section is expanded, showing the 'T.38 Fax' checkbox, which is circled in red. Other settings include 'Use G.711 only', 'Use FIBBRemoval', 'Max. UDP Datagram Size for T.38 Fax (bytes)', 'Error Correction Used for T.38 Fax (UDP)', 'Enable ECM', 'ClearChannel', 'Frame Size', 'Transmission of Fax/Modem Tones according to RFC2833', 'Transmission of DTMF Tones according to RFC2833', 'Payload Type for RFC2833', and 'Redundant Transmission of RFC2833 Tones according to RFC2198'.

Codec	Priority	Voice Activity Detection	Frame Size
G.711 A-law	Priority 1	VAD <input type="checkbox"/>	20 msec
G.711 u-law	Priority 2	VAD <input type="checkbox"/>	20 msec
G.729A	Priority 3	VAD <input type="checkbox"/>	20 msec
G.729AB	Priority 4	VAD <input type="checkbox"/>	20 msec

**Enhanced DSP Channels**

Use G.711 only ☐

**T.38 Fax** ☒ (Circled in red)

Use FIBBRemoval ☐

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

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

**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: ☐

Reboot system after applying changes, in order to take effect