**AAPT** [BusinessConnect]

# **OpenScape Business V3**

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

Australia

# 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.

# **Table of Contents**

Information	
Trunk Configuration Data provided by AAPT SIP Connect	4
Configuration Wizard	4
Internet Telephony	4
Define bandwidth (# Trunks)	10
Special phone numbers	11
DID configuration	12
Additional Configuration	
License	13
Route Configuration (Best practice)	
Known limitations and restrictions:	
Mandatory configuration in Expert Mode	
Port management – no change	14
Codec Parameters, deactivate T.38 Fax	15

# **Table of History**

Date	Version	Changes	
2022-01-13	1.0	released version for OpenScape Business V3R2	
2024-09-10	1.1	editorial changes	

**Note**: The basis for this document is the current OpenScape Business at the time of certification. 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.

## Information

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

#### **Trunk Configuration Data provided by AAPT SIP Connect**

Via email

# **Configuration Wizard**

#### **Internet Telephony**

Go to Central Telephony – "Internet Telephony"

		Logoff
Home Administrators Set	tup Expert mode Data Backup License Management Service Center	
Setup		
▼ Wizards	Central Telephony	0
Basic Installation		-
Network / Internet	Editt CO Trunk ISDN / Analog / ITSP Point-to-multipoint connections (MSN) and PABX number for ISDN connections, and assignment of analog and ITSP trunks	
Telephones / Subscribers Central Telephony	Lattr Internet Telephony Access parameters of the Internet Telephony Service Provider (ITSP), e.g., user account, password, SIP station number	
User Telephony Security	Editt Voicemail Access numbers for integrated voicemail. Set up of voicemail boxes	
UC Suite Circuit	Edit. Phone Book / Speed Dialing Set up central speed-dial destinations for the system's internal phone book	
	Editt Multisite Management Configuration of multi-ITSP connections	
	Editt Call Detail Recording Set up call detail recording connection parameters for call detail applications	
	Editt Music on Hold / Announcements Record new melodies and announcements for Music on Hold and announcement before answering	
	Editt Entrance telephone Set up call allocation and access authorization for the entrance telephone at the analog station connection	

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.

Setup - Wizards - Central Telephony - Internet Telephony			
	Provider configuration and activation for Internet Telephony		
Note: changes done in ex	pert mode must be reviewed/repeated after running through the w	No call via Internet.  Country specific view.  Australia	
	Activate Provider	Internet Telephony Service Provider	
Add	_	Other Provider	
Edit		AAPT SIP Connect	
Edit		Broadcloud	
Edit		COLT UK & Europe	
Edit		COLT VPN	
Edit		Commander Primus	
Edit		Engin	
Edit		gn Tel	
Edit		Internode	
Edit		Skype Connect	
Edit		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 Telephon	y Service Provider
Provider Name:	AAPT SIP Connect
Enable Provider:	
Secure Trunk:	
Domain Name:	qld.sip-t.aaptbc.com.au
Transport protocol:	udp 🗸
Provider Registrar	
Use Registrar:	
IP Address / Host name:	qld.sip-t.aaptbc.com.au
Port	
Reregistration Interval at Provider (sec)	600
Provider Proxy IP Address / Host name:	old.sip-t.aaptbc.com.au
Port	
Provider Outbound Proxy	
Use Outbound Proxy:	
IP Address / Host name:	0.0.0
Port:	0
Provider Inbound Proxy	-
Use Inbound Proxy:	
IP Address / Host name:	0.0.0.0
Port:	0

Click [OK & Next].

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

Setup - Wizards - Cen	tral Telephony - Internet Telephony
	Internet Telephony Stations for AAPT SIP Connect
	Name of Internet Telephony Station
Add	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 <b>Main Number</b> is entered here.

Setur , Wizarts , Central Telenhony , Internet Telenhony	
Internet Telephony Statio	on for AAPT SIP Connect
Internet telephony station	0735323821
Authorization name	0735323821
Password	••••••
Confirm Password	
Call number assignment	
Use public number (DID)	<b>~</b> ]
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 num All call numbers supplied by your network provider are to be entered within the trunk and telephones configuration (DID field) primary CO access.	ber is available for the respective call.

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

Setup - Wizards - Central Telephony - Internet Tele	ephony
	Internet Telephony Stations for AAPT SIP Connect
	Name of Internet Telephony Station
Edit	0735323821
Help Abort Back	OK & Next

#### Click [OK & Next]

fizards - 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
complete the configuration please verify that the re	levant user DIDs are set in stations.(Telephones / Subscri	bers configuration)	

Click [OK & Next] (no input needed)

Setup - Wizards - Central	Telephony - Internet Telephony	
		Provider configuration and activation for Internet Telephor
		No call via Internet:
		Country specific view: Australia
Note: changes done in expe	ert mode must be reviewed/repeated after running	ig through the wizard.
	Activate Provider	Internet T
Add		Other Provider
Edit		AAPT SIP Connect
Edit		Broadcloud
Edit		COLT UK & Europe
Edit		COLT VPN
Edit		Commander Primus
Edit		Engin
Edit		gnTel
Edit		Internode
Edit		Skype Connect
Edit		Skype for Business
Edit		Telstra Australia
Edit		Verizon
Edit		VoIPXS
Help Ab	ort 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 communicate Upstream up to (Kbps) = 10000	ed by your Provider. You have typed in				
In the 'Change Feature> Internet Telephony' Assistant. This upstream allows you to conduct up to 74	8 Internet phone calls simultaneously. If the call qual	ity deteriorates due to the netw	ork load, you will ne	ed to reduce this number of sim	
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	Configu	red Lines			
AAPT SIP Connect		0		4	
neip Abort Back UK & 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.

Setup - Wizards - Central Telephony - Internet Telephony		
	Special phone numbers	
Note: Please make sure that all special call numbers are supported by the	e selected provider without fall.	
Special phone number	Dialed digits	Dial over Provider
1	0C000	AAPT SIP Connect V
2		AAPT SIP Connect V
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 🗸
Help Abort Back OK & N	ieat	

Click [OK & Next]

On next page status of ITSP is displayed.

Setup - Wizards - Central Telephony - Internet Telephony						
	Statu	us for the Internet Telepl	hony Service Provider (ITSP)			
	Provider			User		
Restart	AAPT SIP Connect	Enabled	0735323821	registered	Diagnose	

Click [Next]

"Exchange Line Seizure": Select which trunk will access code 0.

	×
Exchange Line Seizure	
Trunk Access Cede 889	
Dial over Provider AAPT SIP Connect 💌	

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'	
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.

Expert mode - Telephony Server									
Station	LIPO Stations								
▼ Station	or o outlons		let Subceniber			100 Ma	etor/Claus		
▼UP0 Stations						OPU Ha	ster/ stave		
0 100 UP0e 1	Callno	DD	First Name	Last Name	Display	Clip/Lin	Active	Device Type	Fax Calln
1 101 UP0e 2	Search								
2 102 UP0e 3	Search.	1					_		
3 103 UP0e 4							_		
4 104 -	100	812	UP0e 1	-	UP0e 1	J	<b>~</b>	OpenStage 40	-
5 105 -	101	823	UP0e 2	-	UP0e 2	]-	- [	OpenScape Desk Phone CP 400T	-
6 106 -	102	824	UP0e 3	-	UP0e 3	-	<b>-</b>	OpenScape Desk Phone CP 400T	-
7 107 -	103	825	UP0e 4	-	UP0e 4	-	- 1	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

Home Ad	dministrators	Setup	Expert mode	Data Backup	License Management	Service Center	
License Manag	jement						
License inform	ation	cc	O Trunks				
▼Additional Pr	roducts						
OpenScape P	Personal Edition	The	access to central of	fice via PRI(S2m/T1	) trunks or via Internet telephor	y is licensed by CO trunk licenses	
▼Local User lic	censes	SID	trunko			Available licenses for SIP and PRI(	32m/T1) trunks: 246
Overview		SIF	TUTIKS			The configured number of simultaneou	us Internet calls
IP User						for each Internet Telephony Service	rice Provider is: 4
TDM User						License number of simultaneous Internet ca	Ills in this node: 4
Mobility User	r				Lice	se demand for number of simultaneous Internet ca	Ills in this node: 4 🗸
Deskshare U	lser	PRI	(S2M/T1)				
CO Trunks				Type Slot	Por	Feature	Demands
System License	es				I		
▼License Profi	iles						
Create Profile	les						

#### **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

Expert mode - Telephony Server				
Trunks/Routing	Route			
Trunks	Charge Banks	Channes Dawline Desembles		Consid Descentes aboves
▼Route	Change Koute	Change Houting Parameters		Special Parameter change
ISDN		Paula Nama:	AART SIR C	
Trk Grp. 2		Note Name.	ART SILC	
Trk Grp. 3		Selzure code:	0	
Trk Grp. 4		CO code (2nd trunk code):		
Trk Grp. 6		CO CODE (2110 BUIK CODE).		
Trk Grp. 6	Gateway Location			
IIC Suite		Country code:	61	
Trk Gro. 9		Local area code:	7	
Trk Grp. 10		PABX number:	30510	
Trk Grp. 11	DABX number incoming			
AAPT SIP C		Courte and	64	
Trk Grp. 13		Country code:	61	
Trk Grp. 14		Local area code:	7	
Trk Grp. 15		P101/	0.0540	
Networking		PABX number:	30510	
QSIG-Feature		Location number:		
MSN assign	PABX number-outgoing			
ISDN Parameters		Country code:	61	
		ound four		
		Local area code:	7	
		PABX number:	30510	
			0	
		Suppress station number:	U	
	Overflow route			
		Overflow route :	None 👻	
	Digit transmission			
		Digit transmission	en bloc sending ¥	
		Urgit transmission.	on new containing *	
	Mobile Extension Number (MEX)			
		MEX Number		
	Trusted External Users			
		Trusted External Users:		
	Apply Undo Help			

Trunks/Routing	Dauta			
Trunks	Route			Provide Deservative deserva-
▼Route	Change Kouce			Special Parameter Change
ISDN	Routing flags			
Trk Grp. 2		Digit repetition on:		
Trk Grp. 3		Analysis of second dial tone / Trunk monitoring:		
Trk Grp. 4		Intercent per direction	_	
Trk Grp. 5		intercept per direction.		
Trk Grp. 6		Over. service 3.1 kHz audio:		
UC Suite		Add direction prefix incoming:	<b>~</b>	
Trk Grp. 9		Add direction prefix outgoing:		
Trk Grp. 10		Call Ne with international ( national profes		
Trk Grp. 11		Call No. with international / hational preto.		
AAPT SIP C		Ringback tone to CO:		
Trk Grp. 13		Name in CO:		
Trk Grp. 19		Segmentation	ves V	
Networking				
Q SIG-Feature		deactivate UUS per route:	U	
MSN assign		Always use DSP:		
ISDN Parameters				
		Analog trunk seizure:	no pause 👻	
		Trunk call pause:	Pause 6 s 👻	
		Type of seizure:	linear 🗸	
		Route type:	C0 ¥	
		No. and type, outgoing:	Local area code 🗸	
		Call number type:	Direct inward dialing V	
	Derouting			
	Reforming	Change route allowed:		
		Route optimize active:	No 🗸	
	Apply Undo Help			

## **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  $\rightarrow$  Telephony Server  $\rightarrow$  Basic Settings  $\rightarrow$  Port Management Default port configuration was used during the certification, *no changes required*.

Expert mode - Telephony Server							
Basic Settings	Port Management						
▼System	Edit Global Part Management Settings						
System Flags							
Time Parameters	Protocol Name	Port Number					
Display DISA	CSP	8800	single				
Intercept/Attendant/Hotline	HFA	4060	single				
LDAP	HFA_EXT	4062	single				
Texts Elevible menu	HFA_TLS	4061	single				
Speed Dials	HFA_TLS_EXT	4063	single				
Service Codes	MEB_SIP	15060	single				
HFA Registration Password	RTP_MIN	29100	min. (ext. RTP-port range 30274-30529)				
DynDNS	SIP	5060	single				
Quality of Service	SIP_EXT	5070	single				
Date and Time	SIP_TLS_SUB	5062	single				
Port Management	SIP TLS SUB EXT	5071	single				
Voicemail / Announcement Player	SIPS	5061	single				
Phone Parameter Deployment	VSL_MULTISITE	6778	single				

#### **Codec Parameters, deactivate T.38 Fax**

Go to Expert Mode  $\rightarrow$  Telephony Server  $\rightarrow$  Voice Gateway  $\rightarrow$  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.

Expert mode - Telephony Server					×
Voice Gateway	Codec Parameters				
SIP Parameters		Edit Codec Parameters			
ITSP Loc-ID Settings					
Codec Parameters	Codec	Priority	Voice Activity Detection	Frame Size	
Destination Codec Parameters	G.711 A-law	Priority 1 V	VAD:		20 🗸 msec
Internet Telephony Service Provider	G 711 uslaw	Priority 2 ¥			20 ¥ msec
SIDO Interconnection	0.770	District 2 in	140		20 - 11000
Native SID Server Trunk	G.729A	Phonty 3 V	VAD:		ZU ♥ msec
	G.729AB	Priority 4 🗸	VAD: CI		20 V msec
	Enhanced DSP Channels		-		
		Use G.711 only	0		
	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 V msec	
	RFC2833				
	Tran	smission of Fax/Modern Tones according to RFC2833:			
		Transmission of DTMF Tones according to RFC2833:			
		Payload Type for RFC2833:	98		
	Redundant Tr	ansmission of RFC2833 Tones according to RFC2198:			

Reboot system after applying changes, in order to take effect