

OpenScape Business V3

How to Configure SIP Trunk for: Spark Voice Connect - New Zealand

About this document

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

Note: The basis for this document is the current OpenScape Business *V3R2.1*. 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.

System	OpenScape Business
Released with Version	V3R2.1
Spark Voice Connect SIP	Features & Capabilities
Account (DID/Client)	DID
Multisite	yes - single trunk
CLIP / CLIR	yes / yes
CLIP no Screening	yes
COLP	по
Call Forwarding (302)	по
DTMF (RFC2833/4733)	yes
Codecs G711/G729	yes / yes
T.38 Fax	yes
Secure trunk	по

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Table of History

Date	Version	Changes
2022-12-01	1.0	profile released with OpenScape Business V3R2.1

Information

The Spark Voice Connect SIP-Trunk will be released for the first time with OpenScape Business V3R2 FR1.

Trunk Configuration Data provided by Spark Voice Connect

The configuration data required to setup the SIP trunk is provided via email from your Spark account manager. The password is provided in a separate email. See example below -

🖫 '9 🔿 \uparrow \downarrow $=$ Your Voice Connect SIP configuration de	tails - GLOBALO	COMMS LIMITED	- Tauranga - Message (HTML)	Q Search			-	o x
File Message Help								
Ignore Image: Constraint of the sector of	Move Actio	3	Categorize Follow y Up y Tags Fs Editing	Aloud Reader	Translate Zo	om		^
Your Voice Connect SIP configuration details		-						
	e_Connect_PB) File	(_Interface_Guide	<u>e_v3_02.pdf</u>		← Reply	≪ Reply All	→ Forwar Tue 26/04/20	
,pur ne					I.			
NTP server IP address	s: 122.5	56.252.129 and	d 122.56.252.137					
SIP Outbound		Preference	PROXY ADDRESS - A RR	PROXY ADDRESS	- IP			
proxy address		First	pak0102-p01.spark.co.nz	122.56.253.231				
(choose one):		Second Third	pro0102-p01.spark.co.nz	122.56.254.167 122.56.255.167				
		Third	ric0102-p01.spark.co.nz	122.30.233.167				
SIP Realm:	sip-net.s	park.co.nz						
Context/Authenticatio	n sip-vc.sp	ark.co.nz						
Digest Realm								
SIP Username:	7571428	0						
SIP Password:	To be ser	nt in separate (email.					
Channels:	5							
Test Numbers:	7571427	8 to 75714279	1					
New DDIs:	7571567	0 to 75715679)					

Next: - Your PABX or IT person can use the above details to configure your PABX.

- Please note:

 - se note: The Voice Connect Lead Number is toll barred for security reasons, and is not intended for use for direct calling. The provided test numbers are live and ready to use for testing now, but will be disconnected after your production numbers are migrated. The new numbers listed are live and ready to use immediately once your PBX is configured. Your PABX system should have a Permit To Connect (PTC). For support in configuring your PABX, please contact the organisation who arranged the PTC.

Any questions? Get in touch

If you have any questions in the meantime, please don't hesitate to get in touch. Reply to this email or call us on 0800 763 772, option 4.

Configuration Wizard

Internet Telephony

Go to Central Telephony – "Internet Telephony"

	nize Iterprise	OpenScape Business Assistant
		administrator@system Logoff
Home Administrators S	etup Expert mode Data Backup License Management Service Center	
Setup		
▼ Wizards	Central Telephony	0
Basic Installation Network / Internet Telephones / Subscribers Central Telephony	Edit C0 Trunk ISDN / Analog / ITSP Point-to-multipoint connections (MSN) and PABX number for ISDN connections, and assignment of analog and ITSP trunks Internet Telephony Access parameters of the Internet Telephony Service Provider (ITSP), e.g., user account, password, SIP station number	
User Telephony Security	Edit 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	
	Edit Multisite Management Configuration of multi-ITSP connections	
	Edit Call Detail Recording Set up call detail recording connection parameters for call detail applications	
	Edit Music on Hold / Announcements Record new melodies and announcements for Music on Hold and announcement before answering	
	Edit 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 most flexible type of configuration is to enter the Country code only.

Setup - Wizards - Central Telephony - Internet Telephony			
	Ove	view	
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	64	(mandatory)
	Local area code: 0	7	(optional)
	PABX number:	[(optional)

Click [OK & Next].

Provider configuration and activation for Internet Telephony -> No call via Internet -> uncheck Use County specific view: New Zealand and select "Spark Voice Connect".

Setup - Wizards - Cent	ral Telephony - Internet Telephony	
		Provider configuration and activation for Internet Telephony
		No call via Interne Country specific view: <mark>New Zealand ❤</mark>
Note: changes done in e	expert mode must be reviewed/repeated after running through the Activate Provider	he wizard. Internet Telephony Service Provider
Add		Other Provider
Edit		AAPT SIP Connect
Edit		Broadcloud
Edit		COLT UK & Europe
Edit		COLT VPN
Edit		gnTel
Edit		Orcon Ltd
Edit		Skype Connect
Edit		Skype for Business
Edit		Spark Voice Connect SIP
Edit		Telstra Clear WSIP

Activate Provider and click on [Edit].

On the next page enter the following information:

- **Domain Name** the SIP Domain Name can be found in the Spark email can be found in the section called SIP Realm.
- **Provider Registrar** can be found under the section Context/Authentication
- **Provider Proxy** is the same as the Domain Name
- **Provider Outbound Proxy** is found on the same email under SIP Outbound proxy address (Choose one).

Setup - Wizards - Central Telephony - Internet Telephony	
Internet Telephon	y Service Provider
Provider Name:	Spark Voice Connect SIP
Enable Provider:	
Secure Trunk:	
Domain Name:	sip-net.spark.co.nz
Transport protocol: Provider Registrar	tcp 🗸
Use Registrar:	
IP Address / Host name:	sip-vc.spark.co.nz
Port:	5060
Reregistration Interval at Provider (sec)	3600
Provider Proxy	· · ·
IP Address / Host name:	
Port: Provider Outbound Proxy	5060
Use Outbound Proxy	
IP Address / Host name:	pak0102-p01.spark.co.nz
Port:	5060
Provider Inbound Proxy Use Inbound Proxy:	
IP Address / Host name:	
Port:	
Provider STUN	
Use STUN:	
IP Address / Host name:	
	3478
Provider Feature Route optimize active:	
Help Abort Back OK & Next Delete Data	

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

Click [OK & Next].

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

Setup - Wizards - Central Telephony - Internet Telephony		
	Internet Telephony Stations for Spark Voice Connect SIP	
	Name of Internet Telephony Station	
Add	New Internet Telephony Station	
Edit	75714280	

Click on [Add].

Data provided on the Spark email under SIP Username: and the separate email containing the SIP Password:

Internet telephony station:	Username is inserted here (e.g: 75714280)
Authorization name:	Username is inserted here (e.g: 75714280)
Password:	Password provided in a separate Spark email
Default number:	Main number of connection. The default number is used as the outgoing number when no DDI number is assigned to a station. (e.g: 75714280). Usually the Main Number is entered here.

Setup - Wizards - Central Telephony - CO Trunk ISDN / Analog / ITSP	
Internet Telephony Station for	or Spark Voice Connect SIP
Internet telephony station:	75714280
Authorization name:	75714280
Password:	···· ·
Confirm Password:	
ITSP-multiple route:	
Default Number:	75714280
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 numb All call numbers supplied by your network provider are to be entered within the trunk and telephones configuration (DID field) primary CO access.	per is available for the respective call.

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

etup - Wizards - Central Telephony - Internet Telephony
Internet Telephony Stations for Spark Voice Connect SIP
Name of Internet Telephony Station
Add New Internet Telephony Station
Edit 75714280

Click [OK & Next] (no input needed)

Setup - Wizards -	Central Telephony - Internet Telephony		
		Provider configuration and activation for I	nternet Telephony
		No call via Internet:	~
Note: changes don	e in expert mode must be reviewed/repeated after running thro Activate Provider	ugh the wizard.	Internet Telephony Service Provider
Add	Activate i Tovidei	Other Provider	internet relephony Service i Tovider
Edit		AAPT SIP Connect	
Edit		Broadcloud	
Edit		COLT UK & Europe	
Edit		COLT VPN	
Edit		gnTel	
Edit		Orcon Ltd	
Edit		Skype Connect	
Edit		Skype for Business	
Edit		Spark Voice Connect SIP	
Edit		Telstra Clear WSIP	
Edit		Verizon	
Edit		VoIPXS	
Edit		Voyager	
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 which is defined in the Spark configuration email under Channels:

SIP Realm:	sip-net.spark.co.nz
Context/Authentication Digest Realm	sip-vc.spark.co.nz
SIP Username:	75714280
SIP Password:	To be sent in separate email.
Channels:	5
Test Numbers:	75714278 to 75714279
New DDIs:	75715670 to 75715679

Setup - Wizards - Central Telephony - Internet Telephony								
Simultaneous Internet Calls	Settings for Internet Telephony							
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	d by your Provider. You have typed in							
In the 'Change Feature> Internet Telephony' Assistant. This upstream allows you to conduct up to 78	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 calls.							
The number of simultaneous Internet Calls also depends on the licensing.								
	Upstream up to (Kbps): 10000							
	Number of Simultaneous Internet Calls:	Distribute Lines						
Line assignment								
Internet Telephony Service Provider	Configured Lines	Assigned Lines						
Spark Voice Connect SIP	5	5						

Click [OK & Next]

Special phone numbers

In this dialog it is possible to route special phone numbers. In New Zealand **111 is the emergency** number for Police, Fire and Ambulance.

Setup - Wizards - Central Telephony - Internet Telephony		
	Special phone numbers	
Note:		
Please make sure that all special call numbers are supported by	y the selected provider without fail.	
Special phone number	Dialed digits	Dial over Provider
1	10111	Spark Voice Connect SIP 🗸
2		Spark Voice Connect SIP 🗸
3		Spark Voice Connect SIP 🗸
4		Spark Voice Connect SIP 🗸
5		Spark Voice Connect SIP 🗸
6		Spark Voice Connect SIP 🗸
7		Spark Voice Connect SIP 🗸
8		Spark Voice Connect SIP 🗸
9		Spark Voice Connect SIP 🗸
10		Spark Voice Connect SIP 🗸
11		Spark Voice Connect SIP 🗸
12		Spark Voice Connect SIP 🗸
13		Spark Voice Connect SIP 🗸
14		Spark Voice Connect SIP 🗸
15		Spark Voice Connect SIP 🗸
Help Abort Back OK	& Next	

Click [OK & Next]

On next page status of ITSP is displayed.

Setup - Wizards - Central Telephony - Ir	nternet Telephony				8
	Stat	us for the Internet Telep	hony Service Provider (ITSP)		
	Provider			User	
Restart	Spark Voice Connect SIP	Enabled	75714280	registered	Diagnose

Click [Next]

"Exchange Line Seizure":

Select which trunk will access code 1.
--

Setup - Wizards - Central Telephony - Internet Telephony	
Exchange Line Seizure	Exchange Line Seizure
Exchange Line Seizine	Trunk Access Code 1
	Dial over Provider Spark Voice Connect SIP 🗸

Click [OK & Next]

Overview with all configured "Outside line Seizure" are displayed.

Setup - Wizards - Central Telephony - Internet Teleph	
	Seizure Code for the 'Outside line Seizure'
	Seizure code for 'Outside line Seizure'
Spark Voice Connect SIP	1

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

DID configuration

Enter the DID number including the area code in the Edit subscriber configuration. The CLIP/Lin field can also be modified if a different CLI is required.

UP0 Stations											
or o stations	20023	conditions.			100 11-0	an (Clause			Davies Info		
					OPO Mass	er/stave			Device Info		
Callno	DID	First Name	Last Name	Display	Clip/Lin	Active	Device Type	Fax Callno	Fax DID	Access	ITSF
Search											
						1 (
400)(] [0			101 MILLA & Manhar	
	/5/156/4	<u>-</u>	*	•			OpenStage 40	-			*
101	101					•				SLMU 1-2 Master	
102 -	102		+			ī.				SLMU 1-3 Master	
	Search:	Calino DID Search:	Search: 100 → [5715674 - 101 → 101 -	Calino DID First Name Last Name Seach	Callno DID First Name Last Name Display Search -	Callno DID First Name Last Name Display ClipI.in Seach - - - - - - - - - - - - - - - - - - 101 - - - - - - - 101 -	Callno DID First Name Last Name Display ClipLin Active Seach - - - - - - - - - - - - - - - - 101 - <td< td=""><td>Calino DID First Name Last Name Display Clip/Lin Active Device Type Search 0 > 5715074 - - - OpenStage 40 101 > 101 - - - - OpenStage 40</td><td>Callno DID First Name Last Name Display Clip/Lin Active Device Type Fax Callno Search 00 → [5715674] -</td><td>Callno DID First Name Last Name Display ClipILin Active Device Type Fax Callino Fax DID Seach -</td><td>Callno DID First Name Last Name Display ClipLin Active Device Type Fax Callno Fax DID Access Search - - - - - - - - - - - - - - - - SLMU 1-1 Master 101 - 101 - - - - - SLMU 1-2 Master</td></td<>	Calino DID First Name Last Name Display Clip/Lin Active Device Type Search 0 > 5715074 - - - OpenStage 40 101 > 101 - - - - OpenStage 40	Callno DID First Name Last Name Display Clip/Lin Active Device Type Fax Callno Search 00 → [5715674] -	Callno DID First Name Last Name Display ClipILin Active Device Type Fax Callino Fax DID Seach -	Callno DID First Name Last Name Display ClipLin Active Device Type Fax Callno Fax DID Access Search - - - - - - - - - - - - - - - - SLMU 1-1 Master 101 - 101 - - - - - SLMU 1-2 Master

Additional Configuration

License

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

UNIFY								
Home Administrators	Setup Expert mode	Data Backup License N	lanagement Service (Center				
License Management								
License information	CO Trunks							
 Additional Products 								
OpenScape Personal Edition	The access to central of	fice via PRI(S2m/T1) trunks or via	Internet telephony is licensed	I by CO trunk licenses Available licenses for SIP and PRI(S2m/T1) t	runko: 24E			
▼Local User licenses	SIP trunks			Available licenses for SIF and FRI(3211/11)	IUIIKS. 240			
Overview				The configured number of simultaneous Interne				
IP User				for each Internet Telephony Service Provi	deris: 5			
TDM User		License number of simultaneous Internet calls in this node: 5						
Mobility User			License demand	for number of simultaneous Internet calls in this	node: 5 🗸			
Deskshare User	PRI (S2M/T1)							
CO Trunks		Type Slot	Port	Feature	Demands			
System Licenses								
 License Profiles 								
Create Profiles								
Assign Profiles								
Registration								
Activate License Online								
Activate CLS Connect								
Activate License File								

Trunks/Routing Configuration

Number and type outgoing should be set to Unknown and Call number type set to Direct Inward dialing.

Route		
Change Route	Change Routing Parameters	Special Parameter change
Routing flags		
	Digit repetition on:	
	Analysis of second dial tone / Trunk monitoring:	
	Intercept per direction:	
	Over. service 3.1 kHz audio:	
	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:	C0 •
	No. and type, outgoing:	Unknown V
	Call number type:	Direct inward dialing V

Known limitations and restrictions:

Restrictions about certain use cases observed during certification should be listed

Number presentation for external transferred calls or calls via the Auto Attendant, the A party number is not displayed at the C party, the Default number is displayed instead.

Mandatory configuration in Expert Mode

Port management

Go to Expert Mode \rightarrow Telephony Server \rightarrow Port Management Port management remains default no changes are required.

Basic Settings	Port Management							
▼System	- or management	Edit Global Port Management Settings						
System Flags								
Time Parameters	Protocol Name	Port Number	Port Type					
Display	CSP	8800	single					
DISA			-					
Intercept/Attendant/Hotline LDAP	HFA	4060	single					
Texts	HFA_EXT	4062	single					
Flexible 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					
Gateway	_							
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					
Call Charges	SIPS	5061	-					
Voicemail / Announcement Player			single					
Phone Parameter Deployment Power Management	VSL_MULTISITE	8778	single					

Codec Parameters

Go to Expert Mode \rightarrow Telephony Server \rightarrow Voice Gateway \rightarrow Codec Parameters

To comply with the requirements of Spark Voice Connect the following codec parameters **MUST** be changed:

- 1. RFC 2833 payload type **MUST** be 101.
- 2. T.38 Fax is supported and can be left activated.
- 3. G.729AB and G.729A are NOT supported and SHOULD be disabled.

Expert mode - Telephony Server					
Voice Gateway	Codec Parameters				
SIP Parameters	Edit Codec Parameters				
TSP 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 µ-law	Priority 2 V	VAD:		20 v msec
Networking					
SIPQ-Interconnection Native SIP Server Trunk	G.729A	not used 🗸	VAD:		20 ∨ msec
Native SIP Server Irunk	G.729AB	not used 🛩	VAD: 🖾		20 v msec
	Enhanced DSP Channels				
	Use G.711 only T38 Fax:				
		<			
	Max. UI	1472			
	En	ror Correction Used for T.38 Fax (UDP)	t38UDPRedundancy 🗸		
	T.30 Fax				
	Misc.				
	ClearCh			Frame Size: 20 🕶 msec	
	RFC2833		-		
	Transmission of Fax/Modem Tones according to RFC2833. 🗹 Transmission of DTMF Tones according to RFC2833. 🗹				
	Payload Type for RFC2833 [101]				
	Redundant Transmission of RFC2833 Tones according to RFC2198:				
	Apply Undo Help				