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R10.0

SIP TRUNK SOLUTION OVH (FR): CONFIGURATION GUIDELINE RCE100

This document details how to set up an IPBX OXO RCE100 for enabling a public SIP trunk of the Operator OVH in FRANCE. The different SIP solutions offered by OVH ("Ligne SIP Entreprise", "Ligne SIP individuelle" and "Ligne SIP Trunk") are compatible with OXO R10.0 under the conditions detailed in Chapter 1.

Revision History

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1 General

This document details the process for configuring from scratch a public SIP trunk of the Operator **OVH** on a system OXO RCE100.

The compatibility of OXO R10.0 with the different solutions/ options proposed by OVH is the following (current status):

A) the solution "Ligne SIP individuelle" featuring one single DDI number is COMPATIBLE. Whenever several DDI's are needed by the customer, it's possible to connect more of these individual lines and then use the multi-account function of OXO R10.0.

B) the solution "Ligne SIP Entreprise" featuring several DDI numbers is COMPATIBLE.
C) the solution "Ligne SIP Trunk" featuring several DDI numbers is COMPATIBLE, no matter if the option "Amélioration de la présentation du numéro appelé" is active or not.

The above solutions B and C bring natural DDI numbers on the IPBX which supersede the multi-account configuration of OMC.

The guide is based on the OMC service "SIP Easy Connect" which permits to import a SIP Trunk Profile and to simplify drastically the configuration task.



For an easier OMC configuration and optimized usage of this guide, you should have at your disposal the reference SIP Trunk Profile of the Operator OVH (i.e. the spf file "FR_OVH_RCE100_SIP_edxx.spf" delivered by Alcatel-Lucent).



The doc TC1994 describes the general management of SIP Trunk profiles files in OMC and how to retrieve the up-to-date profile edition of any GA-released Operator. **If you don't have the import file requested here above, please read carefully the addendum of Ch.6 before proceeding.**

1.1 References

Alcatel-Lucent documentation available on the Business Partner Web Site:

- [1] Alcatel-Lucent OmniPCX Office Communication Server Expert Documentation
- [2] Technical Bulletin TC1284 Public SIP Trunking Interoperability and Technical Support Procedure
- [3] Technical Bulletin TC1994 SIP Easy Connect: SIP Trunk Profile Import/Export
- [4] Technical Bulletin TC1143 Security Recommendations For OmiPCXOffice RCE

1.2 Scope & usage of the configuration guide

This guide is intended for normal-skilled engineers who are familiar with OMC and with the very basic set up of the IPBX.

For the basic system configuration, it is essential to consider the state-of-the art rules of security reminded in the bulletin TC1143.

Well-known configurations like that for the IP-LAN or for "Traffic Sharing and Barring" are just reminded without any details. For simplification reasons as well, the description of user menus and the OMC screenshots are based on the selection of the English language in OMC.

Within the guide, each configuration parameter has been assigned a specific name which is proper to the document and derived from the OMC menu path. These parameters can be easily identified via the purple color and the heading sign \Im . Some examples:

- NP_International_Prefix = "00" (between quotes when value is freely editable in OMC)
- **VoIPgen_RTP_Direct = False** (no quotes when value is selected from a pick-list in OMC)
- GWdom_IP_Address = (N/A) (when the parameter is hidden or disabled in OMC)

Although pre-configured by SIP Easy Connect, some few parameters may be subject to additional site tuning and are marked with a distinctive heading sign \Im . Example:

 $rac{1}{\sim} \Phi \sqrt{VoIPdsp_DSP_Echo_Cancel} = True$



The setting values given in the doc must be strictly respected, unless specific note or if the parameter name ends with "**_example**". Indeed, that suffix is a mark indicating that the value is site-dependent and needs to be customized. Example: $\bigcirc \bullet$ Access_Channels_example = 8

As they are taken from a real customer site, the _example values carrying private data have been masked or partially masked with asterisks (i.e. user logins and passwords, public phone numbers...). Example: The second s

1.3 Scope of Alcatel-Lucent's support

The support delivered for this SIP Trunk solution is strictly delimited by the approval context and the system configuration detailed in this document. The protocol and the functional aspects of the SIP trunk are in the scope, but not the audio quality of calls for the part incumbent on the Operator or on the client's infrastructure. Beyond this, the deployment of the solution is submitted to the SLA conditions proper to the support model agreed: either LA mode (Limited Availability) or GA mode (General Availability).

The type of support model (LA or GA) of an approved SIP Trunk solution must be checked from the up-to-date TC1284 doc available on ALE's Web portal.

1.4 Software/ Hardware components on customer's infrastructure

INFRA COMPONENT	MODEL	VERSION (min compatible)
OXO IP-PBX system	Alcatel-Lucent OmniPCX Office	RCEFR100/026.001
OMC Management Application	Alcatel-Lucent OMC	OMC10.0/9.1a
Router with enabled SIP/ALG	Provided by 3 rd party (see Warning Note below)

The SIP Operator does not provide SIP/HNAT on its IP network but delivers to the customer a packaged solution that includes a CPE router with local SIP/ALG. This CPE router managed by the Operator is not in the scope of this document: as a very general rule, troubleshooting and support service of the solution does only cover what is transmitted/ received on OXO's LAN port port.

1.5 Feature List & Set Compatibility

1.5.1 Supported Features & Sets

The following tables list the main inter-operation features and the range of sets that are supported with this SIP Trunk solution. For the different items, refer to the indication given in the Support column which is marked as "OK" (for full support), or "WR" (support with restriction), or "NOK" (for Not OK or Not Applicable), or "NT" (for Not Tested).



In case of any doubt about the compatibility tables hereafter, or, if you want to contribute to the validation of items that are not yet tested, you can notify us via the e-mail address sip-for-smb@alcatel-lucent.com.



OVH (TOPOLOGY B)	40x8 80x8	40x9 80x9 Z DECT	IP DECT DAP's	MyIC 8082	8002 8012	MyIC And.	MyIC IPhone	MyIC SIP
SETS Supported	ок	ок	NT	NT	NT	NT	NT	NT
USER Basic Features								
Outbound Basic Call	ОК	ОК						
Inbound Basic Call	ОК	ОК						
Inbound Call to DDI	ОК	ОК						
Call Release	ОК	ОК						
Call Hold & Music	ОК	ОК						
Emission of DTMF	ОК	ОК						
Reception of DTMF	ОК	ОК						
Internal Call Forward	WR	WR						
Internal Call Transfer	ОК	ОК						
CLIP Inbound	ОК	ОК						
CLIP Outbound	ОК	ОК						
Emergency Calls	ОК	ОК						
USER Extended Features								
External Call Forward	WR	WR						
External Call Transfer	ОК	ОК						
COLP	WR	WR						
Dynamic Call Routing	WR	WR						
Conference with 2 Ext.	ОК	ОК						
Busy State	ОК	ОК						
General Preannouncement	ОК	ок						
CLIR In & Outbound	ОК	ОК						

SYSTEM Global Features	Supported
Outbound Fax T38	WR
Inbound Fax T38	WR
Outbound Fax G711	NT
Inbound Fax G711	NT

1.5.2 Restrictions

External Call Forward, Internal Call Forward and Dynamic Call Routing \rightarrow Issue with COLP. **Inbound & Outbound Fax T38** \rightarrow Not tested in all the scenarios requested.

2 System General Info and Basic Setup

2.1 Pre-required information

The table below gathers the settings inherent to the SIP Trunk and delivered by the Operator (empty values correspond to not relevant or not mandatory parameters). This data must be available for completing the OXO configuration.

Data Type	Parameter role	Name in doc	Value
Specific to	OP Gateway IP@	GWdom_IP_Address	
SIP Trunk	OP Domain name (*)	GWdom_Target_Domain	sip.ovh.fr (or siptrunk.ovh.net)
model	Outbound Proxy	GWdom_Outb_Proxy	
delivered	DNS IP@ (1 st choice)	GWdns_Prim_DNS	8.8.8.8
by OVH	DNS IP@ (2 nd choice)	GWdns_Sec_DNS	
(see note	Registrar IP address	GWreg_Reg_IP_Address	
Delow)	Registrar name (*)	GWreg_Reg_Name	sip.ovh.fr (or siptrunk.ovh.net)
	SIP Realm	GWdom_Realm	
	Local Domain name(*)	GWdom_Local_domain_Name	sip.ovh.fr (or siptrunk.ovh.net)
	Installation Number	NP_Instal_Number_example	97***2347
Site	Instal. Alternative CLI	SIPnum_Alt_CLIP_example	
specific	Public DDI range	NP_DDI_Range_example	2347 2350
(example	Registered Username	SIPaccnt_Reg_Username_example	003397***2347
values)	Authentication Login	SIPaccnt_Login_example	003397***2347
	Password	SIPaccnt_Password	****

OVH SIP TRUNK PARAMETERS



(*) This variant data is ruled by the Operator and is specific to the solution model delivered. Today:

- "sip.ovh.fr" (for solutions "Ligne SIP Entreprise" or "Ligne SIP individuelle") or
- "siptrunk.ovh.net" (for the solution "Ligne SIP Trunk")

ALE may not be aware of changes made by the Operator. In case of set up issue and doubt concerning the SIP Trunk parameters, please contact the Operator directly.

2.2 System Connection procedure

The configuration task involves on-line connection to the IPBX with OMC Expert-level. Setting up the LAN parameters for OXO (i.e. "IP address", "subnet mask" and "Def. Router Address") is consequently the prime action to complete. When connected, we recommend you select the English language in OMC via the menu <u>Options -> Language</u>.

2.3 Checking the SW license

A specific SW licence is mandatory to enable IP trunks on the system. In the OMC tab Hardware and Limits -> Software Key Features -> _Multi-site, check that the number of IP Trunks "Really activated" (i.e. the max number of channels simultaneously usable on the VOIP trunk) is greater than zero and well adapted to the customer site.



2.4 Numbering Plan configuration

2.4.1 Installation numbers

No matter the type of Trunk considered, the system management for public numbers is first based on the "Installation Numbers" data configurable via the OMC menu Numbering -> Installation Numbers.

				- il
03 Numbering	Installation Numbers			
Installation Numbers Default Configuration	Installation Number	1	97***2347	
- I Numbering Plans	International Prefix	2	00	
	International Code	3	33	
Number Modification Table	Intercity Prefix	4	0	
Splitting Table 	Intercity Code	5		
😑 🖴 Automatic Routing Selection	Recall Prefix	6	0	
- Automatic Routing: Prefixe - E Gateway Parameters	Alternative System CLIP	7		
SIP Accounts	Private Installation Number			
SIP Public Numbering 	Private Level2 Prefix			
	Private Level2 Code			i
Day Groups Providers/Destinations	Private Level1 Prefix			!
	Private Level1 Code			
Tone/Pause-MF	VPN Escape Prefix to remove			i

- Check/ edit the corresponding numbers as illustrated above:

Install. Number : **NP Instal Number example = "97***2347" S** • NP_International_Prefix = "00"

• NP_Intercity_Code_example = ""

• NP International Code = "33"

• NP_Intercity_Prefix = "0"

- 2 International Prefix :
- (3) International Code :
- 4 Intercity Prefix :
- 5 Intercity Code :
- 6 Recall Prefix :
 - **NP Recall Prefix = "0"**
- Ø Alternative System CLIP : • NP_System_Alt_CLIP_example = ""

2.4.2 DDI numbers

In OMC, the Public Numbering Plan permits to configure the DDI numbers allocated to the IPBX subscribers. On OMC, open the tab: <u>Numbering -> Numbering Plans - Public Numbering Plan</u>.

M OMC	N	umbering Plans							
		Internal Numbering Pl	las Public	Numbering Plan	Restricted	Public Numbering	Plan Private	Numbering Plan	
Installation Numbers Default Configuration Numbering Plans		Function Subscriber 💌	Start	End 2350	Base 110	NMT Drop	▼ No	▼ Fax	SIP Acc.Index
Features in Conversation		Subscriber	2347	2350	110	Drop	No		1
									:

- Check/ edit the configuration for "Public Numbering Plan":
 - DDI range : PDDI_Range_example = "2347 2350"



2.4.3 Internal Numbering Plan

Accessible from OMC <u>Numbering -> Numbering Plans</u> menu, the internal numbering plan is the place where dialing of internal phones is first analyzed by the OXO call server.

омс	Num	bering Plans		•••••						<u> </u>	• X
C OMC	- ALL WARDING	art of the second se								\vdash	
- 💖 Tools	Int	ernal Numbering Plan	Public	Number	ing P Restric	cted Public N	umbe Private N	Jumbering	g Plan	$\langle \rangle$	
≫	-					_					
K Modification typical		unction		Start	End	Base	NMI	Priv			Add
-03 Numbering		Main Trunk Group	•	0	0	ARS	Drop 🔻	No			Delete
Installation Numbe		Main Trunk Group		0	0	ARS	Drop	No			
Default Configurat		Attendant Call Subscriber		10 110	10 199	9 110	Drop Drop	No No			Modify
		Subscriber		200	299	200	Drop	No			LIn
Features in Conver		Subscriber Secondary Trunk Group		300 500	399 534	1	Drop	No			
DDI Number Modi		Hunting Group		540 60	565 60	540	Drop	No		1	Down
- I Number Modificat		Set Replace		611	611		Drop	No			
		Set Retrieve Cancel Booking		612 63	612 63		Drop Drop	No No	1	1	
- End of Dialing Tab		Call Forwarding		64	65	8	Drop	No			
🗄 💼 Automatic Routing		ALD FIERX		660	001	U	Diop	INO			
Collective Speed Dial											
- 🖁 Subscribers/Basestatic					7						
E 👹 Voice Processing			Car	icei							

This example defines access to the internal ARS table for user dialing starting with digit 0. The "Drop" attribute indicates that the number dialed must be analyzed in the ARS Prefix table after dropping the initial digit 0.

2.5 CLI for external Diversion

For the scenario of External Call Forwarding (i.e. Ext caller A -> Int subscriber B -> Ext destination C), this configuration permits to select the CLIP number transmitted to C (i.e. either A or B). The control can be made globally for all PBX users, or extension by extension.

From the tabs "Part 1" and "Part 2" of menu <u>System Misc -> Feature Design</u>, verify the parameters:



After selecting an individual extension from the menu **Subscribers/Basestation List**, use the **Details** button to access the "Feature Rights" screen and then, adjust the CLI parameter in the same way:

"CLI is diverted party": Misc_CLI_is_Diverted_Party = False

	Subscribers/Ba	sestations Lis		
COMC	Phy. Add.	No.	Terminal/Ba	Add
	01-001-01	110	Advanced	
ALE_IIIKITCh_102	· · · · · · · · · · · · · · · · · · ·			Delete
Customer/Supplier Info	01-001-01	110	Advanced	Modify
- 🔍 Modification typical	01-002-01	111	4029	
03 Numbering	01-003-01	112	Accès UA	Details
- 🖓 Collective Speed Dialing) [01-004-01		Anneslia	~)
	Subscriber		A A A A A A A A A A A A A A A A A A A	
— Subscribers/Basestations List	Diversity			
Voice Processing	Phy. Add.		Keys V 24	
Time Ranges	Name		Features Passwo	d les
Phy. Add. No. Termin	al	Name		
01-001-01 110 Advan	ced	Ext110		
Feature Bights Part 2				
Transfer to external		📝 Join incomi	ng and incoming	
		Bemote cur	stom. Company greeting	
CLI is diverted party		D	tonic company grooting	
Conterence Bridge Allowed		Hemote cus	stomizatióň	
	2			

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2.6 Traffic Sharing and Barring (reminder)

Though it's not described here, a correct configuration of traffic sharing, barring and feature rights mechanisms is necessary to allow call features for the subscribers and outbound calls over the SIP trunk.

3 ENABLING SIP TRUNKING

3.1 Signaling protocol and number of physical Channels

The VoIP trunk uses a specific signaling protocol (i.e. SIP) and some physical resources of the IPBX (i.e. DSP channels). On OMC, open the tab <u>VoIP:Parameters/General</u> and control/ adapt the following parameters that are necessary for creating the SIP trunk in the system:





- ② Number of VoIP-Trunk Channels :

Image: Provide the second second

3.2 Line Access associated to the SIP Trunk

From OMC External Lines -> List of access, select the VoIP access associated to the VoIP trunk.



Alcatel-Lucent OmniPCX Office – R10.0 SIP Trunk Solution OVH (FR): Configuration Guideline RCE100 Copyright © 2015 by Alcatel-Lucent. All rights reserved Then, configure the parameters corresponding to this VoIP Access:

- ② number of channels allocated ("VoIP-Trunk Ch."):
 - Access_Channels_example = 8
- ③ "Alternative CLIP/COLP Number" :

Access_Alt_CLIP_example = ""

3.3 Hosting System Trunk Group

To enable phone calls over the SIP trunk, it's necessary to have this latter included within one Trunk Group of the system. Two alternative cases (variants) are considered here below.

1) Select the OMC menu External Lines ->List of Trunk Groups and carry out the selections and push-button steps shown in the following picture (i.e. step1 to step 6 depicted by blue digits 1 to 6).

OMC		List of Trun	k Groups			+ [
🍪 Tools		 Index 	🔘 No. 🛛 🔘	Туре 🤅	Name	Modify	y
		1	Cy	clic 🔻	2-	Datai	3
🗄 b Subscribers	Misc	(20		s J J J
🖶 🌊 External Line	es	2	500 Cyc	clic			
- Eist of A	ccesses	3	501 Cyc 502 Cw	slic slic		2b	
List of Tr	runk Groups	5 6 7	Trunk Groups :	Details			×
Traffic C	ounters	8	Index No.	Туре	Name		
		10	1	Cuelie			
		12	linned lin	Cyciic			
	g Call Handling	13 14 15	Phy. Add.	Асс. Туре	Identifier	No of Chan. 5	4 Add
Hardware a	nd Limits						
Metering		Retu					Delete
Selection List		Retu					Delete
Metering Selection List		Ret	©			0	
Selection List	Internal	Retu Public	Private	Ident	Nbr.B-	© Туре	Delete
Belection List	. Internal No.	Rett O Public No.	Private No.	Ident	Nbr.B- Channel	 Туре	Delete
Metering Selection List Objc Phy.Addr. Index ACC 01-009-07	. Internal No.	Public No.	© Private No.	Ident	Nbr.B- Channel 2	© Туре Т0	Delete
Metering Selection List Objc Phy.Addr. Index ACC 01-009-07 ACC 91-0109-07 ACC 91-0109 ACC 91-	Internal No.	Public No.	Private No.	Ident N001 N002 V001	Nbr.B- Channel 2 2 5	Type To To VolP	Name
Metering Selection List Objc Phy.Addr. Index ACC 01-009-07 ACC 95-001-07	Internal No.	Public No.	Private No.	Ident N001 N002 V001	Nbr.B- Channel 2 5	Type To To To VolP	Name
Metering Selection List Obje Phy.Addr. Index ACC 01-009-07 ACC 95-001-0	Internal No.	Public No.	Private No.	Ident N001 N002 V001	Nbr.B· Channel 2 5	Type To To To VolP	Name
Metering Selection List Obje Phy.Addr. Index ACC 01-009-07 ACC 95-001-0	Internal No.	Public No.	Private No.	Ident N001 N002 V001	Nbr.B- Channel 2 5	Type To To To VolP	Name
Metering Selection List Obje Phy.Addr. Index ACC 01-009-07 ACC 95-001-07 ACC G	Internal No.	Public No. 5	Private No.	Ident N001 N002 V001	Nbr.B- Channel 2 5	Type Type T0 T0 VoIP	Name

2) As a configuration variant, at step 2 you can include the SIP trunk access into the OXO's main Trunk Group (i.e. step 2a for index #1) or into one of the secondary Trunk Groups (e.g. step 2b for index #2).

The SIP trunk can be placed freely into one or several Trunk Groups of the system thus permitting to manage a differentiated control of traffic sharing for internal subscribers.

The index number selected at step 2a or 2b is relevant for the further configuration of section 3.4.

3.4 ARS Trunk Groups Lists

To enable voice calls via the ARS system, it's necessary to have ARS Trunk Groups created via the OMC menu <u>Numbering -> Automatic Routing Selection -> Trunk Groups Lists.</u>

In this menu, new lines are created after clicking the mouse right button and selecting function "Add"

List ID	and the second se							+
LIVERD	Index	No.	Char	Provider/Dest	tination	Access Digits	Auth.Code ID	Tone/Pause
1	1			None			None	None
2 🖉			3					
Calanti								
Selection	on List							
0	0	0	0				٢	0
Ubjc	Phy.Addr.	Internal	Public	Private	Ident	NDI.B-	Type	Name
	IIIAGA	110.	110.	110.		Cindrandi	4a	
BDL	2	500					Cyclic	D4h
BDL	3 4	501 502					Cyclic Cyclic	
BDL BDL	5	503 504					Cyclic Cyclic	
BDL	7	505 506					Cyclic	
BDL	9	507					Cyclic	
DUL	10	506					Cyclic	
	5							
	OK	Cancel						
	UK	Cancer						
	2 Selecti Obje BDL BDL BDL BDL BDL BDL BDL BDL BDL BDL	2 Selection List Objc PryAdd. Index DDL 2 BDL 3 BDL 4 BDL 5 BDL 6 BDL 7 BDL 8 BDL 9 BDL 9 BDL 10 5 OK	2 Selection List Objc PhyAdd. Internal Index No. DD1 2 500 BDL 4 502 BDL 5 503 BDL 7 505 BDL 9 507 BDL 9 507 BDL 9 507 BDL 10 508 5 0K Cancel	2 3 Selection List	2 3 Selection List Objc PhyAdd. Internal Public Private Index No. No. No. DDU 2 500 BDL 4 502 BDL 5 503 BDL 6 504 BDL 7 505 BDL 9 507 BDL 9 507 BDL 9 507 BDL 9 Concel	2 3 Selection List Objc PhyAdd. Internal Public Private Ident Index Na. Na. No. DD1 2 500 BD1 4 502 BD1 5 503 BD1 5 503 BD1 6 504 BD1 7 505 BD1 9 507 BD1 9 507 BD1 10 508 5 OK Cancel	2 3 Selection List Objc Phy Add. Internal Public Private Ident Nbc B. Index No. No. No. No. Channel BDL 2 501 BDL 5 503 BDL 5 507 BDL 5	2 3 Selection List Opic PryAdd Internal Public Private Ident Nbz.B- Type Index No. No. No. No. Channel 4a BDL 2 900 Octo Octo 0cto 4a BDL 5 503 Opto Octo 0cto 0cto 4a BDL 5 503 Opto Octo 0cto 0cto 0cto 4a BDL 5 503 Opto Opto 0cto 0cto

Carry out the selections and push-button steps 1 to 5 above. At step 4, you need to select the line index corresponding to the System Trunk Group previously defined at section 3.3 (i.e. action 4a for the Main Trunk Group with index #1, or action 4b for the secondary Trunk Group with index #2).

4 SIP TRUNK SETUP

4.1 Importing the Operator's reference profile (SIP Easy Connect)

For proceeding with the next configuration steps, you should have on your PC the file "FR_OVH_RCE100_SIP_edxx.spf" which contains the SIP Trunk Profile associated to this guide.

0000000

If you don't have the dedicated profile file mentioned here above, please read carefully the particular instructions of Ch.6 Addendum before proceeding.

The general operation and conditions for importing a SIP Trunk profile are detailed in the Technical Bulletin TC1994. The drawing hereafter does just outline and remind the successive steps for achieving the import.

OMC	Import/Expo	rt Data	×	SIP Trunk Profile: Import	23
SI OMC	6				
×	File	FR_OVH_RCE100_SIP_edxx.spf	File 1	Doyou really want to import the selected SIP Trunk profile in Warning: The profile data will override the existing data !	ito your databa
- Broadcast Groups	Data Type	spf SIP Trunk Profile 🔹	Import 2		
Pickup Groups			Export	Desile la ferre stien	
Manager-Secretary Relations			Export	Profile Information	
🗄 🤷 Subscribers Misc			Edit	Profile name OVH	1
🗄 🏀 External Lines					
🕀 😎 Hardware and Limits				3	
🗄 🚳 Metering	Return			OK Cancel	
🗄 🐻 Traffic Sharing & Barring		~			
🗄 🖏 Network Management Contro	OMC	<u> </u>			
🖽 🦥 Voice Over IP					
🗄 <table-of-contents> System Miscellaneous</table-of-contents>	SIP T	runk profile successfully imported !			
🖨 🚍 Import/Export					
- 🔤 Import/Export Data					
Exporting Labels		ок			
🖶 🔩 History & Anomalies					
🗄 🏷 Data Saving & Swapping					

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Reminder: once the Operator profile has been successfully imported, you need to carry on a system reboot (warm reset).

4.2 Complementary Setup

The steps of Ch. 4.2 are needed for completing the OMC configuration not managed by SIP Easy Connect (or the variant data that is specific to the Operator's solution).

4.2.1 ARS-GW Domain Proxy & Registration

Double-click on menu <u>Numbering -> Automatic Routing Selection -> Gateway Parameters</u> ①. A new window "Gateway Parameters List" is displayed that focuses the index entry #1 of the SIP Operator.

омс	ateway	Parameters	>					+ X
□ □-03 Numbering	Gateway	Parameters List	13					
Installation Numbers	Index	Index Label	IP Type	IP Address	Hostname	Domain Name		
	1	OVH	Dynamic					
								Create
Features in Conversation							2	Details
DDI Number Modification Table							4	Detaile
								Delete
End of Dialing Table								
🖃 🗀 Automatic Routing Selection	-							
Gateway Parameters	0	к	Cancel					
SIP Accounts								

Press the button "Details" ② and then select the Domain Proxy tab of the "Gateway Parameters Details" window.

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·····,
Gateway Parameters Details
General Domain Proxy Registration Media DNS Identity Protocol
IP Type Dynamic
IP Address
Hostname
Default Transport Mode UDP
Target Domain Name 1 sin ovh fr (or sintrumk ovh pet)
Lond Domain Name 2 sin out fr (or sintrank out not)
OK Cancel
Fill-in the Domain parameters:
• U "Target Domain Name": • Gwdom_Target_Domain = ** (see note below)
 ② "Local Domain Name":

(*) This variant data is ruled by the Operator and is specific to the solution model delivered. **The current known values are:**

- "sip.ovh.fr" (for solutions "Ligne SIP Entreprise" or "Ligne SIP individuelle") or
- "siptrunk.ovh.net" (for the solution "Ligne SIP Trunk")

Switch to the Registration tab and fill in the parameter Registrar Name respecting the same variant rules.

Gateway Parameters Details	
General Domain Proxy Registration Media DNS Identity Protocol	
Requested	
Registration check for sending requests	 -
Registrar Name 1 sip.ovh.fr (or siptrunk.ovh.net)	
Registrar IP Address	
Port 5060	 -
Expire Time 1800	
	Ĩ
	1 • •
• ① "Registrar Name": ☞ ● ✓ GWreg	Reg_Name = "sip.ovh.fr" (c

When completed, press button "OK" to validate the changes.

4.2.2 ARS Prefixes

ARS Prefixes are used in the system to build up the routing table of external calls. The initial digits dialed by a user are looked-up in the table lines, trying to match an existing prefix/range number. Whenever a match line is found, the call is conveyed thru the specific trunk gateway (GW index) associated to this line.

On OMC, go to menu Numbering -> Automatic Routing Selection -> Automatic Routing Prefixes.



As illustrated in the following picture, you can first insert a route-line covering all type of external calls: use the Add function to create a new line and then, configure the line parameters as indicated.

ļ	Automati	c Routing:	Prefi	kes											⊧ i
į	Activation	Network	Prefix	Ranges	Substit	TrGpList	Called(ISVPN	User com	Metering	Calling	Called/PP	Destination	Gateway Alive S	Gateway Paramete	rs Index
į	Yes	pub		0-9		1	het		Blank	default	default	SIP Gateway	Alive	1 OVH	
į															
ļ		- · - · - ·	- • - •		- · - · - ·	- · - · -		· - · - · - ·	- · - · -		- · - · - ·	- · - · - · -			

Depending on the customer context, you may of course need to refine the routing table and add lines for specific number ranges (e.g. short numbers or emergency numbers). Here below is given a more complete example with four Prefix lines (customized values in area 1 and 2) routing all the required phone numbers:

Automa	ic Routin	g: Prefi	ixes											+
Activation	Network	Prefix	Ranges	Substitute	TrGpList	Called(ISVPN/H	User comment	Meteri	Calling	Called/PP	11	Destination	Gateway Alive St	Gateway Parameters Index
Yes	pub	0	0-9		1	het	Line 1	Blank	default	default	П	SIP	Alive	1.
Yes	emerg				1		Line 2	Blank	default	default		Gateway	Alive	1
Yes	pub	3	0-9	3	1		Line 3	Blank	default	pub shortN		,	Alive	1
Yes	pub	1	0-9	1	1	ł	Line 4	Blank	default	default			Alive	1
i			1			2				3		4		
i							r.,						r	
4														

- Line 1: for standard calls starting with digit 0 (national and international calls)
- Line 2: for external emergency numbers (e.g.112). This line is linked to the system emergency numbers specified in the system flag table "EmergNum" (refer to Technical Bulletin TC80)
- Line 3 and 4: for external short numbers. Depending on the country, the complete list of short numbers will require one or several ARS lines.
 - Line 3: example for France, for short numbers that begin with digit 3 (e.g. 3611, 3900, ...)
 - Line 4: example for France, for short numbers that begin with digit 1 (e.g. 11, 118712, ...)

In area 3, "Calling" and "Called/PP" fields must be set as shown in the example. In area 2 and 4, values must also be respected:

- The Gateway Alive Status is automatically updated by OXO via the "keep alive" control mechanism
- The Index of Gateway field is accessible after the Gateway parameters menu is completed (step 3.5.2)

4.2.3 ARS SIP Accounts

The menu <u>Numbering -> Automatic Routing Selection -> SIP Accounts</u> permits to configure the user credentials delivered by the SIP Operator for authentication.

ОМС	<	SIP A	ccounts			+			
ST OMC	-	Index	Login	Password	Registered Username	Gateway Parameters Index			
Automatic Routing: Pro	efixe	1	003397***2347	****	003397***2347	1 OVH			
Gateway Parameters				2	2				
SIP Accounts			.		3				
• 🛈 "Login" :		G • 9	SIPaccnt_Lo	gin_exa	mple = "003397	/***2347"			
• 2 "Password" :		☞● 9	SIPaccnt_Pa	ssword	= "****"				
• 3 "Registered Use	r.″ :	<pre>SIPaccnt_Reg_Username_example = "003397***2347"</pre>							
the "Cotoway D	• ④ the "Gateway Parameters Index" must point to the relevant gateway (i.e. index 1)								

00	0000	00
I =		_
=	_	
	1	
=	1	

For solutions using several individual lines ("Ligne SIP individuelle"), it is necessary to create one SIP Account line per line (multi-account configuration).

Otherwise, for other situations (one "Ligne SIP individuelle", or one "Ligne SIP Entreprise" or one "Ligne SIP Trunk") a single SIP Account line is sufficient.

4.3 Adjustments (fine tuning)

The configuration steps of Ch. 4.3 refer to particular adjustments you can carry on over the data imported by SIP Easy Connect (data highlighted within the OMC screenshots).

4.3.1 VoIP General Tab

Open the OMC tab via the menu Voice Over IP -> VOIP:Parameters - General



- Adjust the IP Quality of Service of VoIP trunks (RTP flow):

① ☞ ● ✓ VoIPgen_IP_QoS_example = 11000000 IP_PRECEDENCE_6

4.3.2 VoIP SIP Trunk Tab

Open the OMC tab via the menu <u>Voice Over IP -> VOIP:Parameters - SIP Trunk</u>



Adjust the IP Quality of Service of SIP Trunk messages (SIP signaling) :
 ① ☞ ● ✓ VoIPsiptrk_QoS_example = 00000000 DIFFSERV_PHB_BE

4.3.3 VoIP DSP Tab

Open the OMC tab via the menu Voice Over IP -> VOIP:Parameters -

S OMC	VoID: Decomptore			
Subscribers Misc	General Gateway DSP	HCP Fax SIP Trunk SIP F	Phone Codecs	
Hardware and Limits Metering Traffic Sharing & Barring	Law Mode	A-Law		
Voice Over IP Voice Voice Parameters	Echo Cancellation	1		
VoIP: Traffic Counters	Voice Active Detection	2		Detection (MAD) inheren

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- VoIP calls:

4.3.4 ARS-GW Media Tab

Double-click on menu <u>Numbering -> Automatic Routing Selection -> Gateway Parameters</u> ①. From the new window "Gateway Parameters List", select the index entry #1 corresponding to OVH and then press the button "Details". After that, select the Media tab of the "Gateway Parameters Details" window.

	ateway Parameters Deta	
	General Domain Proxy Regi	stration Media DNS Identity Protocol
	Fax	▼
	T38 additional signaling	No Signal 🗸
		Called Identification Tone (CED)
	Codec/Framing	Default 🔹
	Gateway Bandwidth 1	128 kBit/s
	+	
<u>.</u>	OK Ca	incel



Adjust the GW Bandwidth to the customer context:

5 SIP trunk Configuration Abstract

The following tables gather the overall system configuration (the SEC column shows a ' \checkmark ' sign for the parameters whose value is automatically imported when the SIP Easy Connect facility is used).

Table 1										
CONFIG OXO	VALUE	SEC	REMARK							
System_Numbering_Plan										
NP_Instal_Number_example	"97***2347"		Value given as example							
NP_International_Prefix	"00"									
NP_International_Code	"33"									
NP_Intercity_Prefix	"0"									
NP_Intercity_Code_example										
NP_Recall_Prefix	"0"									
NP_System_Alt_CLIP_example										
NP_DDI_Range_example	"2347 2350"		Value given as example							
	External Lines/ VoIP Trunk									
Access_is_Public	True									
Access_Alt_CLIP_example	""									
Access_Channels_example	8		Value given as example							
	ARS_SIP Accounts									
SIPaccnt_Login_example	"003397***2347"		Value masked partially							
SIPaccnt_Password	"****									
SIPaccnt_Reg_Username_example	"003397***2347"		Value masked partially							
ARS_Prefixes										
ARS_Called_Mode	het									
ARS_Destination	SIP Gateway									
	VoIP_General									
VoIPgen_Trunk_Channels_example	8		Value given as example							
VoIPgen_IP_QoS_example	11000000 IP_PRECEDENCE_6	\checkmark	Value given as example							
VoIPgen_Protocol	SIP	\checkmark								
VoIPgen_RTP_Direct	False	\checkmark								
VoIPgen_Trunk_Codec_Passthru	False	\checkmark								
VoIPgen_Phone_Codec_Passthru	False	\checkmark								
VoIPgen_G711_MOH	False	\checkmark								
VoIPgen_RTCP_Attribute	False	\checkmark								
	VoIP_Gateway									
VoIPgw_SIPSourcePort	"5060"	\checkmark								
	VoIP_DSP									
VoIPdsp_DSP_Echo_Cancel	True	\checkmark								
VoIPdsp_DSP_VAD	False	\checkmark								
	VoIP_Fax									
VoIPfax_T38_UDP_Redundancy	"1"	\checkmark								
VoIPfax_T38_Fax_Framing	"0"	\checkmark								
VoIPfax_T38_ECM	False	\checkmark								
	VoIP_SIP Trunk									
VoIPsiptrk_QoS_example	00000000 DIFFSERV_PHB_BE	\checkmark	Value given as example							
VoIPsiptrk_SIP_Timer_T1	1000	\checkmark								
VoIPsiptrk_SIP_Timer_T2	4000	\checkmark								
VoIPsiptrk_SIP_N_Retries	6	\checkmark								
VoIPsiptrk_UdpToTcp	False	\checkmark								
VoIPsiptrk_DNS_Auth	False	\checkmark								

Table 2

CONFIG OXO	VALUE	SEC	REMARK
	ARS_GW_DNS		
GWdns_DNS_Mode	DNSSRV	✓	
GWdns_Prim_DNS	"8.8.8.8"	\checkmark	
GWdns_Sec_DNS	""	\checkmark	
	ARS_GW_Domain Proxy		
GWdom_IP_Type	Dynamic	✓	
GWdom_IP_Address	(N/A)	\checkmark	
GWdom_Def_Transport	UDP	\checkmark	
GWdom_Target_Domain	"" Refer to Ch. 2.1 & Ch. 4.2.1	√*	Must be configured manually in OMC
GWdom_Local_Domain_Name	"" Refer to Ch. 2.1 & Ch. 4.2.1	√ *	Must be configured manually in OMC
GWdom_Realm	""	\checkmark	
GWdom_Remote_SIP_Port	"5060"	\checkmark	
GWdom_Outb_Proxy	""	\checkmark	
	ARS_GW_Media		
GWmedia_Fax_Mode	T38	\checkmark	
GWmedia_T38_Add_Signal	No Signal	\checkmark	
GWmedia_T38_CED_Tone	False	\checkmark	
GWmedia_Codec_Framing	Default	\checkmark	
GWmedia_Bwidth_example	128 kBit/s	\checkmark	Value given as example
	ARS_GW_Registration		
GWreg_Reg_Requested	True	\checkmark	
GWreg_Check_Before_Req	True	\checkmark	
GWreg_Reg_Name	"" Refer to Ch. 2.1 & Ch. 4.2.1	√ *	Must be configured manually in OMC
GWreg_Reg_IP_Address	(N/A)	\checkmark	
GWreg_Reg_Port	(Dynamic)	\checkmark	
GWreg_Reg_Expire_Time	"1800"	\checkmark	
GWreg_Reg_AoR_In_Contact	False	\checkmark	
GWreg_Reg_AoR_In_From	False	\checkmark	
GWreg_Reg_AoR_In_PAI	False	\checkmark	
GWreg_Reg_AoR_In_PPI	False	\checkmark	
GWreg_Reg_AoR_In_Rsv1	False	\checkmark	
GWreg_Reg_AoR_In_Rsv2	False	\checkmark	
GWreg_Reg_AoR_In_Rsv3	False	\checkmark	
GWreg_Reg_AoR_In_Rsv4	False	\checkmark	
	ARS_GW_Identity		
GWident_RFC3325	True	\checkmark	
GWident_HistInfo_Enabled	True	\checkmark	
GWident_Inc_CLI_Headers	PAI PPI From Rsv-1 Rsv-2 Rsv-3 Rsv-4 Rsv-5	\checkmark	
GWident_Out_CLI_PPI_Used	False	\checkmark	
GWident_Out_CLI_PAI_Used	True	\checkmark	
GWident_Out_COLP_Headers	PAI PPI Contact To Rsv-1 Rsv-2 Rsv-3 Rsv-4	\checkmark	
	ARS_GW_Protocol		
GWprot_SessTimer_Time	"720"	\checkmark	
GWprot_PEM_Enabled	False	\checkmark	
GWprot_UPDATE_Enabled	True	\checkmark	
GWprot_SNAT_Enabled	False	\checkmark	
GWprot_PRACK_Enabled	True	\checkmark	
GWprot_GWalive_Prot	(N/A)	\checkmark	
GWprot_GWalive_Timer	(N/A)	\checkmark	

CONFIG OXO	VALUE	SEC	REMARK
	ARS-SIP Public Numbering		
SIPnum_Out_Calling_Format	International	\checkmark	
SIPnum_Out_Calling_Prefix	111	\checkmark	
SIPnum_Out_Called_Format	International	\checkmark	
SIPnum_Out_Called_Prefix		\checkmark	
SIPnum_Out_Called_Short_Prefix	""	\checkmark	
SIPnum Inc Calling Format	National	\checkmark	
SIPnum Inc Calling Prefix	111	\checkmark	
SIPnum Inc Called Format	Canonical/International	\checkmark	
SIPnum Inc Called Prefix	111	\checkmark	
SIPnum Alt CLIP example			
	VoIP Codecs		
VoIPcodec Def CodecEraming	30	√	
VoIPcodec_Def_Codecl_ist	G711 a G711 µ G722 G722 2	\checkmark	
VoIPcodec_DTME_Payload	"106"	\checkmark	
VoIPcodec_DTAIL_Tdylodd	"117"	\checkmark	
Voli codec_G/22.2_i ayload	VoIR Topology		
VoIPtono PubIP example	*01P_10p0l0gy	1	SIR-NAT function not used
VolPtopo_Fubir_example	"5060"	\checkmark	SIP NAT function not used
VolPtopo_SIPport_example	3000 "22000 222EE"		SIP-NAT function not used
VolPtopo_KTP_Kange_example	32000-32233		SIP-NAT function not used
VoIPtopo_138_Range_example	"6666-6761"	<u> </u>	SIP-INAT function not used
Miss OLL Fit Diversion	Misc_Feature Design		
Misc_CLI_Ext_Diversion	True		
Misc_CLI_is_Diverted_Party	Faise		
Flag_VOIPnwaddr_Line1		•	
Flag_VOIPnwaddr_Line2		×	
Flag_VOIPnwaddr_Line3		v	
Flag_VOIPnwaddr_Line4		v	
Flag_VOIPnwaddr_Line5		v	
Flag_VOIPnwaddr_Line6	"00 00 00 00 00 00 00 00"	✓	
Flag_VOIPnwaddr_Line7	"00 00 00 00 00 00 00 00"	✓.	
Flag_VOIPnwaddr_Line8	"00 00 00 00 00 00 00 00"	~	
Flag_VOIPnwaddr_Line9	"00 00 00 00 01 00 00 00"	~	
Flag_VOIPnwaddr_Line10	"00 00 00 00 00 00 00 00"	✓	
Flag_VipPuNuA	"00"	~	
Flag_ExtNuFoVoi	"22"	\checkmark	
Flag_MultAnsRei	"00"	\checkmark	
Flag_SimulIpAlt	"00"	\checkmark	
Flag_PrefCodec	"00 00"	\checkmark	
Flag_PrefFramin	"1E"	\checkmark	
Flag_FaxPasCd	"01 FF"	\checkmark	
Flag_SIPInDspNm	"01"	\checkmark	
Flag_SIPOgDspNm	"01"	\checkmark	
Flag_INVwSDPtrk	"00"	\checkmark	
Flag_SIPdtmfInB	"00"	\checkmark	
Flag_SuprAlerTo	"00"	\checkmark	
		1	
	1	1	

Table 3

6 ADDENDUM: CONFIGURATION WITHOUT SIP EASY CONNECT

If you can't import the dedicated profile file of the Operator, you will need then to configure all SIP data manually. Such operation is not recommended as SIP Easy Connect makes the process easier and safer. As an additional constraint, **you must follow strictly and carefully the stages 1 to 5 hereafter** which supersede the chapter organization of this guide:

- Stage 1): complete normally all steps of the guide until reaching Ch.4 (i.e. Ch.2 & Ch.3)
- **Stage 2**): create manually a new SIP Gateway entry as illustrated in the following picture:
 - Double-click on menu <u>Numbering -> Automatic Routing Selection -> Gateway Parameters</u> ①. A new window "Gateway Parameters List" is displayed.
 - Press the button "Create" ②. A second window "Gateway Parameters Details" is displayed.
 - From the General tab, select the Index value 1 3 and the SIP Numbers Format Index 1 4

C Collinian Table	Gateway Parameters	+ Σ
Splitting Table	Baleway Parameters List	
Automatic Pourting Selection		
Automatic Routing Selection	Index Index Label IP Type IP Address Hostname Domain Name	
Automatic Routing: Prefixes		2
Gateway Parameters		Create
SIP Accounts		0.00.0
SIP Public Numbering	Gateway Parameters Details	Details
- IIII Trunk Groups Lists		
Hours	General Domain Proxy Registration Media DNS Identity Protocol	Delete
Day Groups		
Providers/Destinations		
- Authorization Codes	index ' 3	
- Tone/Pause-MF	Index Label OVH 5	Copy
ARS Miscellaneous		
Collective Speed Dialing	SIP Numbers Format Index 1 • 4	Paste
🤍 Directory		
Subscribers/Basestations List		
- 📁 Voice Processing		
···· 🕐 Time Ranges		
Attendant Groups		
		-
Broadcast Groups		
Pickup Groups	6	
Manager-Secretary Relations		
- de Subscribers Misc	OK Cancel	
😤 External Lines		

• Optionally, you can type an id name (e.g. "OVH") as Index Label (5)

- **Stage 3):** for completing the creation of the new Gateway, OMC will force you to configure previously the following Gateway Parameters tabs:

- DNS
- Domain Proxy
- Media

These tabs are not specifically detailed in the doc, nor illustrated with OMC screenshots. **You must then refer to the configuration abstract of Ch.5** and apply on OMC the exact value of parameters found in the respective sections "**ARS_GW_DNS**", "**ARS_GW_Domain Proxy**" and "**ARS_GW_Media**" of **Table 2**. When achieved, terminate the creation of the gateway by pressing OK button (ref. 6) in picture above).

- **Stage 4):** proceed with all the operation steps of Ch.4.2 and Ch.4.3: **although it will not be mentioned** there, you will need to tune up the whole configuration values visible in the dedicated OMC screenshots (I.E the parameters highlighted in yellow plus those not highlighted).

Stage 5): to finalize the overall SIP configuration, you must revise carefully all the specific parameters which are normally incumbent to SIP Easy Connect: i.e. the OMC screens for "VoIP parameters", ARS ("ARS Prefixes", "Gateway Parameters", "SIP Public Numbering") and "Misc. Memory Read/Write". To do it, use the abstract of Ch.5 and refer to columns "VALUE" and "SEC" of the tables.

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Please connect to our <u>eService Request</u> application.

Before submitting a Service Request, make sure that:

- In case a Third-Party application is involved, that application has been certified via the AAPP
- You have read through the Release Notes which lists new features available, system requirements, restrictions etc. available in the <u>Technical Documentation Library</u>
- You have read through the Troubleshooting Guides and Technical Bulletins relative to this subject available in the <u>Technical Documentation Library</u>
- You have read through the self-service information on commonly asked support questions, known issues and workarounds available in the <u>Technical Knowledge Center</u>

- END OF DOCUMENT -