Technical Bulletin | Alcatel-Lucent OmniPCX Office

TC2031 ed.01 **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

Edition 01: March 06, 2015 first RCE100 edition released for GA publication

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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 ☞● . 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☞●✓ . Example:

☞●✓ **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: ☞● **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:☞● **NP_Instal_Number_example = "97***2347"**

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 3rd 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	**OK**	**OK**	**NT**	**NT**	**NT**	**NT**	**NT**	**NT**
USER Basic Features								
Outbound Basic Call	**OK**	**OK**						
Inbound Basic Call	**OK**	**OK**						
Inbound Call to DDI	**OK**	**OK**						
Call Release	**OK**	**OK**						
Call Hold & Music	**OK**	**OK**						
Emission of DTMF	**OK**	**OK**						
Reception of DTMF	**OK**	**OK**						
Internal Call Forward	**WR**	**WR**						
Internal Call Transfer	**OK**	**OK**						
CLIP Inbound	**OK**	**OK**						
CLIP Outbound	**OK**	**OK**						
Emergency Calls	**OK**	**OK**						
USER Extended Features								
External Call Forward	**WR**	**WR**						
External Call Transfer	**OK**	**OK**						
COLP	**WR**	**WR**						
Dynamic Call Routing	**WR**	**WR**						
Conference with 2 Ext.	**OK**	**OK**						
Busy State	**OK**	**OK**						
General Preannouncement	**OK**	**OK**						
CLIR In & Outbound	**OK**	**OK**						

SYSTEM Global Features	**Supported**	
Outbound Fax T38	**WR**	
Inbound Fax T38	**WR**	
Outbound Fax G711	**NT**	**Tested in T38**
Inbound Fax G711	**NT**	**Tested in T38**

1.5.2 Restrictions

External Call Forward, Internal Call Forward and Dynamic Call Routing → Issue with COLP.
Inbound & Outbound Fax T38 → 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.

OVH SIP TRUNK PARAMETERS

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

(*) 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")

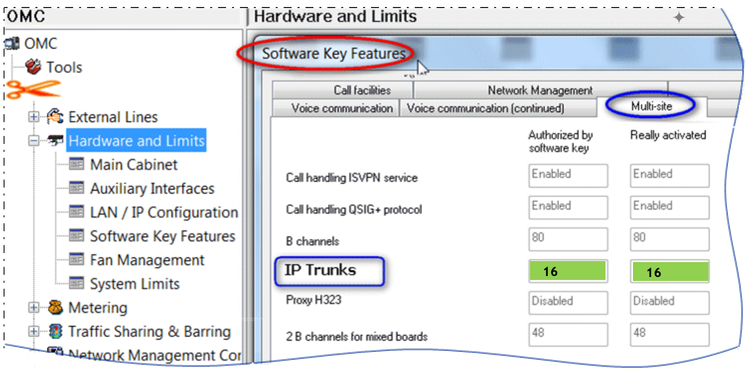
ALE may not be aware of changes made by the Operator. In case of set up issue and doubt concerning the SIP Trunk parameters, please contact the Operator directly.

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 Options -> Language.

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.

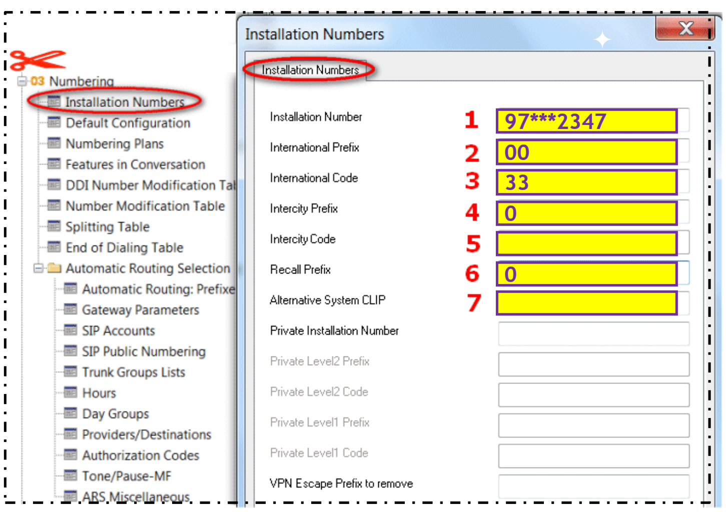


The highlighted values in green result of the SW license and are only readable.

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.

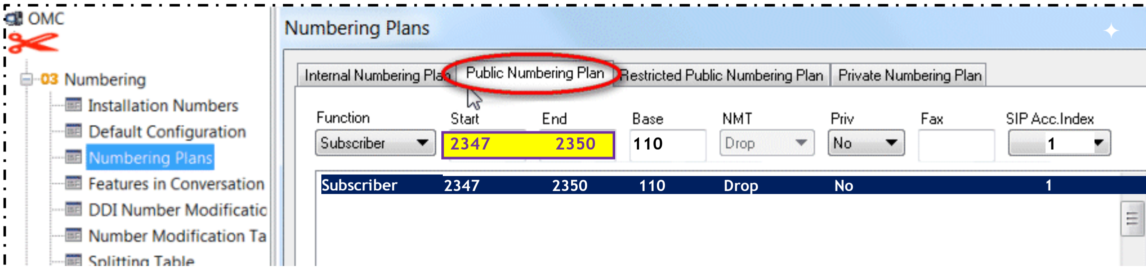


- Check/ edit the corresponding numbers as illustrated above:

- ① Install. Number : ☞● **NP_Instal_Number_example = "97***2347"**
- ② International Prefix : ☞● **NP_International_Prefix = "00"**
- ③ International Code : ☞● **NP_International_Code = "33"**
- ④ Intercity Prefix : ☞● **NP_Intercity_Prefix = "0"**
- ⑤ Intercity Code : ☞● **NP_Intercity_Code_example = ""**
- ⑥ Recall Prefix : ☞● **NP_Recall_Prefix = "0"**
- ⑦ 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: Numbering -> Numbering Plans – Public Numbering Plan.



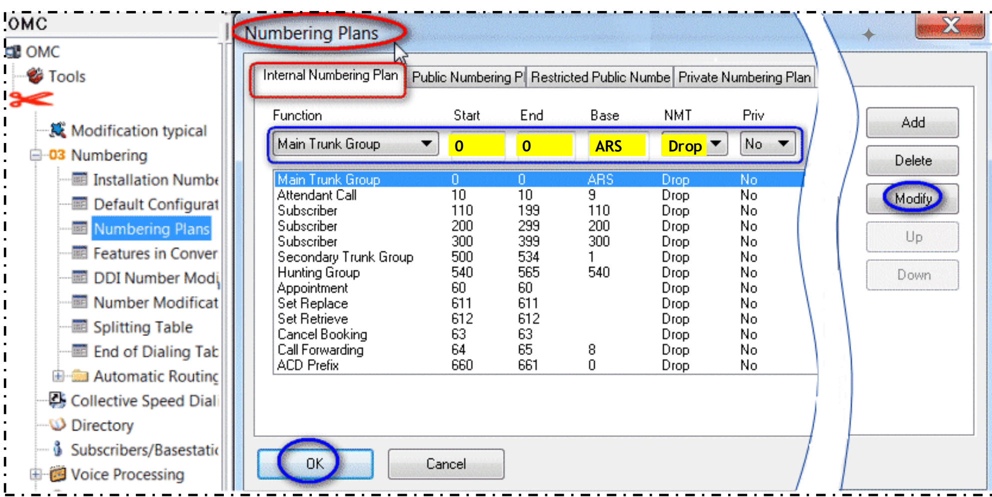
- Check/ edit the configuration for "Public Numbering Plan":

- ① DDI range : ☞● **NP_DDI_Range_example = "2347 2350"**

In conjunction with the configuration of section 2.4.1, this basic example allocates the DDI range "2347 2350" (i.e. 097***2347 - 097***2350) to the range of extensions "110 - 113".

2.4.3 Internal Numbering Plan

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



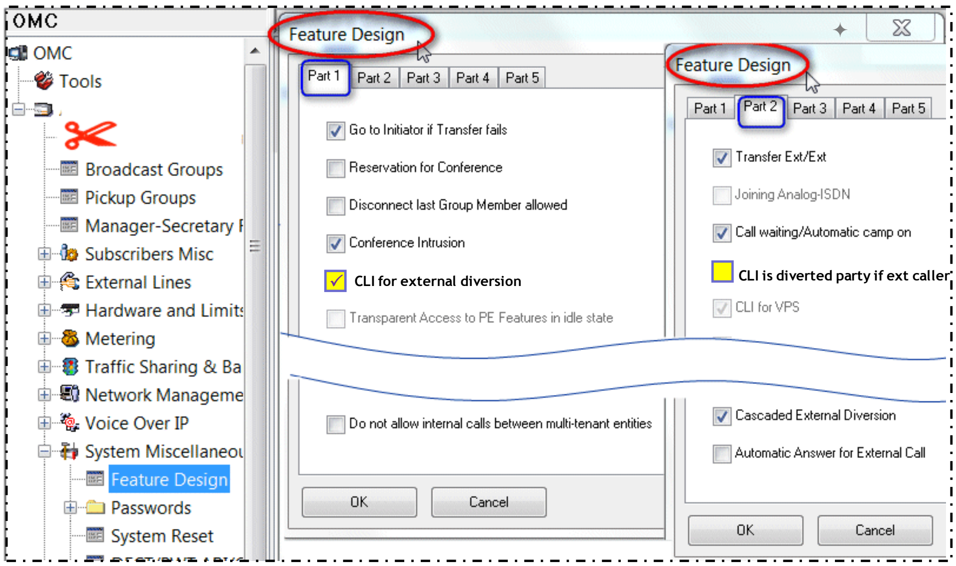
This example defines access to the internal ARS table for user dialing starting with digit 0. The "Drop" attribute indicates that the number dialed must be analyzed in the ARS Prefix table after dropping the initial digit 0.

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.

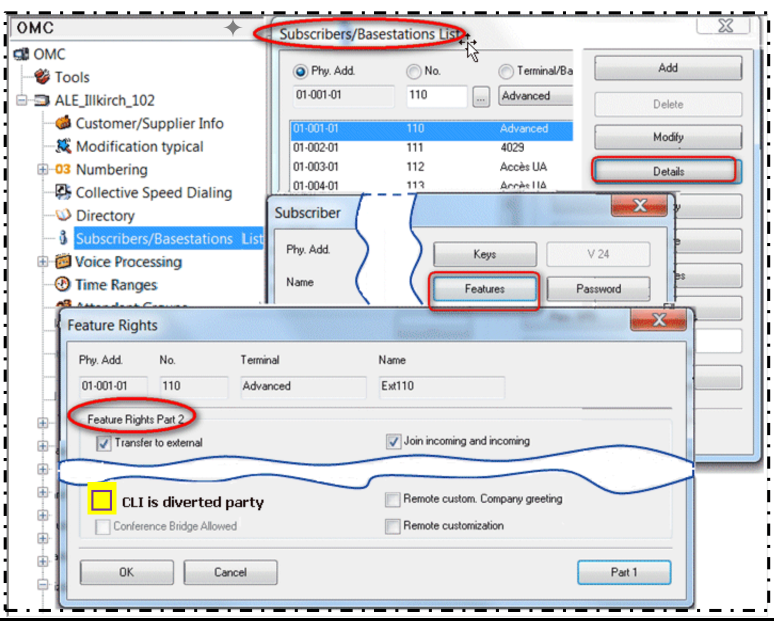
From the tabs "Part 1" and "Part 2" of menu System Misc -> Feature Design, verify the parameters:

- "CLI for external diversion": ☞● **Misc_CLI_Ext_Diversion = True**
- "CLI is diverted party if ext...": ☞● **Misc_CLI_is_Diverted_Party = False**



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**



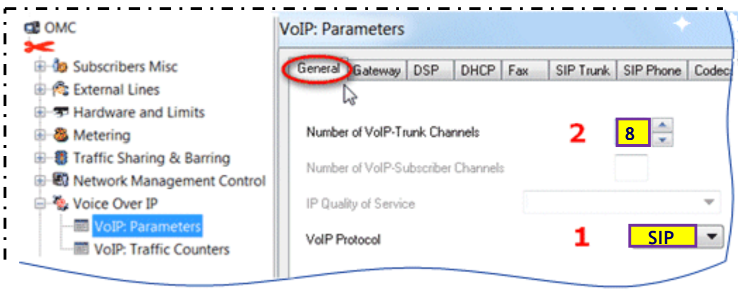
2.6 Traffic Sharing and Barring (reminder)

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 VoIP:Parameters/General and control/ adapt the following parameters that are necessary for creating the SIP trunk in the system:



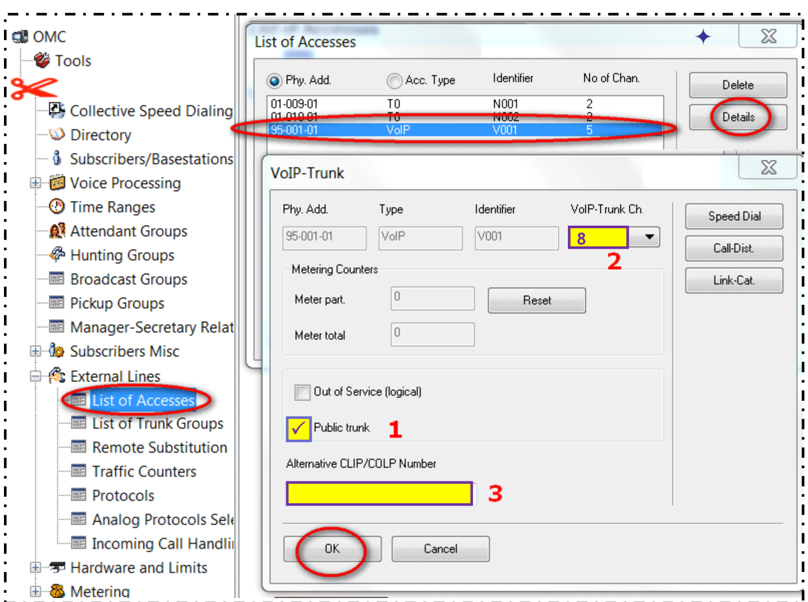
Number of Channels must be greater than zero and adapted to the site traffic (defining max number of simultaneous calls on the VoIP trunk)

- ① VoIP Protocol : ☞● **VoIPgen_Protocol = SIP**
- ② Number of VoIP-Trunk Channels :

  ☞● **VoIPgen_Trunk_Channels_example = 8**

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.



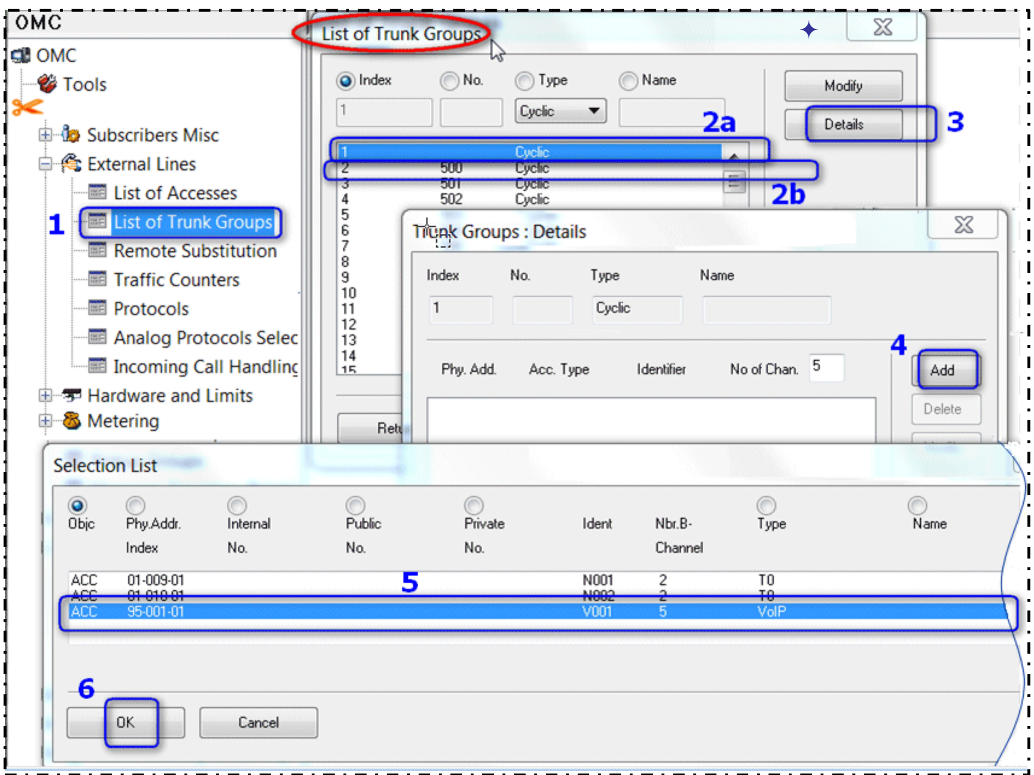
Then, configure the parameters corresponding to this VoIP Access:

- ① "Public trunk" option : ☞● **Access_is_Public = True**
- 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).



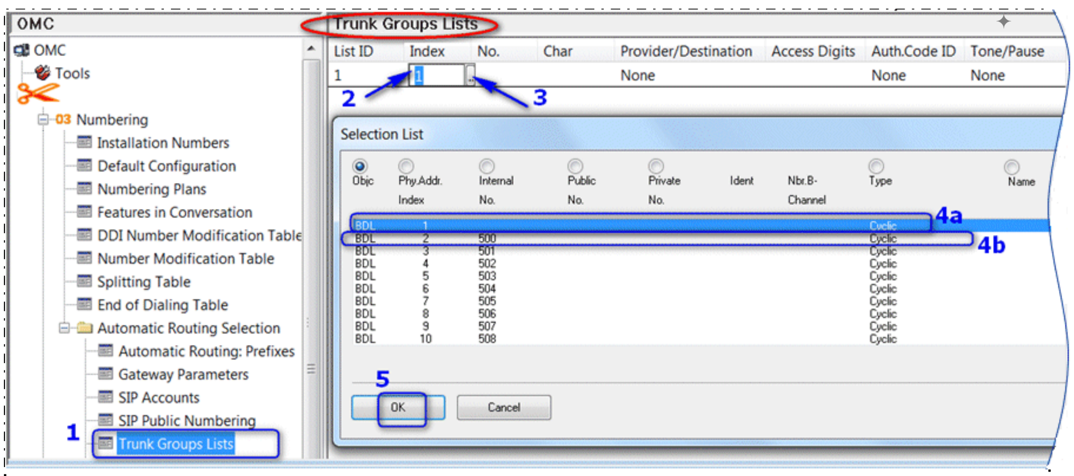
2) As a configuration variant, at step 2 you can include the SIP trunk access into the OXO's main Trunk Group (i.e. step 2a for index #1) or into one of the secondary Trunk Groups (e.g. step 2b for index #2).

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 Numbering -> Automatic Routing Selection -> Trunk Groups Lists.

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



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).

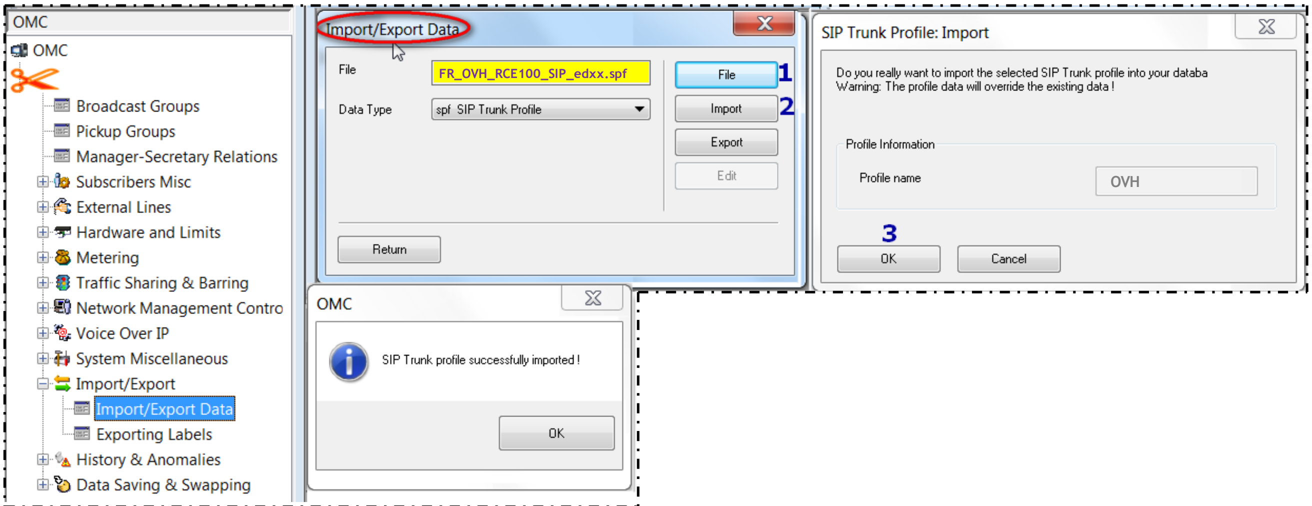
4 SIP TRUNK SETUP

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.

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.



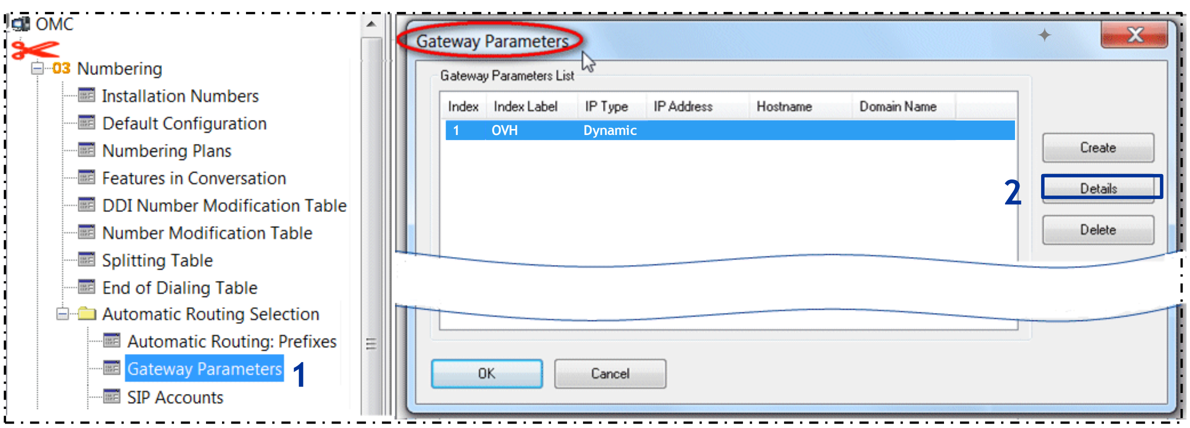
Reminder: once the Operator profile has been successfully imported, you need to carry on a system reboot (warm reset).

4.2 Complementary Setup

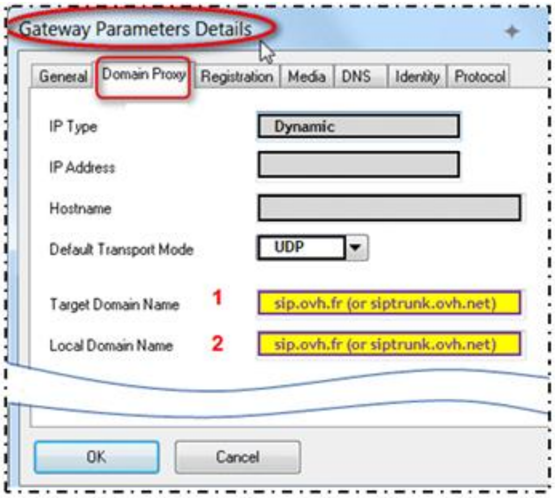
The steps of Ch. 4.2 are needed for completing the OMC configuration not managed by SIP Easy Connect (or the variant data that is specific to the Operator's solution).

4.2.1 ARS-GW Domain Proxy & Registration

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



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

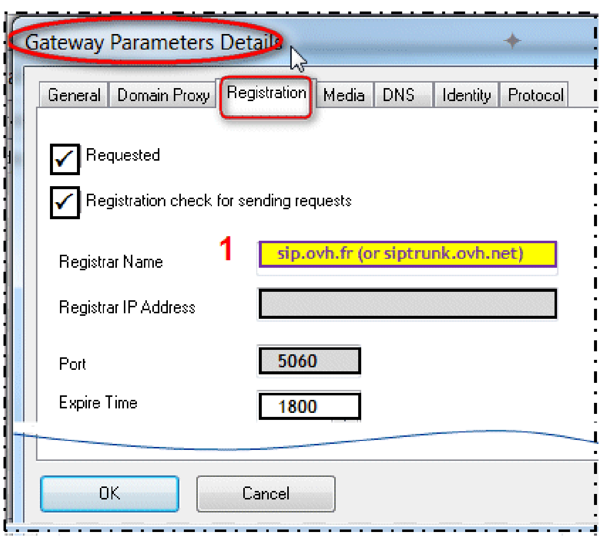


Fill-in the Domain parameters:

- ① "Target Domain Name": ☞●✓ **GWdom_Target_Domain = * (see note below)**
- ② "Local Domain Name" : ☞● ✓**GWdom_Local_Domain_Name = * (see note below)**

(*) 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.



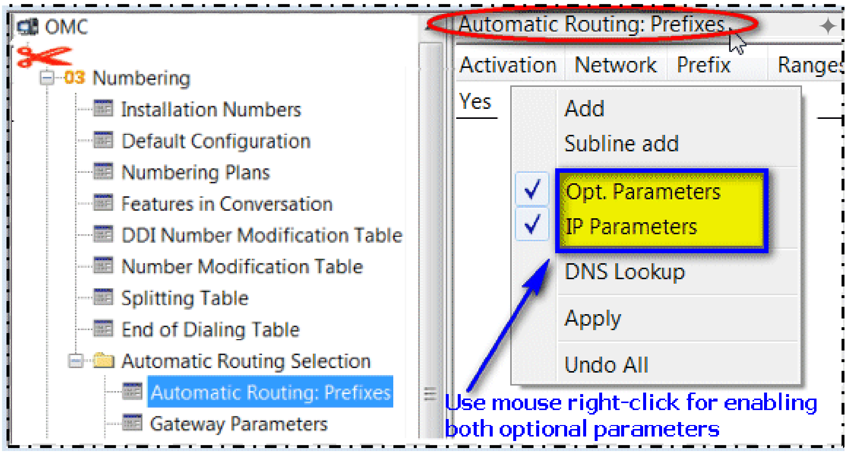
- ① "Registrar Name": ☞●✓ **GWreg_Reg_Name = "sip.ovh.fr" (or "siptrunk.ovh.net")**

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

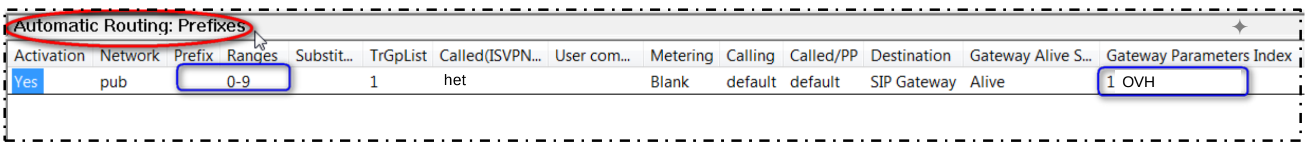
4.2.2 ARS Prefixes

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.

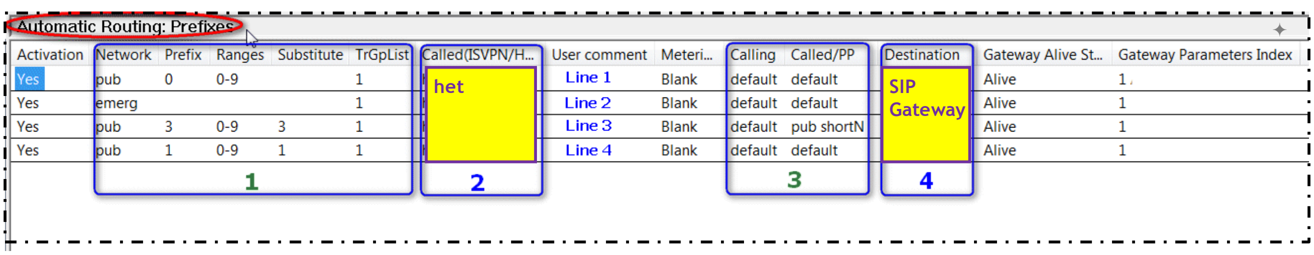


As illustrated in the following picture, you can first insert a route-line covering all type of external calls: use the Add function to create a new line and then, configure the line parameters as indicated.



Activation	Network	Prefix	Ranges	Substit...	TrGpList	Called(ISVPN...	User com...	Metering	Calling	Called/PP	Destination	Gateway Alive S...	Gateway Parameters Index
Yes	pub		0-9		1	het		Blank	default	default	SIP Gateway	Alive	1 OVH

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:



Activation	Network	Prefix	Ranges	Substitute	TrGpList	Called(ISVPN/H...	User comment	Meteri...	Calling	Called/PP	Destination	Gateway Alive St...	Gateway Parameters Index
Yes	pub	0	0-9		1	het	Line 1	Blank	default	default	SIP Gateway	Alive	1
Yes	emerg				1		Line 2	Blank	default	default		Alive	1
Yes	pub	3	0-9	3	1		Line 3	Blank	default	pub shortN		Alive	1
Yes	pub	1	0-9	1	1		Line 4	Blank	default	default		Alive	1

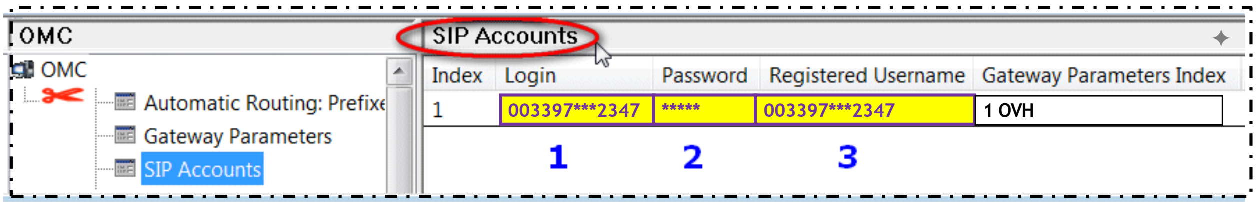
- Line 1: for standard calls starting with digit 0 (national and international calls)
- Line 2: for external emergency numbers (e.g.112). This line is linked to the system emergency numbers specified in the system flag table "EmergNum" (refer to Technical Bulletin TC80)
- Line 3 and 4: for external short 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:

- ② "Called (ISVPN/..)": ☞● **ARS_Called_Mode = het**
- ④ "Destination" : ☞● **ARS_Destination = SIP Gateway**
- 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 Numbering -> Automatic Routing Selection -> SIP Accounts permits to configure the user credentials delivered by the SIP Operator for authentication.



- ① "Login" : ☞● **SIPaccnt_Login_example = "003397***2347"**
- ② "Password" : ☞● **SIPaccnt_Password = "*****"**
- ③ "Registered User." : ☞● **SIPaccnt_Reg_Username_example = "003397***2347"**
- ④ the "Gateway Parameters Index" must point to the relevant gateway (i.e. index 1)

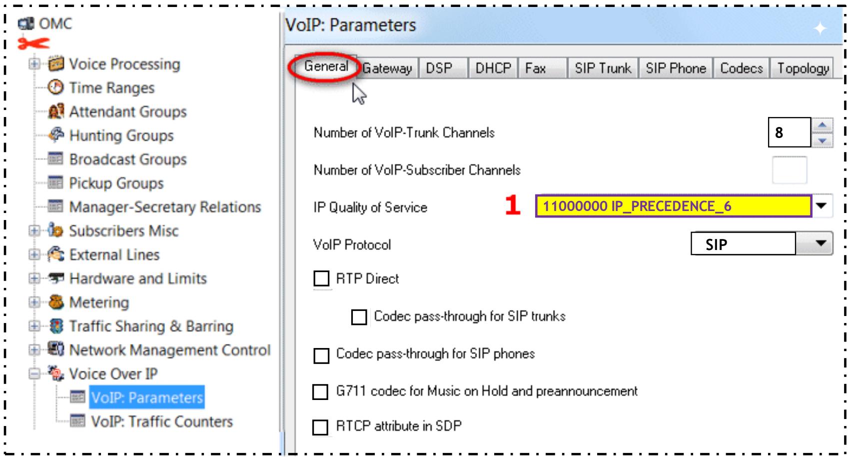
For solutions using several individual lines ("Ligne SIP individuelle"), it is necessary to create one SIP Account line per line (multi-account configuration).
Otherwise, for other situations (one "Ligne SIP individuelle", or one "Ligne SIP Entreprise" or one "Ligne SIP Trunk") a single SIP Account line 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

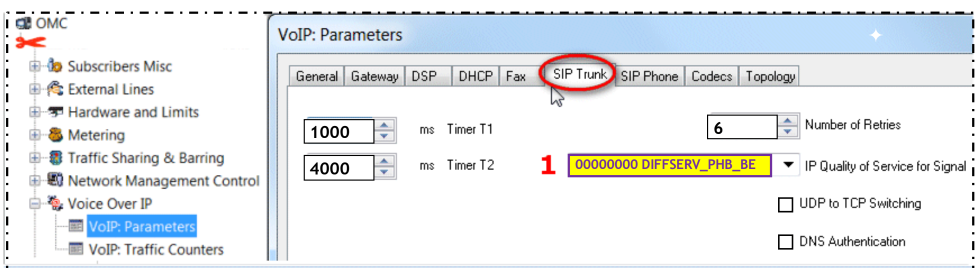


- Adjust the IP Quality of Service of VoIP trunks (RTP flow):

① ☞●✓ **VoIPgen_IP_QoS_example = 11000000 IP_PRECEDENCE_6**

4.3.2 VoIP SIP Trunk Tab

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

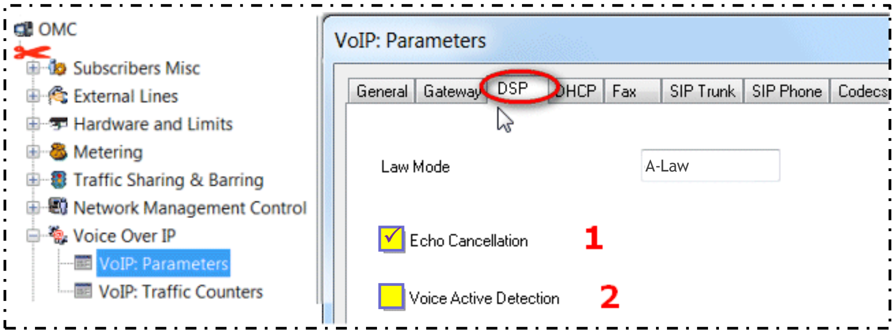


- Adjust the IP Quality of Service of SIP Trunk messages (SIP signaling) :

① ☞●✓ **VoIPsiptrk_QoS_example = 00000000 DIFFSERV_PHB_BE**

4.3.3 VoIP DSP Tab

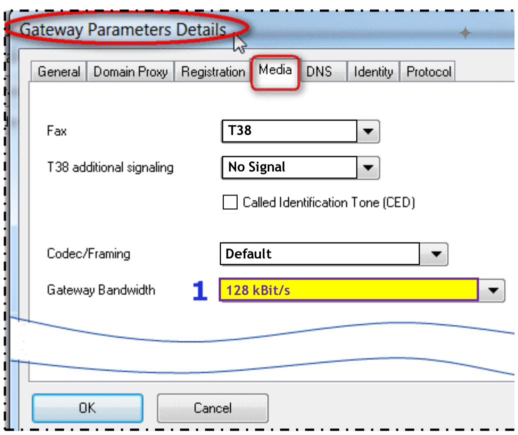
Open the OMC tab via the menu Voice Over IP -> VOIP:Parameters -



- Amend, if necessary, the control of Echo Cancellation and of Voice Active Detection (VAD) inherent to VoIP calls:
    - ① "Echo Cancellation" : ☞●✓ **VoIPdsp_DSP_Echo_Cancel = True**
    - ② "Voice Active Detection" : ☞●✓ **VoIPdsp_DSP_VAD = False**

4.3.4 ARS-GW Media Tab

Double-click on menu Numbering -> Automatic Routing Selection -> Gateway Parameters 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.



The configuration of Fax parameters shown in this picture is only relevant if Fax calls are supported for this Operator (see Ch.1.5.1)

Adjust the GW Bandwidth to the customer context:

① ☞●✓ **GWmedia_Bwidth_example = 128 kBit/s**

5 SIP trunk Configuration Abstract

The following tables gather the overall system configuration (the SEC column shows a '✓' 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	✓	Value given as example
VoIPgen_Protocol	SIP	✓	
VoIPgen_RTP_Direct	False	✓	
VoIPgen_Trunk_Codec_Passthru	False	✓	
VoIPgen_Phone_Codec_Passthru	False	✓	
VoIPgen_G711_MOH	False	✓	
VoIPgen_RTCP_Attribute	False	✓	
VoIP_Gateway			
VoIPgw_SIPSourcePort	"5060"	✓	
VoIP_DSP			
VoIPdsp_DSP_Echo_Cancel	True	✓	
VoIPdsp_DSP_VAD	False	✓	
VoIP_Fax			
VoIPfax_T38_UDP_Redundancy	"1"	✓	
VoIPfax_T38_Fax_Framing	"0"	✓	
VoIPfax_T38_ECM	False	✓	
VoIP_SIP Trunk			
VoIPsiptrk_QoS_example	00000000 DIFFSERV_PHB_BE	✓	Value given as example
VoIPsiptrk_SIP_Timer_T1	1000	✓	
VoIPsiptrk_SIP_Timer_T2	4000	✓	
VoIPsiptrk_SIP_N_Retries	6	✓	
VoIPsiptrk_UdpToTcp	False	✓	
VoIPsiptrk_DNS_Auth	False	✓	

Table 2

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

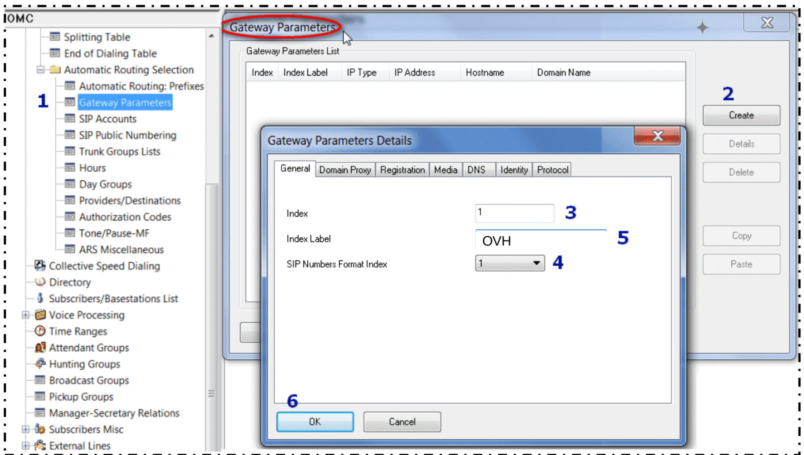
Table 3

CONFIG OXO	VALUE	SEC	REMARK
	ARS-SIP Public Numbering		
SIPnum_Out_Calling_Format	International	✓	
SIPnum_Out_Calling_Prefix	""	✓	
SIPnum_Out_Called_Format	International	✓	
SIPnum_Out_Called_Prefix	""	✓	
SIPnum_Out_Called_Short_Prefix	""	✓	
SIPnum_Inc_Calling_Format	National	✓	
SIPnum_Inc_Calling_Prefix	""	✓	
SIPnum_Inc_Called_Format	Canonical/International	✓	
SIPnum_Inc_Called_Prefix	""	✓	
SIPnum_Alt_CLIP_example	""		
	VoIP_Codecs		
VoIPcodec_Def_CodecFraming	30	✓	
VoIPcodec_Def_CodecList	G711.a G711.µ G722 G722.2	✓	
VoIPcodec_DTMF_Payload	"106"	✓	
VoIPcodec_G722.2_Payload	"117"	✓	
	VoIP_Topology		
VoIPtopo_PubIP_example	""		SIP-NAT function not used
VoIPtopo_SIPport_example	"5060"	✓	SIP-NAT function not used
VoIPtopo_RTP_Range_example	"32000-32255"	✓	SIP-NAT function not used
VoIPtopo_T38_Range_example	"6666-6761"	✓	SIP-NAT function not used
	Misc_Feature Design		
Misc_CLI_Ext_Diversion	True		
Misc_CLI_is_Diverted_Party	False		
	Misc_Memory Read/Write		
Flag_VOIPnwaddr_Line1	"00 00 00 00 00 00 00 00"	✓	
Flag_VOIPnwaddr_Line2	"00 00 00 00 00 00 00 00"	✓	
Flag_VOIPnwaddr_Line3	"00 00 00 00 00 00 00 00"	✓	
Flag_VOIPnwaddr_Line4	"00 00 00 00 00 00 00 00"	✓	
Flag_VOIPnwaddr_Line5	"00 00 00 00 00 00 00 00"	✓	
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"	✓	
Flag_MultAnsRei	"00"	✓	
Flag_SimulIpAlt	"00"	✓	
Flag_PrefCodec	"00 00"	✓	
Flag_PrefFramin	"1E"	✓	
Flag_FaxPasCd	"01 FF"	✓	
Flag_SIPInDspNm	"01"	✓	
Flag_SIPOgDspNm	"01"	✓	
Flag_INVwSDPtrk	"00"	✓	
Flag_SIPdtmfInB	"00"	✓	
Flag_SuprAlerTo	"00"	✓	

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 Numbering -> Automatic Routing Selection -> Gateway Parameters 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
  - 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. ⑥in 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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- You have read through the self-service information on commonly asked support questions, known issues and workarounds available in the Technical Knowledge Center

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