SIP Trunk Solution htp (DE): Configuration Guideline ONE050

This document details how to set up an OXO Connect R50 for enabling a public SIP trunk of the Operator htp (for HTP's commercial offer SIP-Trunk Smart) in DE target.

Revision History

Edition 01: April 29th, 2022	R5.0 Edition

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1 General

This document describes the from-scratch configuration of OXO ONE050 in the context of a SIP trunk solution connected to the public Operator **htp**. The setup is based on the OMC service "SIP Easy Connect" which permits to import a SIP Trunk Profile and then achieve the IPBX configuration in a simplified way.

Warning

For an easier OMC configuration and optimized usage of this guide, you should have at your disposal the reference SIP Trunk Profile delivered by ALE. This reference guide relies on the SIP Trunk Profile " DE_htp_SIP-TrunkSmart_ONE050_SIP_edxx.spf" published by ALE on its Web Portal. Full-manual configuration without using this profile is not recommended. Such operation is not straightforward and is just briefly depicted herein at the Ch.5 addendum.

Note

The bulletin TC1994 explains the overall usage of SIP Trunk profiles in OMC and details how to retrieve and import the ALE profile of an approved Operator.

1.1 References

ALE International 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 engineers who are familiar with OMC and with the basic setup of the IPBX. For simplification reasons, some well-known configurations as those for IP-LAN or "Traffic Sharing and Barring" are just reminded without any details

Warning

In complement to the present guide, the installation must take into account the system security recommendations found in the bulletin TC1143.

The presentation of OMC menus and screenshots corresponds to the selection of the English language in the tool. Every configuration parameter has a specific name which is derived from its menu location in OMC. 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)

☞● **GWmedia_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:

☞●✓ **GWdns_Prim_DNS = "Address of the dns"**

Note	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 for site-dependent values needing to be customized. Example: ☞● **Access_Channels_example = 2** As they are taken from a real customer site, those _example values carrying private data have been masked or partially masked with asterisks (i.e. user logins and passwords, public phone numbers...). Example:☞● **SIPaccnt_Reg_Username_example = " +************"**

1.3 Scope of ALE 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).

Warning	**The support level ensured by ALE for the present solution (i.e. LA or GA) may vary in time and must be checked from the last 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	OXO Connect	ONEDE050/037.001
OMC Management Application	Alcatel-Lucent OMC	OMC50.0/22.1a

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 by the present solution. For each item, the status is given in the column "Support": "OK" (for full support), or "WR" (support with restriction), or "NOK" (for Not OK or Not Applicable), or "NT" (for Not Tested).

Note	For any doubt concerning the tables hereafter, or, if you want to contribute to the validation of items that are not yet tested, you can contact us by e-mail: sip-for-smb@al-enterprise.com

htp TopoC - Hosted NAT without Direct RTP

	40x8 80x8 80X8s	40x9 80x9 Z DECT	IP DECT DAP's	Rainbow	Rainbow with Webrtc	8001 8008CE	OTCV	OTCV with companion
SETS Supported	**OK**	**OK**						
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	**NT**	
Inbound Fax T38	**NT**	
Outbound Fax G711	**WR**	*Not tested in all the requested call scenarios*
Inbound Fax G711	**WR**	*Not tested in all the requested call scenarios*

1.5.2 Restrictions

Internal Call Forward, External Call Forward, Dynamic Call Routing and COLP → Issue with COLP feature. The SIP provider doesn´t take into account the final speaker's number.
Inband & Outband Fax→ Not tested in all the required test call scenarios.

2 System General Info and Basic Setup

2.1 Pre-required information

The table below gathers the specific SIP settings delivered by the Operator (empty values correspond to not relevant or not mandatory parameters). This data is necessary for completing the OXO configuration.

htp SIP TRUNK PARAMETERS

Data Type	Parameter role	Name in doc	Value
Provider-specific	OP Gateway IP@	GWdom_IP_Address	(N/A)
	OP GW Domain name	GWdom_Target_Domain	siptrunk.htp.net
	Outbound Proxy	GWdom_Outb_Proxy	siptrunk.htp.net
	DNS IP@ Prim	GWdns_Prim_DNS	8.8.8.8
	DNS IP@ Sec	GWdns_Sec_DNS	
	Registrar IP address	GWreg_Reg_IP_Address	(N/A)
	Registrar name	GWreg_Reg_Name	siptrunk.htp.net
	SIP Realm	GWdom_Realm	
	Local Domain name	GWdom_Local_domain_Name	
Site specific (example values)	Installation Number	NP_Instal_Number_example	3*****
	Instal. Alternative CLI	SIPnum_Alt_CLIP_example	
	Public DDI range	NP_DDI_Range_example	11 19
	Registered Username	SIPaccnt_Reg_Username_example	+************
	Authentication Login	SIPaccnt_Login_example	+************
	Password	SIPaccnt_Password	*****

Note This data is fixed by the SIP provider and may vary upon the topology model or other criterias proper to the Operator. For the case of "Topology A" model (i.e. direct LAN or VPN connection on the Operator's network), the site data delivered by the Operator may also include the config. values for OXO's LAN.

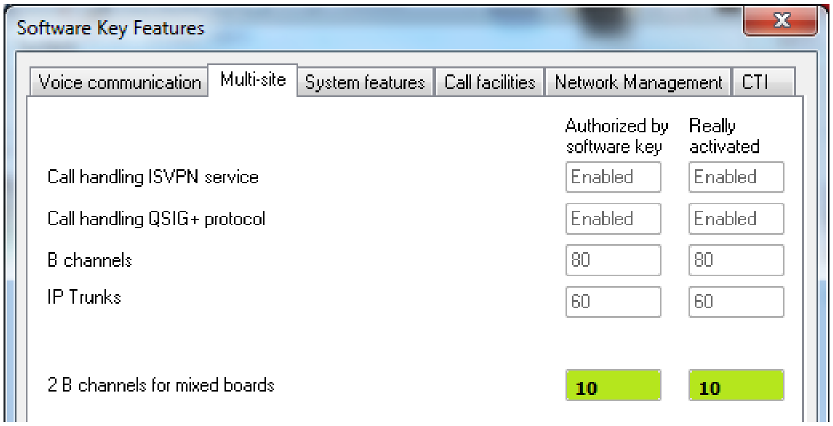
Warning **ALE may not be aware of changes made by the Operator. In case of any issue or doubt in relation with those basic SIP Trunk parameters, please contact the SIP Provider directly.**

2.2 System Connection procedure

The configuration task involves on-line connection to the IPBX using the OMC Expert-level session. 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.



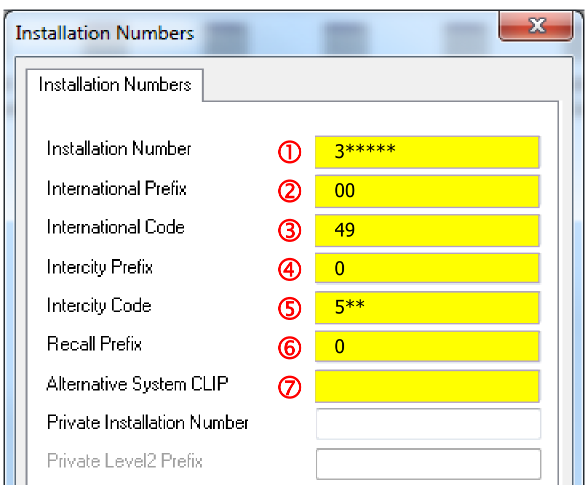
Note

The values highlighted in green are fixed by the system SW license.

2.4 Numbering Plan configuration

2.4.1 Installation numbers

No matter the type of Trunk considered, OXO's handling of public numbers is first based on the "Installation Numbers" data configured in OMC Numbering -> Installation Numbers.

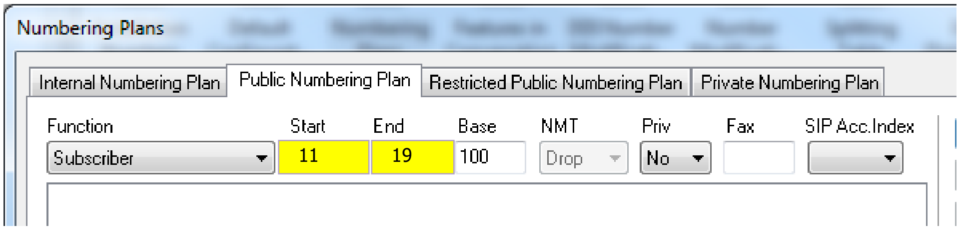


- Check/ edit the corresponding numbers as illustrated above:

- ① Install. Number : ☞● **NP_Instal_Number_example = "3*****"**
- ② International Prefix : ☞● **NP_International_Prefix = "00"**
- ③ International Code : ☞● **NP_International_Code = "49"**
- ④ Intercity Prefix : ☞● **NP_Intercity_Prefix = "0"**
- ⑤ Intercity Code : ☞● **NP_Intercity_Code_example = "5**"**
- 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.



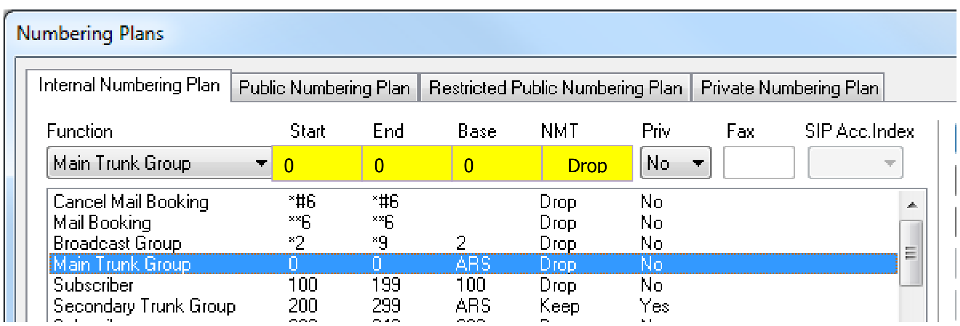
- Check/ edit the configuration for "Public Numbering Plan":

- DDI range : ☞● **NP_DDI_Range_example = "11 19"**

Note In conjunction with the configuration of section 2.4.1, this basic example allocates the DDI range " 11 19" to the range of extensions beginning at "100".

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.



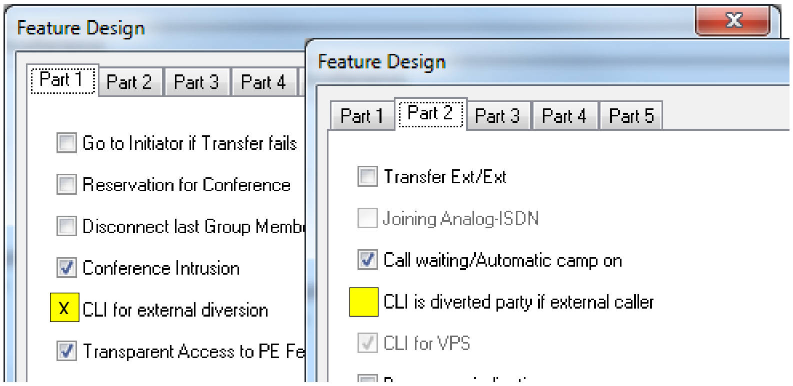
This example defines the access to the system ARS table for phone numbers dialed that start with digit 0. The "Drop" attribute also indicates that the initial 0 of the number is dropped before it is passed to the ARS Prefix table.

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.

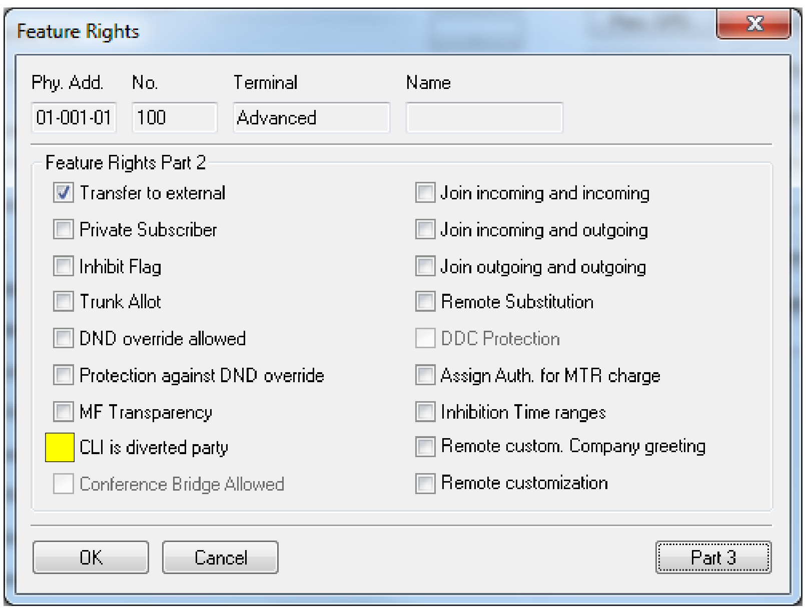
From the tabs "Part 1" and "Part 2" of menu System Misc -> Feature Design, verify the parameters:

- "CLI for external diversion": ☞●**Misc_CLI_Ext_Diversion= True**
- "CLI CLIisdiverted 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 subscriber's feature rights is necessary for enabling outbound calls and other features over the SIP trunk.

3 SIP Trunk Setup

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

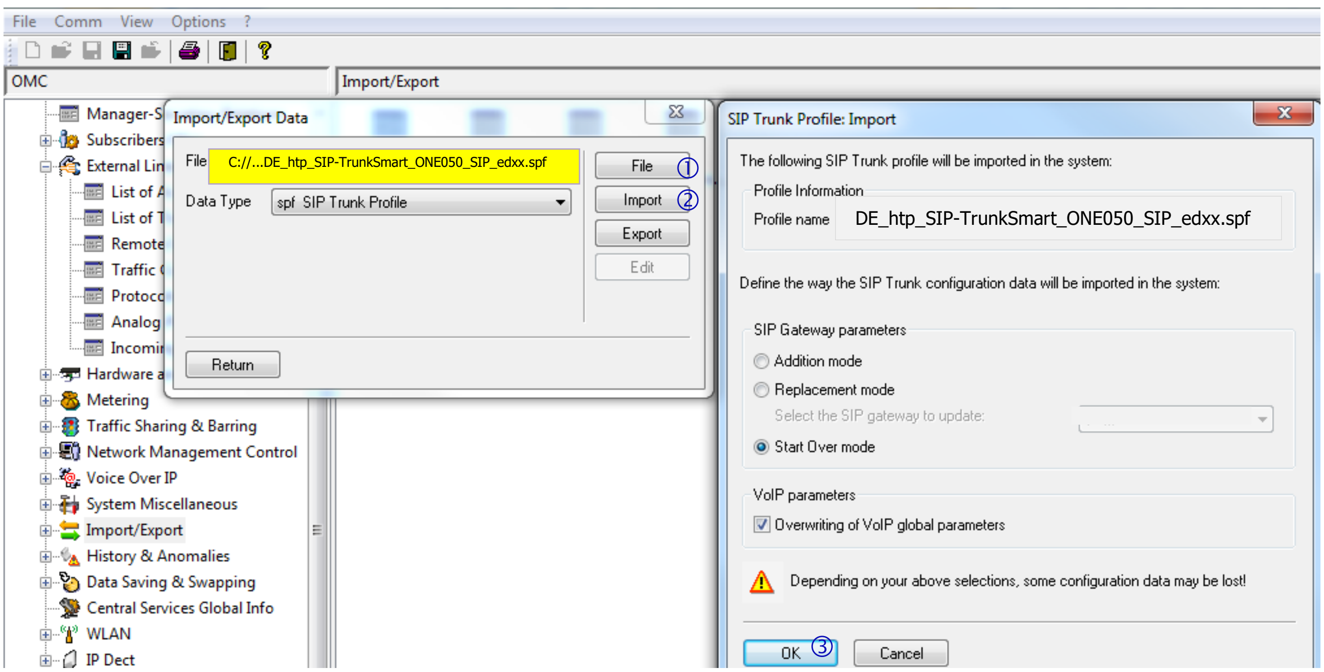
Warning

For proceeding with next configuration steps, you should have on your PC the file "DE_htp_SIP-TrunkSmart_ONE050_SIP_edxx.spf" which is the SIP Trunk Profile associated to this guide.

Note

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 drawing hereafter summarizes the import steps (operation detailed in the Bulletin TC1994).

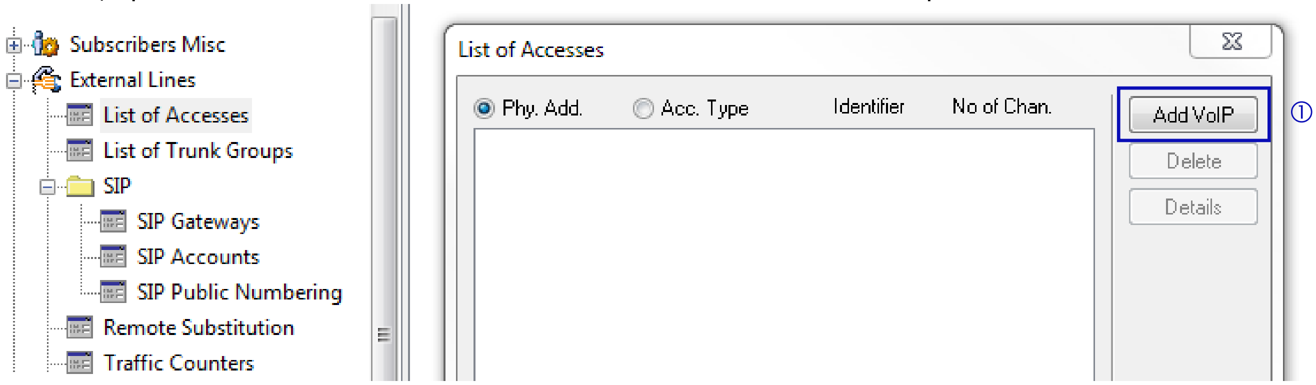


Warning

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

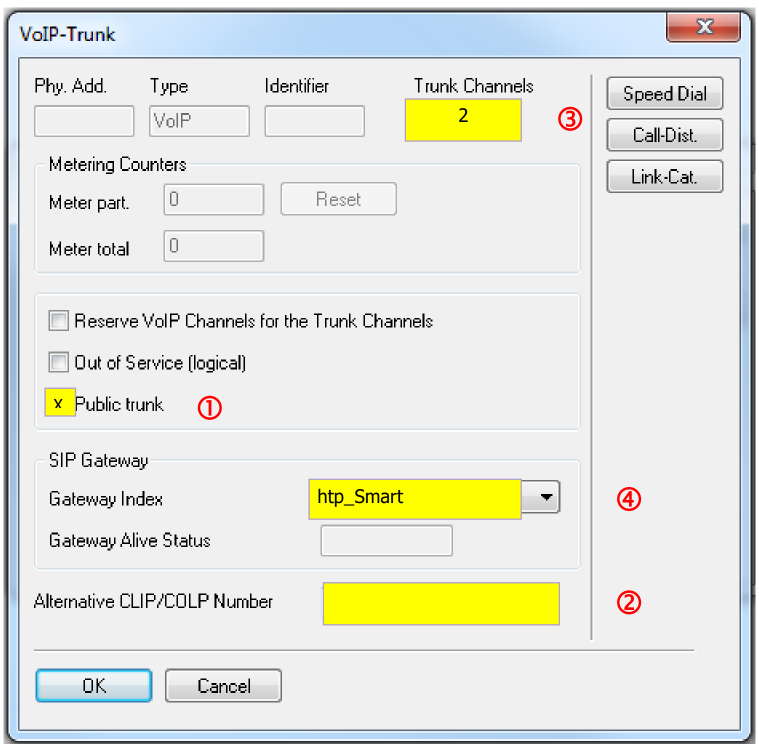
3.2 Creating the Voip Trunk

On OMC, open the tab External Lines -> List of Accesses and click on "Add Voip" to create a VOIP Trunk.



3.3 Assigning the trunk to a Gateway

Complete then the requested parameters

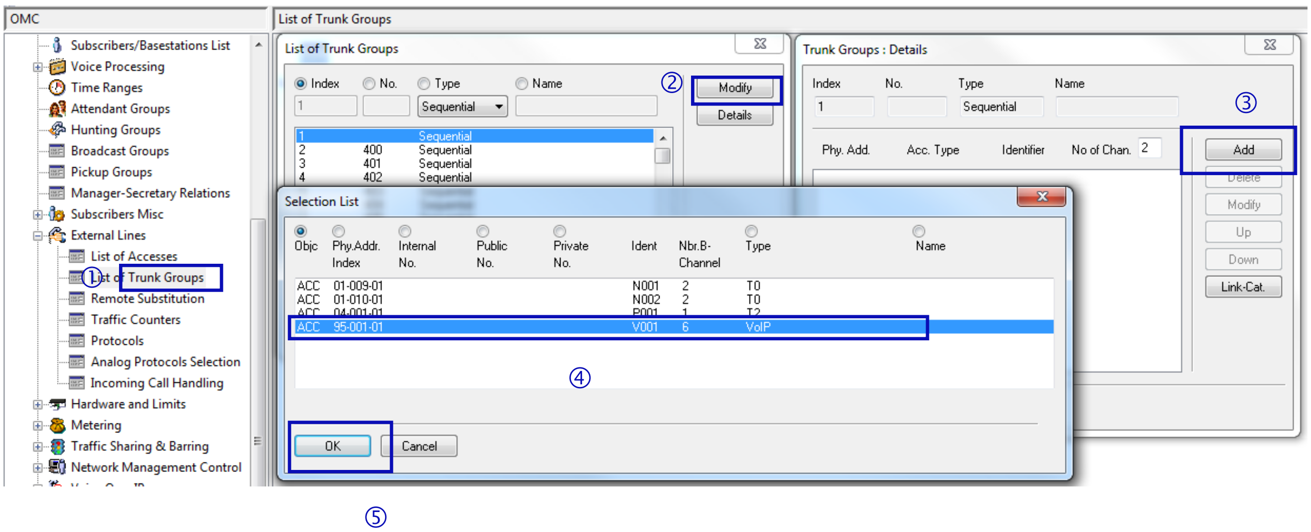


Then, configure the parameters corresponding to this VoIP Access:

- ① "Public trunk" option : ☞● **Access_is_Public = True**
- ② "Alternative CLIP/COLP Number" : ☞● **Access_Alt_CLIP_example = " "**
- ③ Number of VoIP-Trunk Channels : ☞● **VoIPgen_Trunk_Channels_example = 2**
- ④ "Gateway Index" : ☞● **Gatway Index = htp**

3.4 Hosting System Trunk Group

To enable phone calls over the SIP trunk, it's also necessary to have this latter included within one Trunk Group of the system. Two alternative cases (variants) are considered here below.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 5 depicted by blue digits 1 to 5)



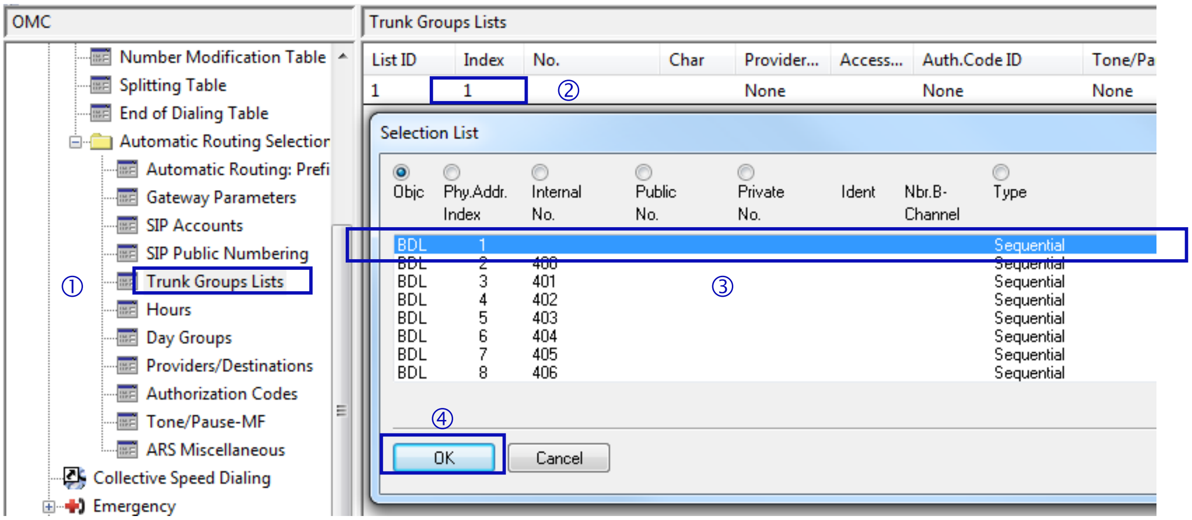
1) As a configuration variant, at step 2 you can include the SIP trunk access into the OXO's main Trunk Group (i.e. step 2a for index #1) or into one of the secondary Trunk Groups (e.g. step 2b for index #2).

Note 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.

3.5 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.

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



Carry out the selections and push-button of steps 1 to 4 above. At step 3, you need to select the line index corresponding to the System Trunk Group previously defined at section 3.4 .

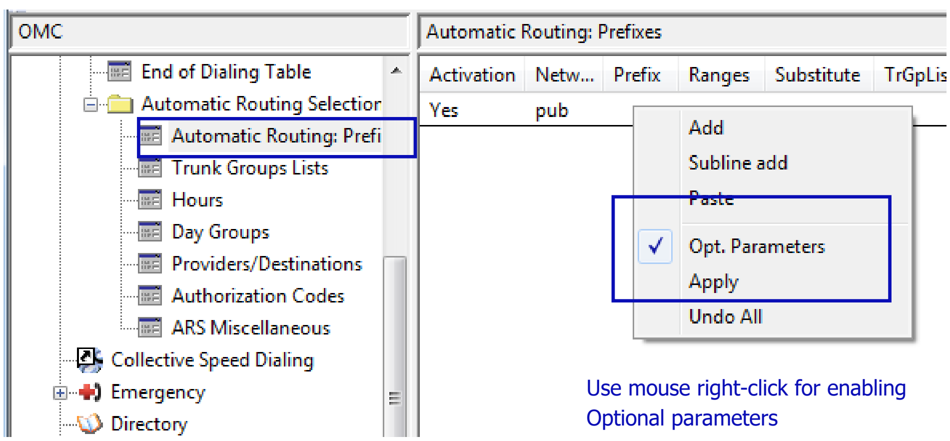
3.6 Complementary Setup

The steps of Ch. 3.6 are needed for completing the OMC configuration not managed by SIP Easy Connect.

3.6.1 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.

Automatic Routing: Prefixes

Activation	Netw...	Prefix	Ranges	Substitute	TrGpList	Called(ISVPN/H450)	User comment	Metering	Calling	Called/PP
Yes	pub	0	0-9	0	1	het		Blank	default	default

In the call routing table, additional lines can be created to cope with specific public phone numbers (e.g. short numbers or emergency numbers). Here below is a typical example for France, comprising four Prefix entries/ ranges (customized values in area 1 and 2):

Automatic Routing: Prefixes

Activation	Netw...	Prefix	Ranges	Substitute	TrGpList	Called(ISVPN/H450)	User comment	Metering	Calling	Called/PP
Yes	pub	0	0-9	0	1	hom fwd	Line 1	Blank	default	default
Yes	emerg			0	1		Line 2	Blank	default	default
Yes	pub	3	0-9	3	1		Line 3	Blank	default	pub shortN
Yes	pub	1	0-9	1	1		Line 4	Blank	default	pub shortN

① ② ③

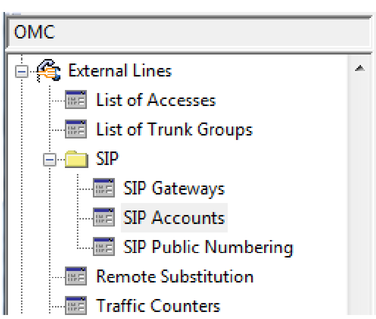
- Line 1: copes with standard phone numbers starting with digit 0 (national / international calls)
- Line 2: copes with all public emergency numbers. The network attribute "emerg" permits the line to point automatically to the system list of emergency numbers. This list is country-dependent and can be edited via the OMC menu Emergency-> Emergency Numbers
- 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 = hom fwd**

3.6.2 ARS SIP Accounts

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



SIP Accounts

Index	Login	Password	Registered Username	Gateway Parameters Index	RFC 6140
1	+***********	*****	+************	htp_smart	False
	①	②	③	④	⑤

- ① "Login" : ☞● **SIPaccnt_Login_example = " +************"**
- ② "Password" : ☞● **SIPaccnt_Password = " *****"**
- ③ "Registered User." : ☞● **SIPaccnt_Reg_Username_example = " +************"**
- ④ the "Gateway Parameters Index" must point to the relevant gateway (i.e. index 1)
- ⑤ "RFC 6140" : ☞● **SIPaccnt_RFC6140_Enabled= False**

Note

For solutions using several individual lines, it is necessary to create one SIP Account line per line (multi-account configuration). Otherwise, for other trunk situations a single SIP Account line is generally sufficient.

Note

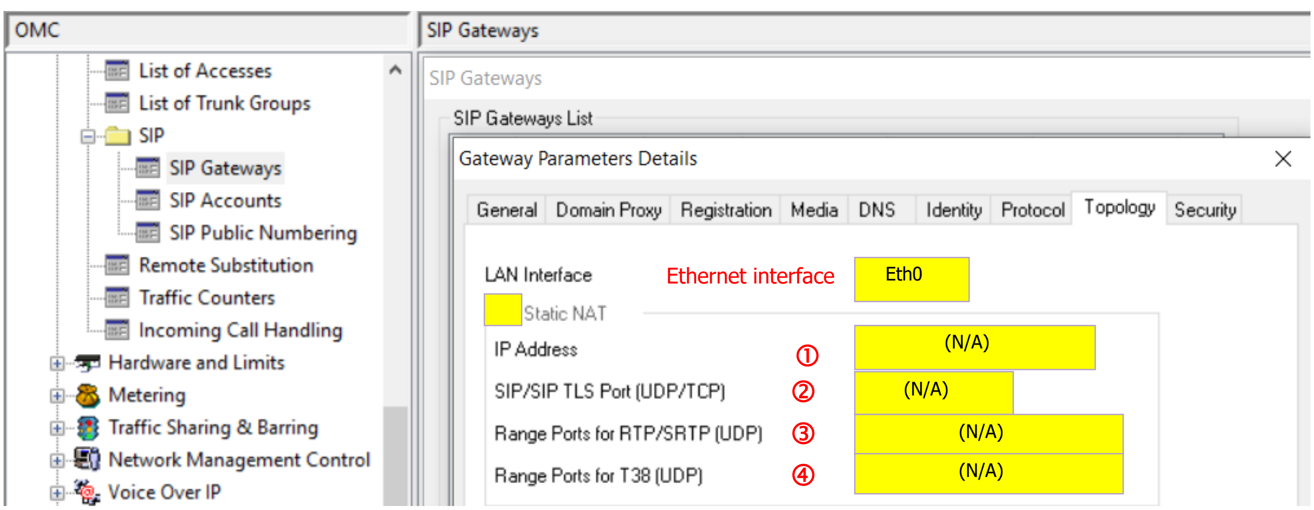
Since OXO Connect R4.0, there are two additional configuration parameters that do not belong to the profile. Their value (if any) is given as example in chapter 4 – Table 3. This is only a display info for the called party in case of outgoing calls.

3.6.3 VoIP Topology Tab

Configuration of "static SIP/NAT" is required for solutions using the topology model referred as "Topology D" (**NOT relevant for this operator**).

Warning

Reminder: on the local CPE router, port forwarding to OXO of the relevant SIP ports must be configured accordingly.



- ① "IP Address" : ☞● **GWtopo_SNAT_PubIP_example = "(N/A)"**
- ② "SIP Port" : ☞●✓ **GWtopo_SNAT_SIP_Port_example = "(N/A)"**
- ③ "RTP Ports Range" : ☞●✓ **GWtopo_SNAT_RTP_Range_example = "(N/A)"**
- ④ "T38 Ports Range" : ☞●✓ **GWtopo_SNAT_T38_Range_example = "(N/A)"**

Note — Internet Interface: This value is relevant only for IPBOX Release >= 5 (Eth1 is not managed on Power CPUEE). It is the value retrieved from the system used as pilot site during the acceptance process.
- OCO: Value will always be Eth0 (Eth1 not managed)
- OCE: Whatever the value configured before, it will be kept unchanged

3.6.4 System Flags

Some specific "Noteworthy addresses" not imported by SIP Easy Connect need to be configured manually. **These system flags are listed at the bottom part of Table 1 in the configuration abstract of Ch.5.**

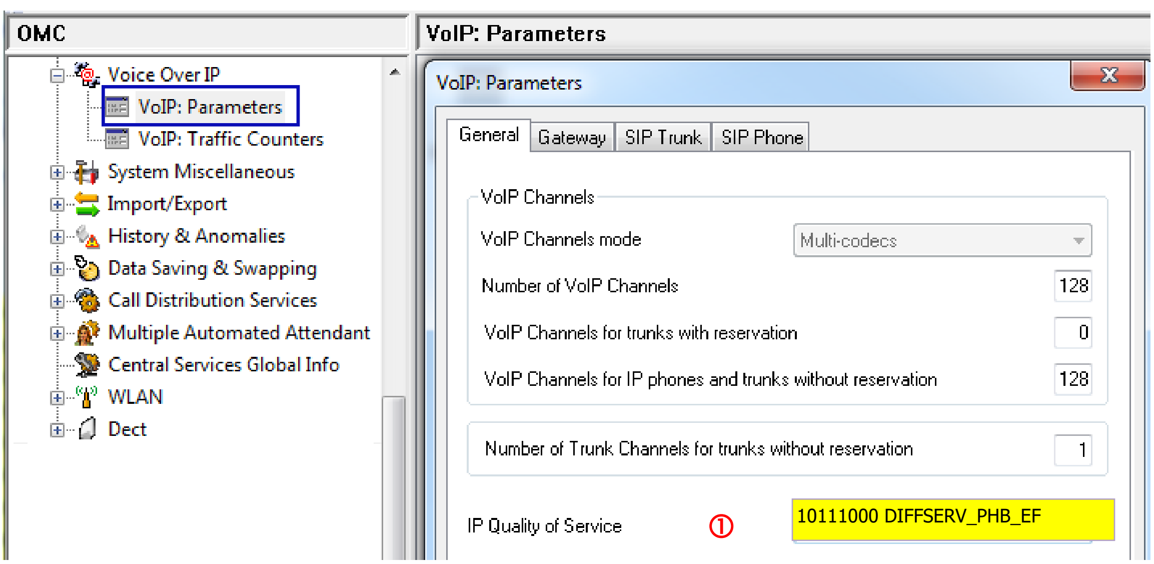
The access to the system flags is made with the OMC menu System Miscellaneous -> Memory Read/Write (Debug Labels or Other Labels). Please refer to the indications and comments given in the configuration abstract Table and apply carefully the required flag changes on OMC.

3.7 Adjustments (fine tuning)

The configuration steps of Ch. 3.7 refer to particular adjustments you can carry on over the data imported by SIP Easy Connect (data highlighted within the OMC screenshots).

3.7.1 VoIP General Tab

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

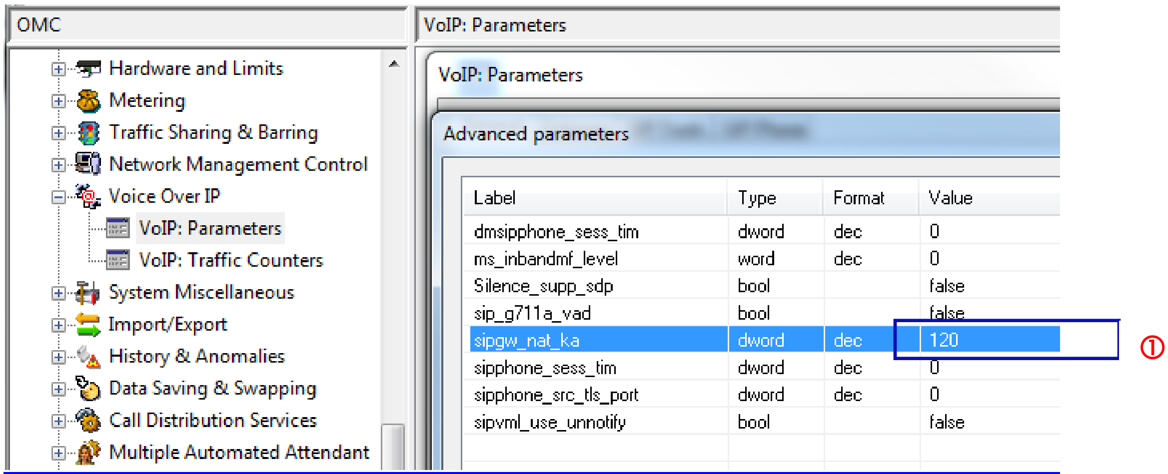


- Adjust the IP Quality of Service of VoIP trunks (RTP flow):

① ☞●✓ **VoIPgen_IP_QoS_example = 10111000 DIFFSERV_PHB_EF**

3.7.2 VoIP Advanced Tab

Open the OMC tab via the menu Voice Over IP -> VOIP:Parameters – SIP Trunk
In any of the tabs, click on advanced



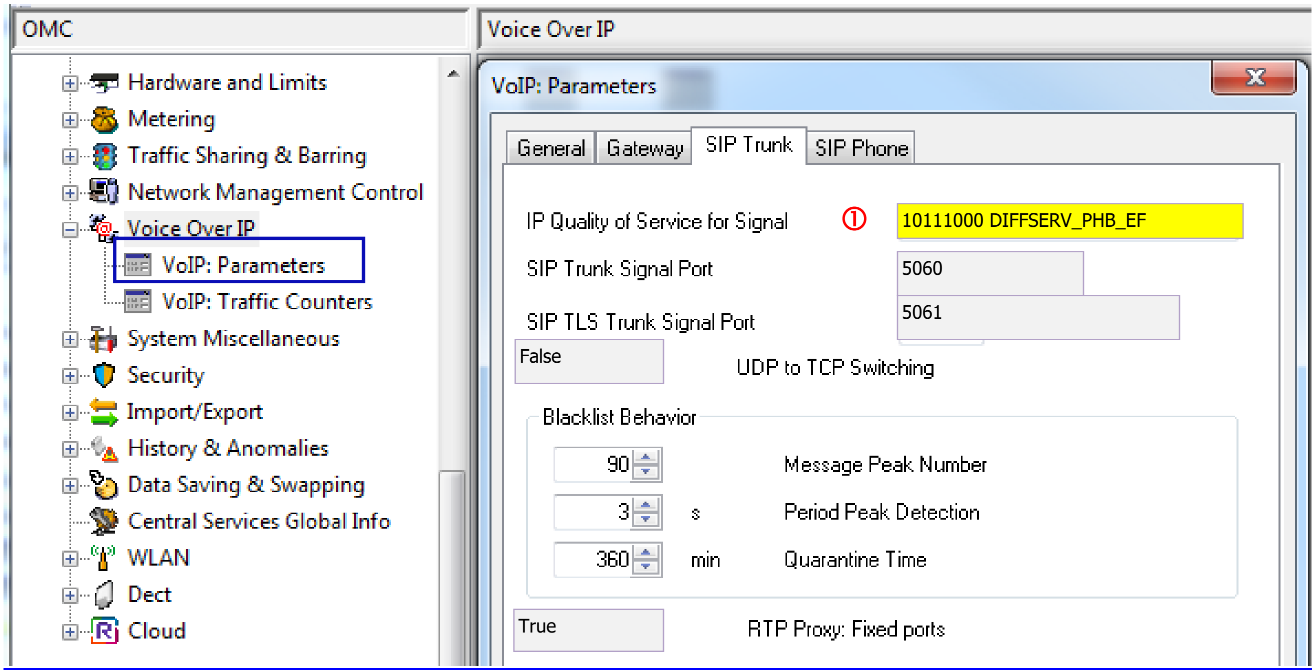
In case of Dynamic Mode, adapt the value of the "sipgw_nat_ka" timer to the router "NAT keep Alive Timer" used to connect to the SIP provider.

① ☞●✓ **sipgw_nat_ka = 120 seconds for ie**

Warning **This is the periodic timer used to send a SIP "OPTION" message to the provider. It can be used to maintain the router bindings opened for receiving incoming requests. Each router Brand/Release/Version may have a specific timer value not known by ALE. In case of any issue or doubt in relation with this parameter, please contact the SIP or router Provider directly.**

3.7.3 VoIP SIP Trunk Tab

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



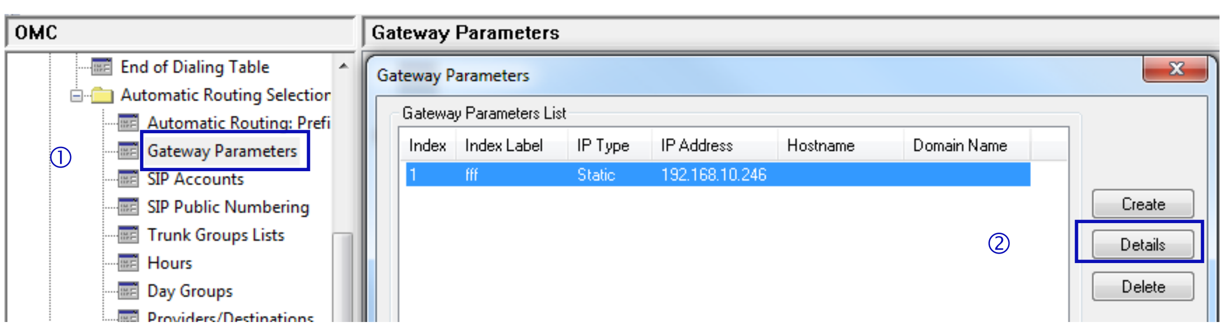
-Adjust the IP Quality of Service of SIP Trunk messages (SIP signaling) :

① ☞●✓ **VoIPsiptrk_QoS_example = 10111000 DIFFSERV_PHB_EF**

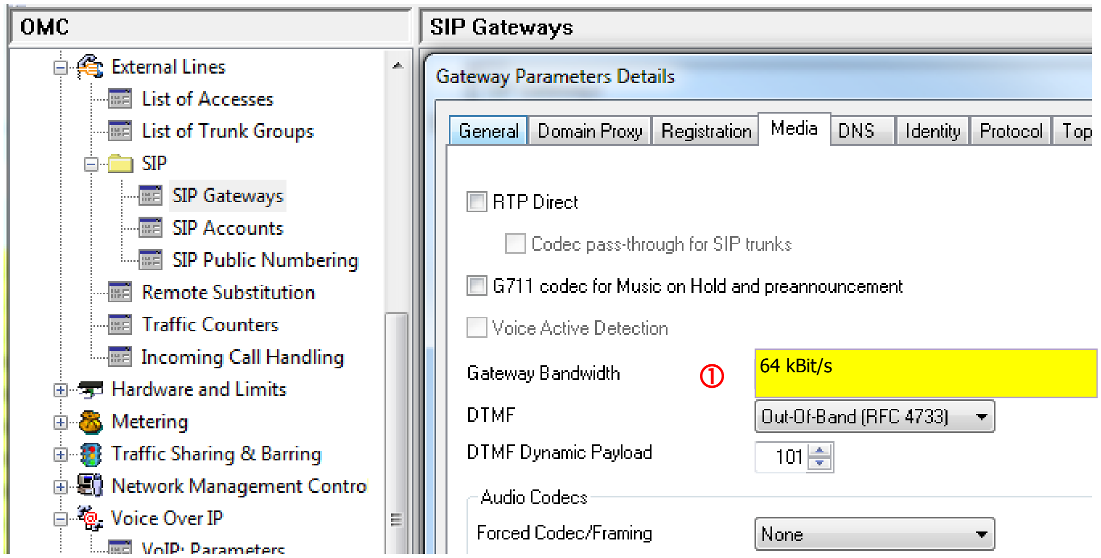
3.7.4 Gateway Media Tab

Double-click on menu External Lines -> SIP -> SIP Gateways

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 Media tab of the window "Gateway Parameters Details".

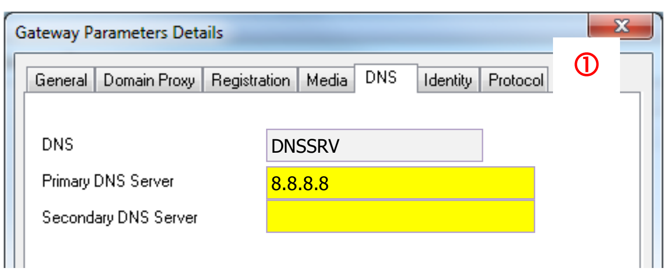


Adjust Bandwidth to the site context:

- ① ☞●✓ **GWmedia_Bwidth_example = 64 kBit/s**

3.7.5 Gateway DNS Tab

From the menu External Lines -> SIP -> SIP Gateways, select the DNS tab of the window "Gateway Parameters Details".

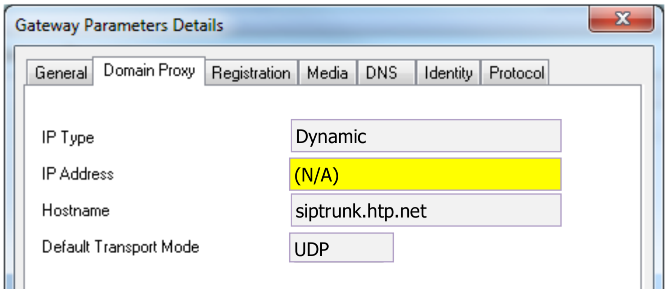


Replace the IP address 8.8.8.8 by the private IP address of the LAN router (same value configured as "Default Router Address" in OMC menu HW & Limits -> LAN-IP configuration -> LAN Configuration Tab)

① ☞●✓**GWdns_Prim_DNS =** (Default Router IPV4 Address)

3.7.6 Gateway Domain_Proxy Tab

The Operator will confirm the exact IP address of its SIP server to be configured in OMC. Use the OMC menu External Lines -> SIP -> SIP Gateways then press the button "Details" and then select the Domain Proxy tab of the window "Gateway Parameters Details".



Change if necessary, the IP address imported from the SIP profile to the relevant values defined by the SIP Operator htp:

☞●✓ **GWdom_IP_Address = ...**

4 SIP trunk Configuration Abstract

The following tables gather the overall system configuration (the '✓' sign of the SEC column corresponds to the values imported via the SIP Easy Connect facility).

Table 1 (System General)

CONFIG OXO ONE030	VALUE	SEC	REMARK
/Numbering/...			
NP_Instal_Number_example	3*****		Value given as example
NP_International_Prefix	00		
NP_International_Code	49		
NP_Intercity_Prefix	0		
NP_Intercity_Code_example	5**		Value given as example
NP_Recall_Prefix	0		
NP_System_Alt_CLIP_example			Value given as example
NP_DDI_Range_example	11 19		Value given as example
/Numbering/ ARS			
ARS_Called_Mode	hom fwd		
/External Lines/ ListOf Accesses -VoIP			
Access_is_Public	True		
Access_Alt_CLIP_example			Value given as example
Access_Channels_example	2		Value given as example
/ Misc/Feature Design			
Misc_CLI_Ext_Diversion	True		
Misc_CLI_is_Diverted_Party	False		
/ Misc/Memory Read-Write			

Table 2 (Voice Over IP)

CONFIG OXO ONE030	VALUE	SEC	REMARK
/VoIP/VoIP Parameters/General			
VoIPgen_IP_QoS_example	10111000 DIFFSERV_PHB_EF	✓	Value given as example
/VoIP/VoIP Parameters/SIP Trunk			
VoIPsiptrk_QoS_example	10111000 DIFFSERV_PHB_EF	✓	Value given as example
VoIPsiptrk_SIPSourcePort	5060	✓	
VoIPsiptrk_SIPTLSSourcePort	5061	✓	
VoIPsiptrk_UdpToTcp	False	✓	
VoIPsiptrk_RTPpxyPortsFixed	True	✓	

Table 3 (SIP Accounts)

CONFIG OXO ONE030	VALUE	SEC	REMARK
/External Lines/SIP/SIP Accounts			
SIPaccnt_Login_example	+************		Value masked partially
SIPaccnt_Password	*****		
SIPaccnt_Reg_Username_example	+************		Value masked partially
SIPaccnt_RFC6140_Enabled	False		
SIPaccnt_Company_Name			Value given as example if any
SIPaccnt_Outgoing	None		

Table 4 (SIP Public Numbering)

CONFIG OXO ONE030	VALUE	SEC	REMARK
/External Lines/SIP/Public Numbering			
SIPnum_Out_Calling_Format	Canonical	✓	
SIPnum_Out_Calling_Prefix	+	✓	
SIPnum_Out_Called_Format	Canonical	✓	
SIPnum_Out_Called_Prefix	+	✓	
SIPnum_Out_Called_Short_Prefix		✓	
SIPnum_Inc_Calling_Format	Canonical/International	✓	
SIPnum_Inc_Calling_Prefix	+	✓	
SIPnum_Inc_Called_Format	Canonical/International	✓	
SIPnum_Inc_Called_Prefix	+	✓	
SIPnum_Alt_CLIP_example			Value given as example and masked

Table 5 (GW Parameters)

CONFIG OXO ONE030	VALUE	SEC	REMARK
/External Lines/SIP/SIP Gateways/Details/Domain Proxy			
GWgen_eod_timeout	3.5 s	✓	
GWgen_eod_table_used	False	✓	
/External Lines/SIP/SIP Gateways/Details/Domain Proxy			
GWdom_IP_Type	Dynamic	✓	
GWdom_IP_Address	(N/A)	✓	Dynamic value
GWdom_Def_Transport	UDP	✓	
GWdom_Target_Domain	siptrunk.htp.net	✓	
GWdom_Local_Domain_Name		✓	
GWdom_Realm		✓	
GWdom_Remote_SIP_Port	5060	✓	Dynamic value
GWdom_Outb_Proxy	siptrunk.htp.net	✓	
/External Lines/SIP/SIP Gateways/Details/Registration			
GWreg_Reg_Requested	True	✓	
GWreg_Check_Before_Req	True	✓	
GWreg_Reg_Name	siptrunk.htp.net	✓	
GWreg_Reg_IP_Address	(N/A)	✓	Dynamic value
GWreg_Reg_Port	(Dynamic)	✓	Dynamic value
GWreg_Reg_Expire_Time	300	✓	
GWreg_Reg_AoR_In_Contact	False	✓	
GWreg_Reg_AoR_In_From	False	✓	
GWreg_Reg_AoR_In_PAI	True	✓	
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	✓	
GWreg_RFC3327_Enabled	False	✓	
/External Lines/SIP/SIP Gateways/Details/Media			
GWmedia_RTP_Direct	False	✓	
GWmedia_Trunk_Codec_Passthru	False	✓	
GWmedia_G711_MOH	True	✓	
GWmedia_DSP_VAD	False	✓	
GWmedia_Bwidth_example	64 kBit/s	✓	
GWmedia_DTMF_Mode	Out-Of-Band (RFC 4733)	✓	
GWmedia_DTMF_Payload	101	✓	Value given as example

GWmedia_Forced_Codec_Framing	None	✓	
GWmedia_Selected_Codecs	G711.a	✓	
GWmedia_Preferred_Framing	20 ms	✓	
GWmedia_Fax_Mode	G711	✓	
GWmedia_T38_Add_Signal	None	✓	
GWmedia_T38_CED_Tone	False	✓	
GWmedia_T38_UDP_Redundancy	1	✓	
GWmedia_T38_Fax_Framing	0	✓	
GWmedia_T38_ECM	False	✓	
/External Lines/SIP/SIP Gateways/Details/DNS			
GWdns_DNS_Mode	DNSSRV	✓	
GWdns_Prim_DNS	8.8.8.8	✓	
GWdns_Sec_DNS		✓	
/External Lines/SIP/SIP Gateways/Details/Identity			
GWident_RFC3325	True	✓	
GWident_Routing_To_Header	False	✓	
GWident_HistInfo_DivHeader	History-Info	✓	
GWident_Inc_CLI_Headers	P-Asserted-Identity P-Preferred-Identity From Reserved-1 Reserved-2 Reserved-3 Reserved-4 Reserved-5	✓	
GWident_Out_CLI_PPI_Used	False	✓	
GWident_Out_CLI_PAI_Used	True	✓	
GWident_Out_COLP_Headers	P-Asserted-Identity P-Preferred-Identity Contact To Reserved-1 Reserved-2 Reserved-3 Reserved-4	✓	
GWident_AltCLIP_Contact	True	✓	
GWident_AltCLIP_From	True	✓	
GWident_AltCLIP_PAI	True	✓	
GWident_AltCLIP_PPI	True	✓	
GWident_AltCLIP_Rsv1	False	✓	
GWident_AltCLIP_Rsv2	False	✓	
GWident_AltCLIP_Rsv3	False	✓	
GWident_AltCLIP_Rsv4	False	✓	
GWident_EmergLocID_PANI	False	✓	
/External Lines/SIP/SIP Gateways/Details/Protocol			
GWprot_SessTimer_Time	720 min	✓	
GWprot_PEM_Enabled	False	✓	
GWprot_UPDATE_Enabled	True	✓	
GWprot_PRACK_Enabled	True	✓	
GWprot_GWalive_Prot	(N/A)	✓	
GWprot_GWalive_Timer	(N/A)	✓	
GWprot_RFC4904_Enabled	False	✓	
GWprot_RFC4904_Trk_GID		✓	
GWprot_RFC4904_Trk_Context		✓	
/External Lines/SIP/SIP Gateways/Details/Topology			
GWtopo_SNAT_Enabled	False	✓	
GWtopo_LAN_Interface	Eth0	✓	
GWPtopo_SNAT_PubIP	(N/A)	✓	Value given as example and masked
GWPtopo_SNAT_SIP_Port	(N/A)	✓	Value given as example
GWPtopo_SNAT_RTP_Range	(N/A)	✓	Value given as example
GWPtopo_SNAT_T38_Range	(N/A)	✓	Value given as example
/External Lines/SIP/Gateway Parameters Details/Security			
GWsec_SIPTLS_Enabled	False	✓	

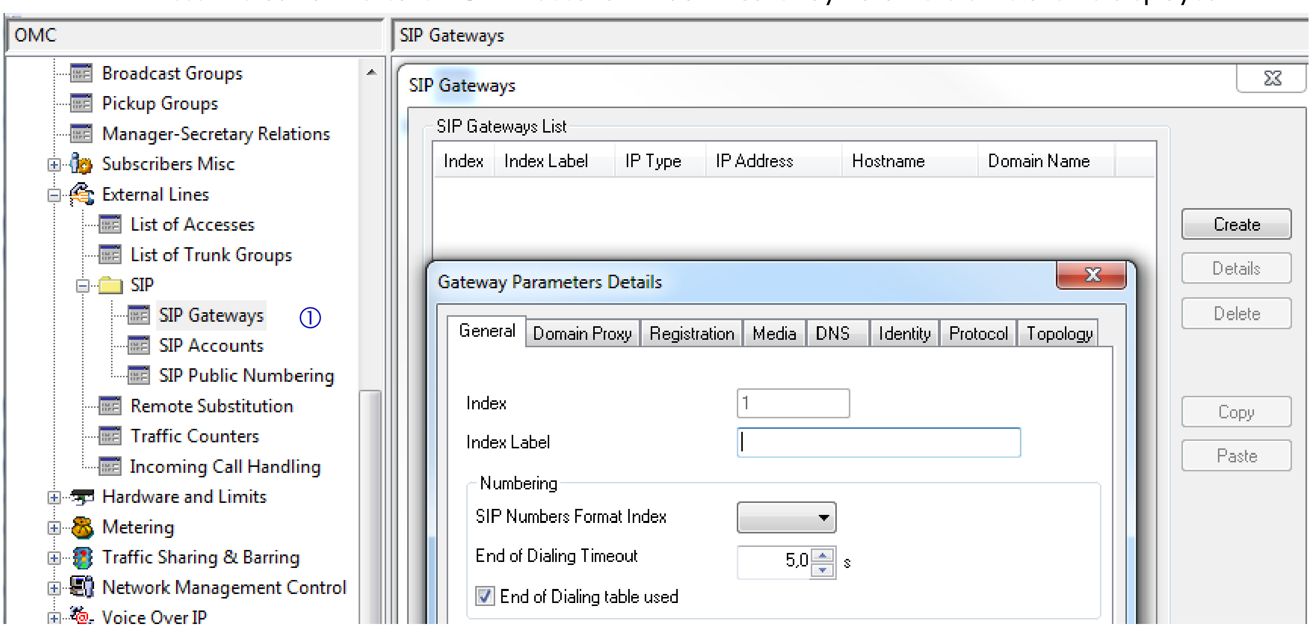
GWsec_SIPTLS_Mutual_Auth_Enabled	False	✓	
GWsec_SIPTLS_Wildcard_Char_Allowed	False	✓	
GWsec_SIPTLS_SIPS_URI_Enabled	False	✓	
GWsec_SRTP_Enabled	False	✓	
GWsec_Crypto_Suites_KeyLifetime_Enabled	False	✓	
GWsec_Crypto_Suites_Selected	AES_CM_128_HMAC_SHA1_80 AES_CM_256_HMAC_SHA1_80 AES_CM_128_HMAC_SHA1_32 AES_CM_256_HMAC_SHA1_32	✓	
/External Lines/SIP/Gateway Parameters Details/Advanced parameters			
GWadv_auth_optimize	false	✓	
GWadv_ExtNuFoVoip	22	✓	
GWadv_FaxPasCd	01ff	✓	
GWadv_FlgHoldExTone	false		
GWadv_inhibit_t38	0	✓	
GWadv_initial_reg_username	false	✓	
GWadv_INVwSDPtrk	false	✓	
GWadv_MultAnsReinv	true	✓	
GWadv_multiple_option_req	false	✓	
GWadv_MYICcaller	00	✓	
GWadv_no_rport	0	✓	
GWadv_PrefCodec	0000	✓	
GWadv_PrefFraming	0	✓	
GWadv_PrimaryGW	0		
GWadv_reg_Aor_localDomain	false		
GWadv_rfc4916_off	0	✓	
GWadv_SimulIpAlt	true	✓	
GWadv_sip_capa	false	✓	
GWadv_SIPdtmfInB	false	✓	
GWadv_sipgw_fax_offer	false	✓	
GWadv_sipgw_namedisp	false	✓	
GWadv_sipgw_noresource_error			
GWadv_sipgw_prefid	0	✓	
GWadv_sipgw_priv_lvl	false	✓	
GWadv_sipgw_reg_trigger_01	0	✓	
GWadv_sipgw_reg_trigger_02	0	✓	
GWadv_sipgw_reg_trigger_03	0	✓	
GWadv_sipgw_reg_trigger_04	0	✓	
GWadv_sipgw_reg_trigger_05	0	✓	
GWadv_sipgw_reg_trigger_06	0	✓	
GWadv_sipgw_reg_trigger_07	0	✓	
GWadv_sipgw_reg_trigger_08	0	✓	
GWadv_sipgw_reg_trigger_09	0	✓	
GWadv_sipgw_reg_trigger_10	0	✓	
GWadv_sipgw_regid	0	✓	
GWadv_sipgw_rem_maxptime	0	✓	
GWadv_sipgw_Req_URI_route_call	false	✓	
GWadv_sipgw_to_ruri	false	✓	
GWadv_sipgw_update_trigger	false		
GWadv_sipgw_voip_caun	false	✓	
GWadv_SIPInDspNm	01	✓	
GWadv_SIPOgDspNm	01	✓	
GWadv_special_char_truncation	false	✓	

GWadv_SuprAlerTone	false	✓	
GWadv_t4_jit	0	✓	
GWadv_trigger_alert	false	✓	
GWadv_USalterfrom	false	✓	
GWadv_userlvlpri	false	✓	
GWadv_v21_jit	0	✓	

5 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.3
- **Stage 2**): complete steps of the guide from chapter 3.2 and create manually a new SIP Gateway entry as illustrated in the following picture:
  - Double-click on menu External Lines -> SIP -> SIP Gateways ①.
  - A new window "SIP Gateways" is displayed.
  - Press the button "Create" ②. A second window "Gateway Parameters Details" is displayed.



- **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
  - Registration
  - Media
  - Etc…

These tabs are not specifically detailed in the doc, nor illustrated with OMC screenshots. **You must then refer to the configuration abstract of Ch.4** and apply on OMC the exact value of parameters found in the **appropriate sections of table 5**. When achieved, terminate the creation of the gateway by pressing OK button

- **Stage 4):** proceed with all the operation steps of Ch.3.3 to Ch.3.7: **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. all parameters also including those not highlighted in yellow).**
- **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.4 and refer to columns "VALUE" and "SEC" of the tables.

Submitting a Service Request

Please connect to our eService Request application.

Before submitting a Service Request, please be sure:

- The application has been certified via the AAPP if a third party application is involved.
- You have read the release notes that list new features, system requirements, restrictions, and more, and are available in the Technical Documentation Library.
- You have read through the related troubleshooting guides and technical bulletins available in the Technical Documentation Library.
- You have read through the self-service information on commonly asked support questions and known issues and workarounds available in the Technical Knowledge Center.

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