SIP TRUNK SOLUTION ENVIATEL (DE): CONFIGURATION GUIDELINE ONE02X

This document details how to set up an IPBX OXO ONE02X for enabling a public SIP trunk of the Operator ENVIATEL in GERMANY.

Revision History

Edition 01: June 29, 2018 first ONE02X edition published for GA support

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1 General

This document describes the from-scratch configuration of OXO ONE02X in the context of a SIP trunk solution connected to the public Operator **EnviaTel**. 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

This reference guide relies on the SIP Trunk Profile " DE_Enviatel_ONE02X_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.6 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)

☞● **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**

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 = 6**
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:☞● **NP_Instal_Number_example = "8***0"**

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	Alcatel-Lucent OmniPCX Office	ONEDE022/029.001
OMC Management Application	Alcatel-Lucent OMC	OMC22.0/11.1b

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

EnviaTel **TOPOLOGY: C** **WITHOUT DIRECT RTP**	40x8 80x8 80X8s	40x9 80x9 Z DECT	IP DECT DAP's	MyIC 8082	8012	4135	8001	MyIC And.	MyIC IPhone	MyIC SIP
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	**NT**	
Inbound Fax G711	**NT**	

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 speakers number.

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.

ENVIATEL SIP TRUNK PARAMETERS

Data Type	Parameter role	Name in doc	Value
Provider-specific	OP Gateway IP@	GWdom_IP_Address	
	OP GW Domain name	GWdom_Target_Domain	sip.enviatel.net
	Outbound Proxy	GWdom_Outb_Proxy	
	DNS IP@ Prim	GWdns_Prim_DNS	193.98.112.238
	DNS IP@ Sec	GWdns_Sec_DNS	193.98.115.214
	Registrar IP address	GWreg_Reg_IP_Address	
	Registrar name	GWreg_Reg_Name	sip.enviatel.net
	SIP Realm	GWdom_Realm	
	Local Domain name	GWdom_Local_domain_Name	sip.enviatel.net
Site specific (example values)	Installation Number	NP_Instal_Number_example	8***0
	Instal. Alternative CLI	SIPnum_Alt_CLIP_example	
	Public DDI range	NP_DDI_Range_example	60-69
	Registered Username	SIPaccnt_Reg_Username_example	0******6
	Authentication Login	SIPaccnt_Login_example	0******6
	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

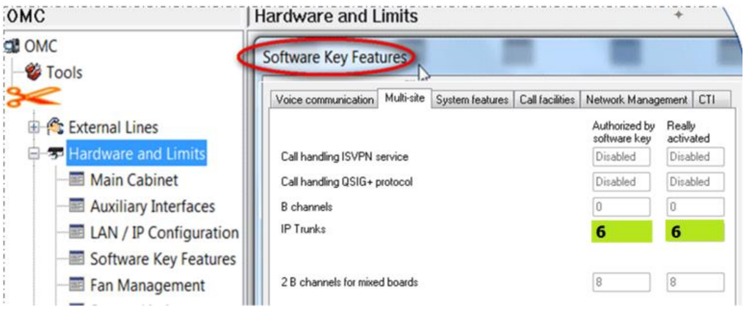
ALE may not be aware of changes made by the Operator. In case of any issue or doubt in relation with those basic SIP Trunk parameters, please contact the SIP Provider directly.

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.



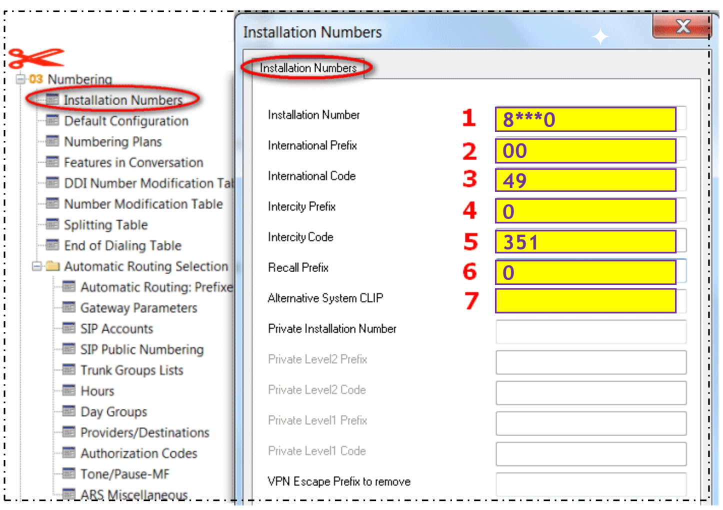
Note

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.

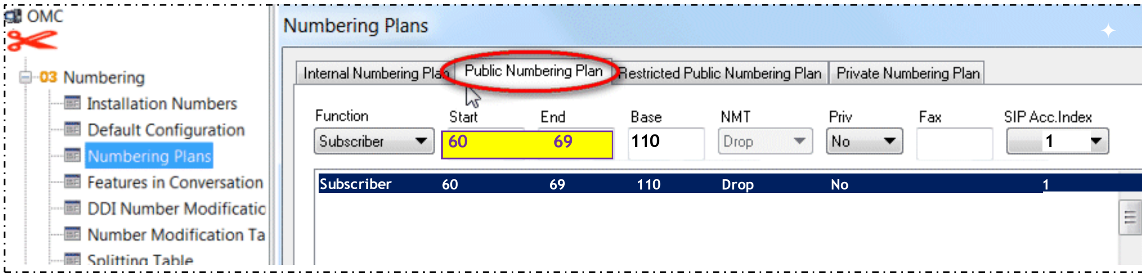


- Check/ edit the corresponding numbers as illustrated above:

- ① Install. Number : ☞● **NP_Instal_Number_example = "8***0"**
- 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 = "351"**
- 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":

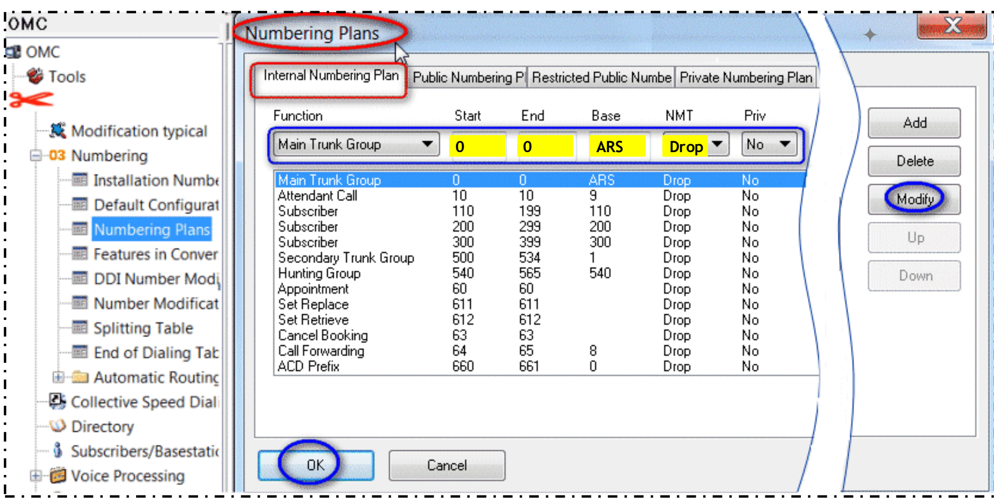
- ① DDI range : ☞● **NP_DDI_Range_example = "60 69"**

Note

In conjunction with the configuration of section 2.4.1, this basic example allocates the DDI range "60-69" (i.e. 03518***060 - 03518***069) to the range of extensions "110 - 119".

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.



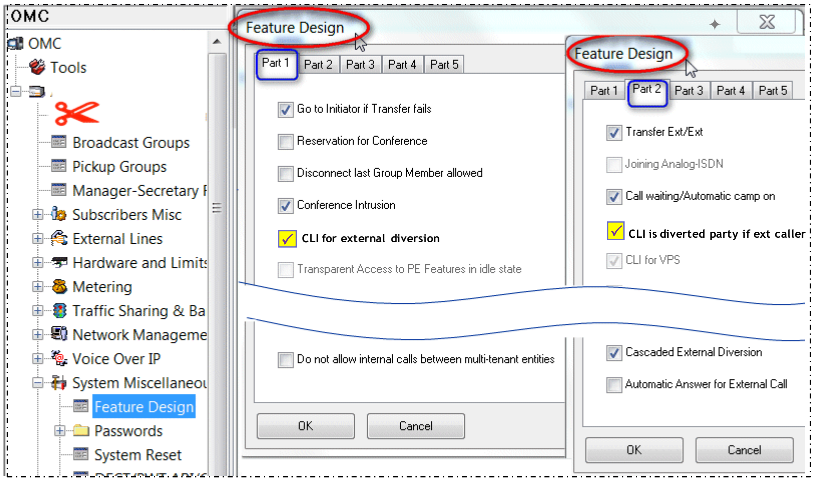
This example defines the access to the system ARS table for phone numbers dialed that start with digit 0. The "Drop" attribute also indicates that the initial 0 of the number is dropped before it is passed to the ARS Prefix table.

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.

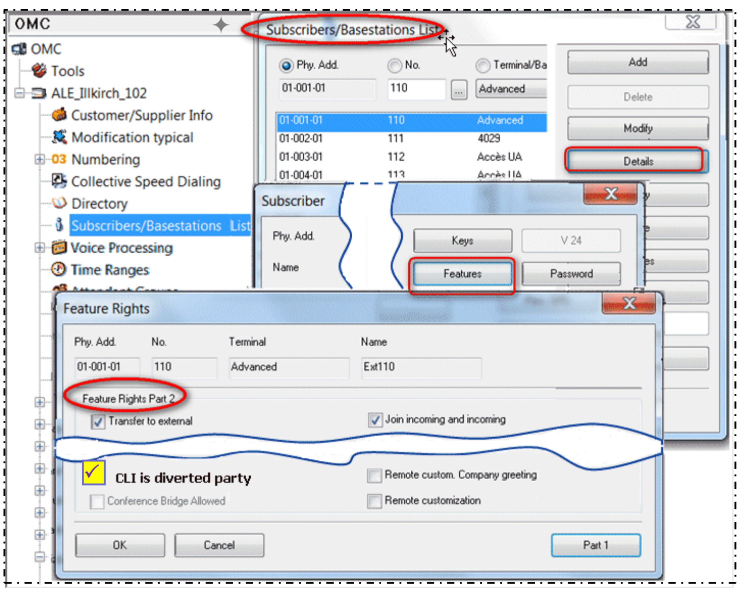
From the tabs "Part 1" and "Part 2" of menu System Misc -> Feature Design, verify the parameters:

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



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 = True**



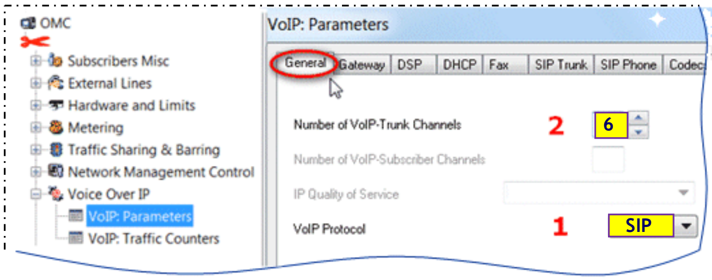
2.6 Traffic Sharing and Barring (reminder)

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 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 a SIP trunk in the system:



Note

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

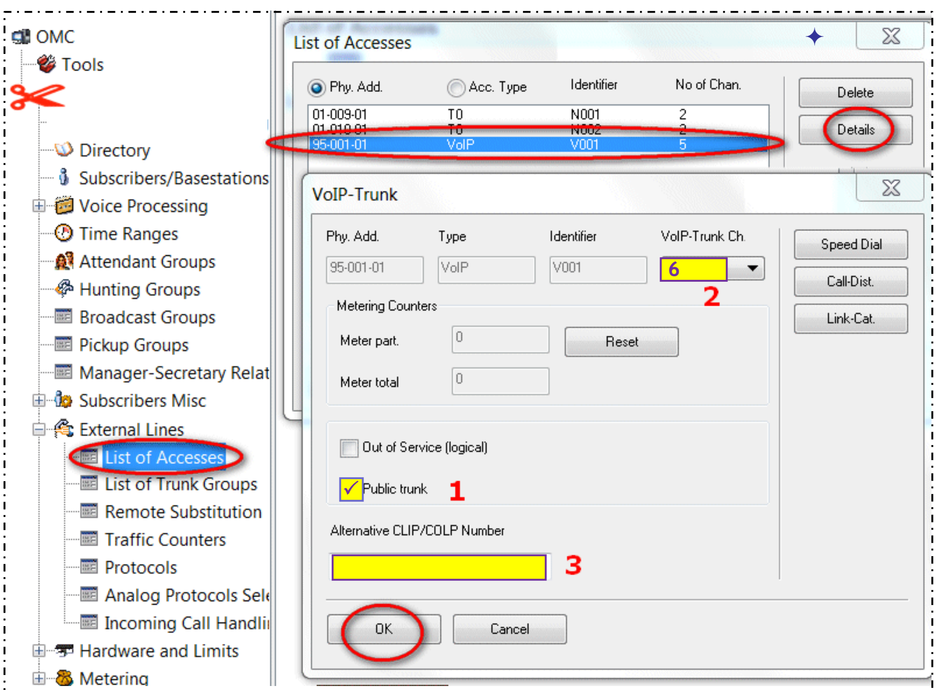
- ① VoIP Protocol : ☞● **VoIPgen_Protocol = SIP**

② Number of VoIP-Trunk Channels :

☞● **VoIPgen_Trunk_Channels_example = 6**

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.



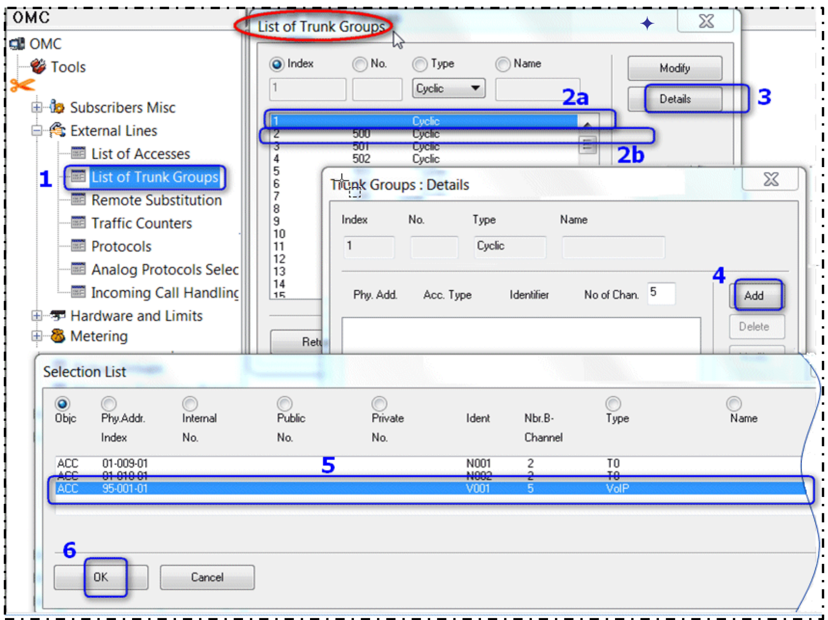
Then, configure the parameters corresponding to this VoIP Access:

- ① "Public trunk" option : ☞● **Access_is_Public = True**
- ② number of channels allocated ("VoIP-Trunk Ch."):
  ☞● **Access_Channels_example = 6**
- 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 also 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).

Note

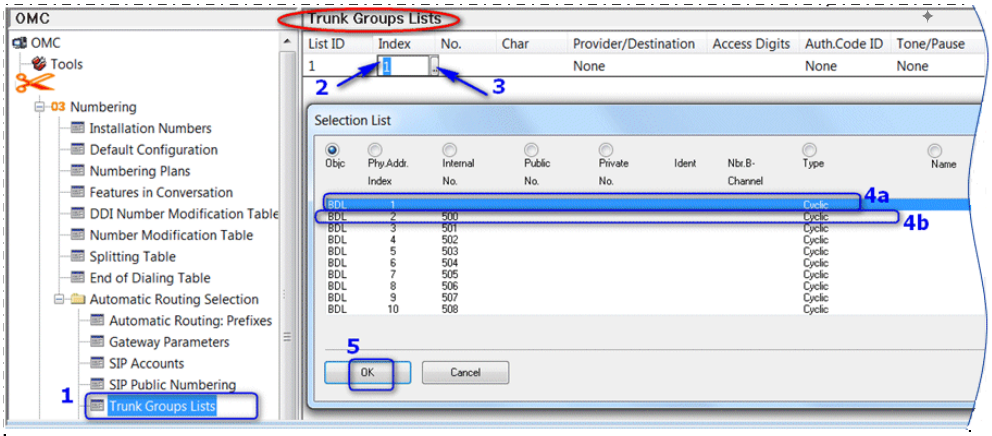
The SIP trunk can be placed freely into one or several Trunk Groups of the system thus permitting to manage a differentiated control of traffic sharing for internal subscribers. The index number selected at step 2a or 2b is relevant for the further configuration of section 3.4.

3.4 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.

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 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)

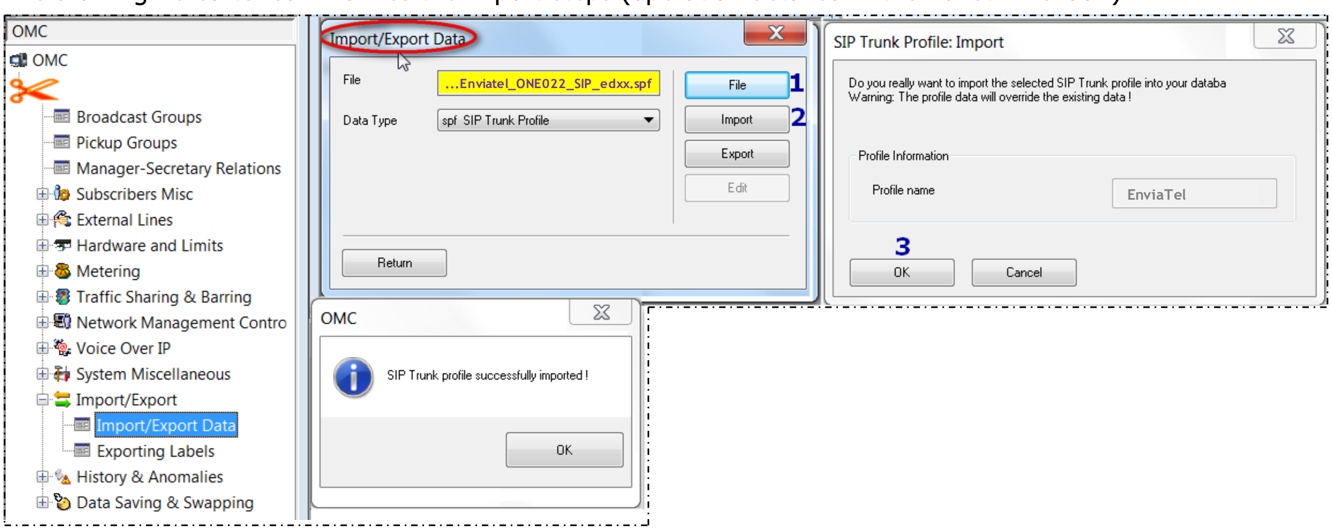
Warning

For proceeding with next configuration steps, you should have on your PC the file "DE_Enviatel_ONE02X_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).

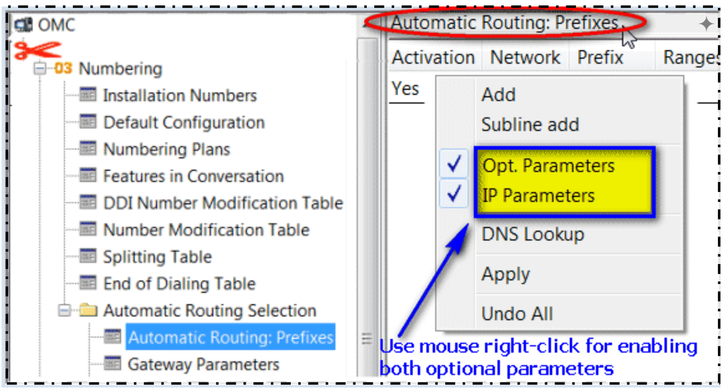
4.2 Complementary Setup

The steps of Ch. 4.2 are needed for completing the OMC configuration not managed by SIP Easy Connect.

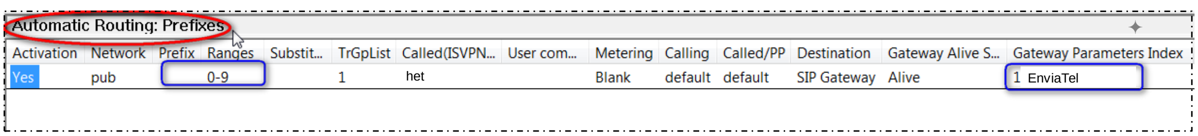
4.2.1 ARS Prefixes

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.

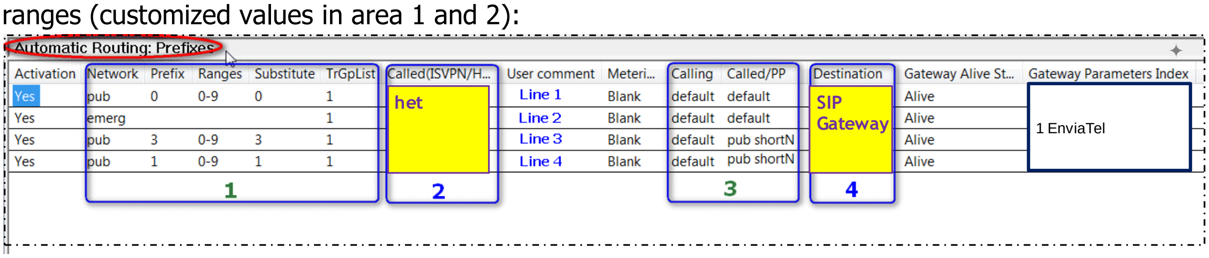


As illustrated in the following picture, you can first insert a route-line covering all type of external calls: use the Add function to create a new line and then, configure the line parameters as indicated.


Automatic Routing: Prefixes

Activation	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 EnviaTel

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	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	0	1	het	Line 1	Blank	default	default	SIP Gateway	Alive	1 EnviaTel
Yes	emerg				1		Line 2	Blank	default	default		Alive	
Yes	pub	3	0-9	3	1		Line 3	Blank	default	pub shortN		Alive	
Yes	pub	1	0-9	1	1		Line 4	Blank	default	pub shortN		Alive	

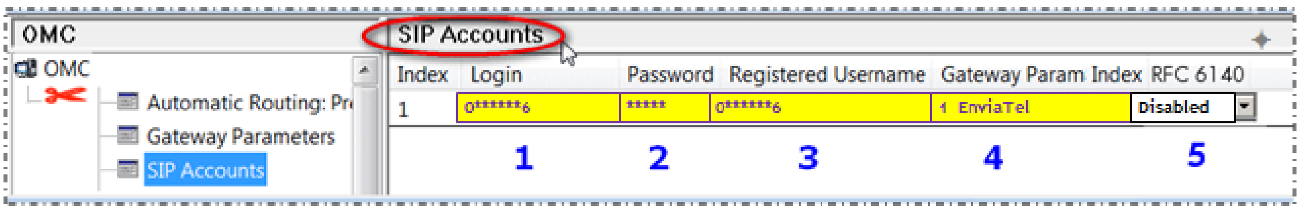
- Line 1: copes with standard phone numbers starting with digit 0 (national / international calls)
- Line 2: copes with all public emergency numbers. The network attribute "emerg" permits the line to point automatically to the system list of emergency numbers. This list is country-dependent and can be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

4.2.2 ARS SIP Accounts

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



- ① “Login” : ☞● **SIPaccnt_Login_example = "0******6"**
- ② “Password” : ☞● **SIPaccnt_Password = "*****"**
- ③ “Registered User.” : ☞● **SIPaccnt_Reg_Username_example = "0******6"**
- ④ the “Gateway Parameters Index” must point to the relevant gateway (i.e. index 1)

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.

4.2.3 System Flags

Warning

For adequate interoperability, this provider needs “RTPpxyPort” Noteworthy set to value “01”, which was introduced in OXO Connect R2.0/038.001 or R2.1/024.004 version (or above) . This Noteworthy MUST be set manually (flag not yet introduced in the SIP profile).

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.**

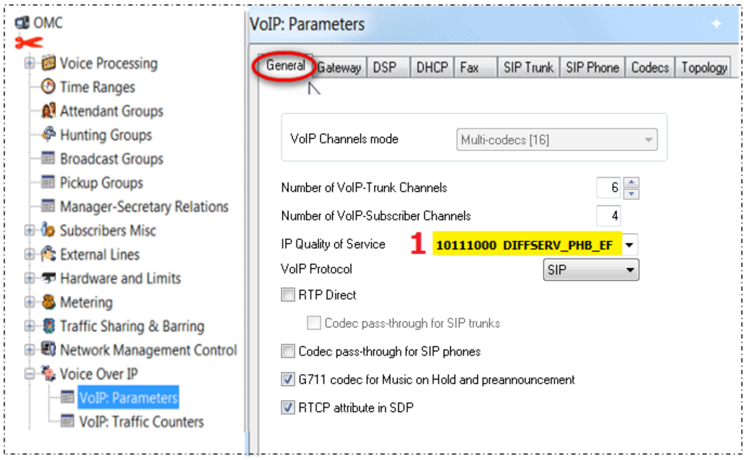
The access to the system flags is made with the OMC menu System Miscellaneous -> Memory Read/Write (Debug Labels or Other Labels). Please refer to the indications and comments given in the configuration abstract Table and apply carefully the required flag changes on OMC.

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

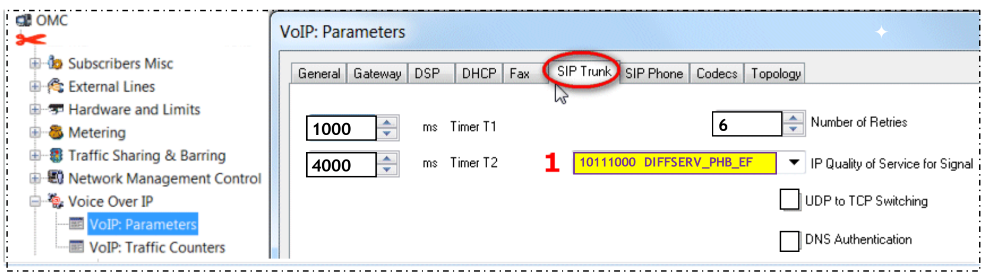


- Adjust the IP Quality of Service of VoIP trunks (RTP flow):

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

4.3.2 VoIP SIP Trunk Tab

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

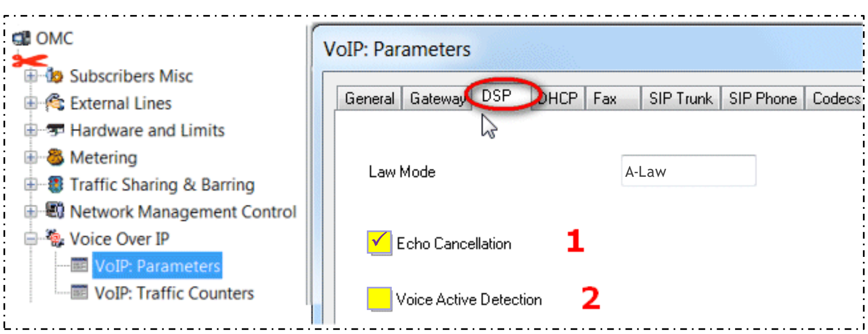


- Adjust the IP Quality of Service of SIP Trunk messages (SIP signaling) :

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

4.3.3 VoIP DSP Tab

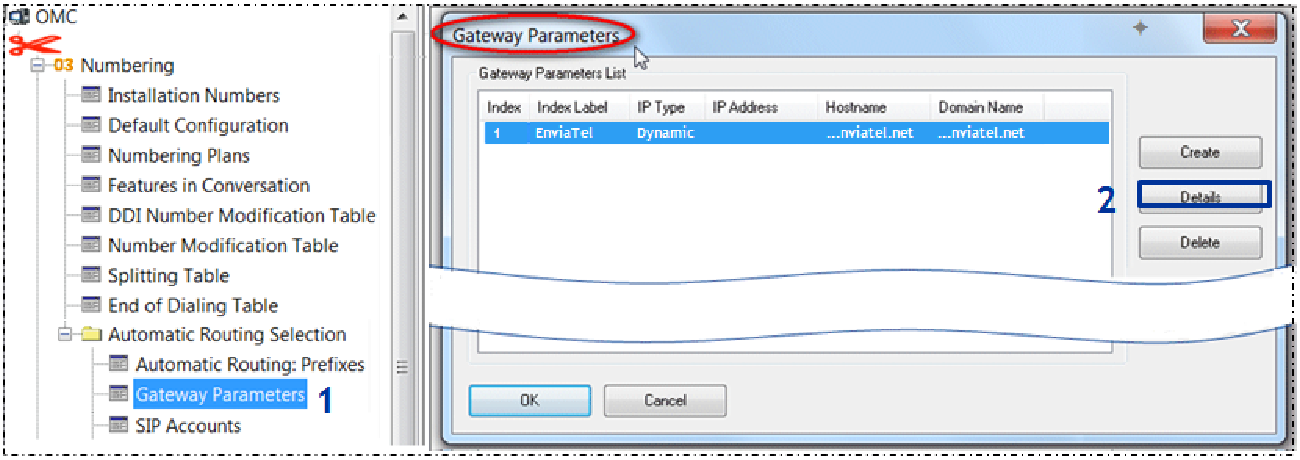
Open the OMC tab via the menu Voice Over IP -> VOIP:Parameters -



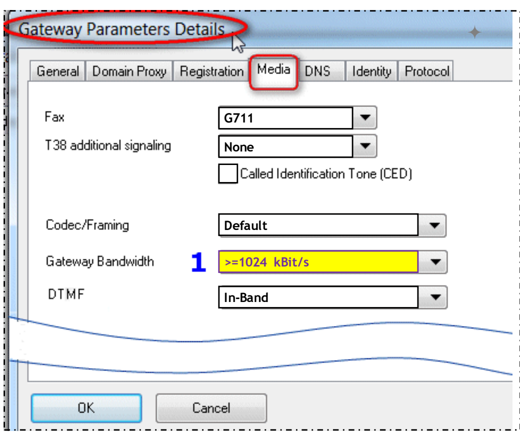
- if necessary, adjust the control of Echo Cancellation and Voice Active Detection (VAD) of VoIP calls:
  - ① "Echo Cancellation" : ☞●✓ **VoIPdsp_DSP_Echo_Cancel = True**
  - ② "Voice Active Detection" : ☞●✓ **VoIPdsp_DSP_VAD = False**

4.3.4 ARS-GW Media Tab

Double-click on menu Numbering -> Automatic Routing Selection -> Gateway Parameters ①. 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".



Warning

This set up for Fax is only relevant if Fax service is supported for the solution (see Ch.1.5.1). **For a Fax machine, it's also mandatory that its port is configured with "Service 1 = Fax 2/3"** in OMC -> Subscribers-BasestationList/Details/Service

Adjust Bandwidth to the site context: ① ☞●✓ **GWmedia_Bwidth_example = >=1024 kBit/s**

5 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 ONE02X	VALUE	SEC	REMARK
/Numbering/...			
NP_Instal_Number_example	"8***0"		Value given as example
NP_International_Prefix	"00"		
NP_International_Code	"49"		
NP_Intercity_Prefix	"0"		
NP_Intercity_Code_example	"351"		Value given as example
NP_Recall_Prefix	"0"		
NP_System_Alt_CLIP_example	""		
NP_DDI_Range_example	"60 69"		Value given as example
/External Lines/ ListOf Accesses -VoIP			
Access_is_Public	True		
Access_Alt_CLIP_example	""		
Access_Channels_example	6		Value given as example
/ Misc/Feature Design			
Misc_CLI_Ext_Diversion	True		
Misc_CLI_is_Diverted_Party	True		
/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 02** 00 00"	✓	
Flag_VOIPnwaddr_Line10	"00 00 00 00 00 00 00 00"	✓	
Flag_VipPuNuA	"00"	✓	
Flag_ExtNuFoVoi	"22"	✓	
Flag_MultAnsRei	"01"	✓	
Flag_SimulIpAlt	"00"	✓	
Flag_PrefCodec	"00 00"	✓	
Flag_PrefFramin	"00"	✓	
Flag_FaxPasCd	"01 FF"	✓	
Flag_SIPInDspNm	"03"	✓	
Flag_SIPOgDspNm	"03"	✓	
Flag_INVwSDPtrk	"00"	✓	
Flag_SIPdtmfInB	"00"	✓	
Flag_SuprAlerTo	"00"	✓	
Flag_RTPpxyPort	"01"		

Table 2 (Voice Over IP)

CONFIG OXO ONE02X	VALUE	SEC	REMARK
/VoIP/VoIP Parms/General			
VoIPgen_Trunk_Channels_example	6		Value given as example
VoIPgen_IP_QoS_example	10111000 DIFFSERV_PHB_EF	✓	Value given as example
VoIPgen_Protocol	SIP	✓	
VoIPgen_RTP_Direct	False	✓	
VoIPgen_Trunk_Codec_Passthru	False	✓	
VoIPgen_Phone_Codec_Passthru	False	✓	
VoIPgen_G711_MOH	True	✓	
VoIPgen_RTCP_Attribute	True	✓	
/VoIP/VoIP Parms/Gateway			
VoIPgw_SIPSourcePort	"5060"	✓	
/VoIP/VoIP Parms/DSP			
VoIPdsp_DSP_Echo_Cancel	True	✓	
VoIPdsp_DSP_VAD	False	✓	
/VoIP/VoIP Parms/Fax			
VoIPfax_T38_UDP_Redundancy	"1"	✓	
VoIPfax_T38_Fax_Framing	"0"	✓	
VoIPfax_T38_ECM	False	✓	
/VoIP/VoIP Parms/SIP Trunk			
VoIPsiptrk_QoS_example	10111000 DIFFSERV_PHB_EF	✓	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	✓	
/VoIP/VoIP Parms/Codecs			
VoIPcodec_Def_CodecFraming	20	✓	
VoIPcodec_Def_CodecList	G711.a G711.µ	✓	
VoIPcodec_DTMF_Payload	"106"	✓	
VoIPcodec_G722.2_Payload	"117"	✓	
/VoIP/VoIP Parms/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

Table 3 (ARS)

CONFIG OXO ONE02X	VALUE	SEC	REMARK
/Numbering/ARS/ARS Prefixes			
ARS_Called_Mode	het		
ARS_Destination	SIP Gateway		
/Numbering/ARS/SIP Accounts			
SIPaccnt_Login_example	"0******6"		Value masked partially
SIPaccnt_Password	"*****"		
SIPaccnt_Reg_Username_example	"0******6"		Value masked partially
SIPaccnt_RFC6140_Enabled	False		
/Numbering/ARS/Public Numbering			
SIPnum_Out_Calling_Format	National	✓	
SIPnum_Out_Calling_Prefix	""	✓	
SIPnum_Out_Called_Format	National/International	✓	
SIPnum_Out_Called_Prefix	""	✓	
SIPnum_Out_Called_Short_Prefix	""	✓	
SIPnum_Inc_Calling_Format	National	✓	
SIPnum_Inc_Calling_Prefix	""	✓	
SIPnum_Inc_Called_Format	National	✓	
SIPnum_Inc_Called_Prefix	""	✓	
SIPnum_Alt_CLIP_example	""		

Table 4 (GW Parameters)

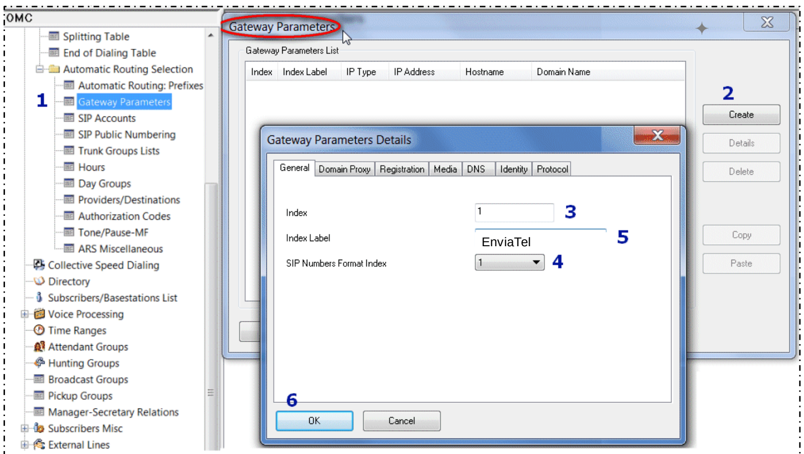
CONFIG OXO ONE02X	VALUE	SEC	REMARK
/Numbering/ARS/GW Parms/Domain			
GWdom_IP_Type	Dynamic	✓	
GWdom_IP_Address	(N/A)	✓	Dynamic value
GWdom_Def_Transport	UDP	✓	
GWdom_Target_Domain	"sip.enviatel.net"	✓	
GWdom_Local_Domain_Name	"sip.enviatel.net"	✓	
GWdom_Realm	""	✓	
GWdom_Remote_SIP_Port	(Dynamic)	✓	Dynamic value
GWdom_Outb_Proxy	""	✓	
/Numbering/ARS/GW Parms/Registration			
GWreg_Reg_Requested	True	✓	
GWreg_Check_Before_Req	True	✓	
GWreg_Reg_Name	"sip.enviatel.net"	✓	
GWreg_Reg_IP_Address	(N/A)	✓	Dynamic value
GWreg_Reg_Port	(Dynamic)	✓	Dynamic value
GWreg_Reg_Expire_Time	"330"	✓	
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	✓	
GWreg_RFC3327_Enabled	False	✓	

/Numbering/ARS/GW Parms/Media			
GWmedia_Fax_Mode	G711	✓	
GWmedia_T38_Add_Signal	"None"	✓	
GWmedia_T38_CED_Tone	False	✓	
GWmedia_Codec_Framing	Default	✓	
GWmedia_Bwidth_example	>=1024 kBit/s	✓	Value given as example
GWmedia_DTMF_Mode	In-Band	✓	
/Numbering/ARS/GW Parms/DNS			
GWdns_DNS_Mode	DNSSRV	✓	
GWdns_Prim_DNS	"193.98.112.238"	✓	
GWdns_Sec_DNS	"193.98.115.214"	✓	
/Numbering/ARS/GW Parms/Identity			
GWident_RFC3325	True	✓	
GWident_HistInfo_DivHeader	History-Info	✓	
GWident_Inc_CLI_Headers	From PAI PPI 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	✓	
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	✓	
/Numbering/ARS/GW Parms/Protocol			
GWprot_SessTimer_Time	"720"	✓	
GWprot_PEM_Enabled	True	✓	
GWprot_UPDATE_Enabled	True	✓	
GWprot_SNAT_Enabled	False	✓	
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	""	✓	

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. "EnviaTel") 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 **appropriate sections of table 4**. 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. 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.5 and refer to columns "VALUE" and "SEC" of the tables.

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