

Architecture guide Business Talk IP IPBX Avaya IP Office

Versions addressed in this guide: Orange 11.0 SP1 version (11.0.0.1.24 Build 2), Avaya 10.1 SP1 (10.1.0.1.0 Build 3) and Orange 10.0 SP3 + CP (10.0.0.3.12 build 2)

Information included in this document is dedicated to customer equipment (IPBX, TOIP ecosystems) connection to Business Talk IP service: it shall not be used for other goals or in another context.

Document Version

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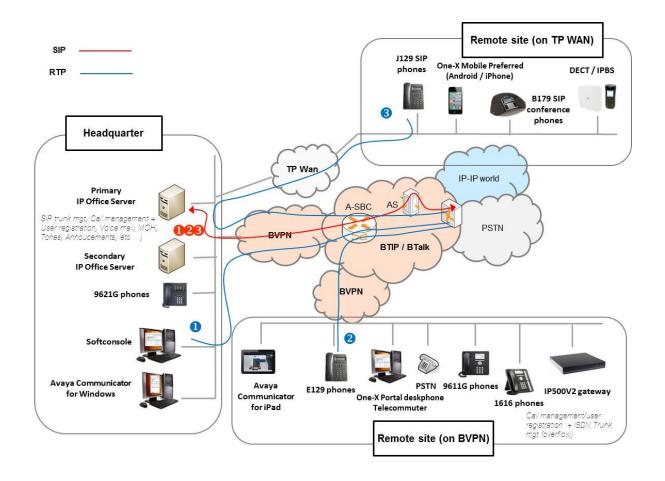
Goal of this document

The aim of this document is to list technical requirements to ensure the interoperability between Avaya IP Office IPBX with OBS service Business Talk IP SIP, hereafter so-called "service".



1 ARCHITECTURE OVERVIEW

1.1 Architecture without "Customer SBC"



Notes:

- in the diagram above, the SIP and proprietary internal flows are hidden.
- call flows will be the similar with or without IPO Call Server redundancy.

In this architecture

- all 'SIP trunking' signalling flows are carried by the IP Office server and routed on the main BVPN connection.
- Media flows are direct between endpoints and the Business Talk/BTIP but IP routing differs from one site to another:
 - o For the Head Quarter site, media flows are just routed on the main BVPN connection.
 - For Remote sites on BVPN, media flows are just routed on the local BVPN connection (= distributed architecture).
 - For Remote sites on Third Party WAN, media flows are routed through the Head Quarter (but not through the IPBX) and use the main BVPN connection (= centralized architecture, cf sizing below).



Call scenario	nb of voice channels/media resources used		
	IPBX	WAN router*	BTIP
1 offnet call from/to the headquarter (HQ)	1 in HQ	1 in HQ	1 in HQ
1 offnet call from/to a remote site (RS) on BVPN	0 in HQ	<mark>0</mark> in HQ	0 in HQ
	1 in RS	1 in RS	1 in RS
1 offnet call from/to a remote site (RS) on TP Wan	0 in HQ 1 in RS	<mark>1</mark> in HQ BVPN 1 in HQ TP Wan 1 in RS TP Wan	0 in HQ 1 in RS
1 offnet call from/to a remote site with put on hold	1 in HQ	1 in HQ	0 in HQ
	1 in RS	1 in RS	1 in RS
1 offnet call from/to a remote site after transfer/forward to BTIP	0 in HQ	0 in HQ	0 in HQ
	0 in RS	0 in RS	2 in RS
1 forced onnet call from Headquarter to a remote site (= through Business Talk IP infrastructure)	2 in HQ	1 in HQ	0 in HQ
	2 in RS	1 in RS	0 in RS

^{*}On the WAN router, 1 voice channel= 80Kb/s

Resiliency consideration

Secondary IP office server can be located on the same site as the primary IP Office server or on a remote site.

All users are registered initially to a nominal central server. Then in case of failure of the primary server:

- HQ users register to the backup server located near the nominal server or distant from the nominal server
- o Some remote users may register to their local GW if it is available
- Some remote users may register to the GW located on another remote site or on the backup server

Codecs consideration

Only G711A 20 ms codec is supported.

G729A codec is not certified.

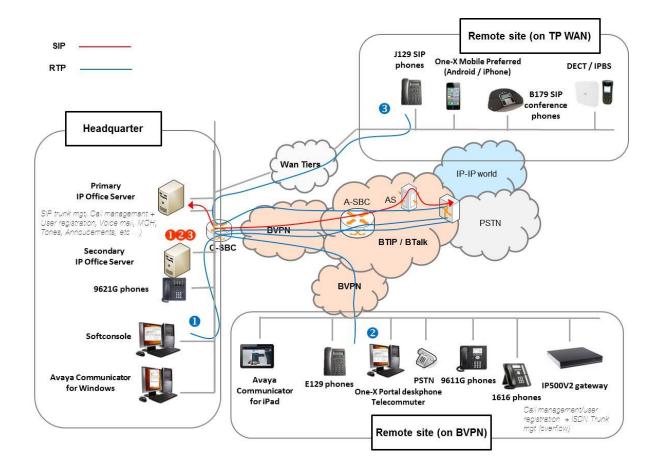
G722 codec may be used for internal calls.

Sizing approach

There is no specific sizing approach to be considered with IP Office solution. The RTP flow is direct between Avaya phones and Orange a-SBC.



1.2 Architecture with "Customer SBC"



Notes:

- in the diagram above, the SIP and proprietary internal flows are hidden.
- call flows will be similar with or without IP Office server redundancy.

In this architecture

- Depending on the SBC equipment we will either provide the same guidelines than the PBX ones or apply a specific "customer SBC process" to qualify the target architecture.
- Both 'SIP trunking' and RTP media flows between endpoints and the Business Talk/BTIP are anchored by the "customer SBC":
 - o for the Headquarter site, media flows are routed through the SBC and the main BVPN connection
 - o for Remote Sites either on BVPN or Third Party WAN, media flows transit through the Headquarter SBC and use the central BVPN connection (= centralized architecture, cf sizing below).



Warning: with a "customer SBC" architecture, site access capacity has to be sized adequately on the Headquarter. Here below a table with a few sizing elements:

Call scenario	nb of voice channels/media resources used			
Guil Goolland	IPBX	WAN router*	BTIP	
1 offnet call from/to the headquarter (HQ)	1 in HQ	1 in HQ	1 in HQ	
1 offnet call from/to a remote site (RS) on BVPN	0 in HQ	2 in HQ	0 in HQ	
	1 in RS	1 in RS	1 in RS	
1 offnet call from/to a remote site (RS) on TP Wan	0 in HQ 1 in RS	<mark>1</mark> in HQ BVPN 1 in HQ TP Wan 1 in RS TP Wan	0 in HQ 1 in RS	
1 offnet call from/to a remote site with put on hold	1 in HQ	<mark>3</mark> in HQ	0 in HQ	
	1 in RS	1 in RS	1 in RS	
1 offnet call from/to a remote site after transfer/forward to BTIP	0 in HQ	0 in HQ*/ <mark>3</mark> in HQ**	0 in HQ	
	0 in RS	0 in RS	2 in RS	
1 forced onnet call from Headquarter to a remote site (= through Business Talk IP infrastructure)	2 in HQ	<mark>3</mark> in HQ	0 in HQ	
	2 in RS	1 in RS	0 in RS	

^{*}on the WAN router, 1 voice channel = 80Kb/s

^{**}if media release is activated on the cSBC

^{***}if media release is not activated on the cSBC



2 PARAMETERS to be PROVIDED by CUSTOMER to ACCESS BTIP service

IP addresses marked in red have to be indicated by the Customer, depending on Customer architecture scenario

Headquarter (HQ)	Level of Service		r IP addresses y the service
architecture without Customer SBC		Nominal	Backup
1 IP Office Server (Call Server)	No redundancy 1 single call server Note: media gateway optional - used for local PSTN access only	IPO IP@	N/A
REDUNDANCY			
2 IP Office call servers (active/active) - 1 NUMBERING PLAN 2 IP Office call servers (active/active), nominal/backup for a group of users (1 numbering plan). The IP office servers can be hosted by the same site or by 2 different physical sites. Each IP Office server (IPO S1 and IPO S2) has its own SIP trunk but IPO S2 is only used as a backup. Both call servers are independent but considered as being part of one HQ. - Nominal mode: All users register with IPO S1 - Backup mode: All users re-register with IPO S2	User registration redundancy (IP phones only) Rerouting at Orange SBC level	IPO1 IP@	IPO2 IP@
2 IP Office servers (active/active) - 2 NUMBERING PLANS 2 IP Office servers (active/active) hosted by 2 different physical sites. Each IP Office server manages a range of users (2 numbering plans). Each IP Office server (IPO S1 and IPO S2) has its own SIP trunk and each manages its own group of users in nominal mode.	For IPO Server HQ1 User registration redundancy (IP phones only) Rerouting at Orange AS level	IPO HQ1 IP€	PO HQ2 IP@
- Nominal mode: All HQ1 users register with IPO Server HQ1 All HQ2 users register with IPO Server HQ2 - Backup mode: In case of IPO Server HQ1 crash, all HQ1 users re-register onto IPO Server HQ2 In case of IPO Server HQ2 crash, all HQ2 users re-register with IPO Server HQ1 warnings: - Both HQ accesses capacity to be sized adequately	For IPO Server HQ2 User registration redundancy (IP phones only) Rerouting at Orange AS level	IPO HQ2 IP@	PO HQ2 IP@
Remote Site (RS) architecture without Customer SBC	Level of Service	Nominal	Backup
Remote site with media gateway (500v2)	Local site survivability and	N/A	N/A



	trunk redundancy via PSTN only		
Remote site without media gateway	No survivability, no trunk redundancy	N/A	N/A

Architecture with Customer SBC	Level of service	@IP used by the service		
1 Customer SBC	No redundancy	cSE	BC @IP	
2 Customer SBC Nominal / Backup mode	Local redundancy: both SBC are hosted on the same site OR Geographical redundancy both SBC are hosted on 2 different sites	cSBC1 @IP	cSBC2 @IP	
2 Customer SBC Load Sharing	- Local redundancy: both SBC are hosted on the same site OR - Geographical redundancy both SBC are hosted on 2 different sites		C1 @IP C2 @IP	
2 Customer SBC HA mode	- Local redundancy: both SBC are hosted on the same site OR - Geographical redundancy both SBC are hosted on 2 different sites warning: Link level 2 between SBC with max delay 50ms required for georedundancy	cSBC	VIP @IP	



3 CERTIFIED SOFTWARE and HARDWARE versions

3.1 Avaya IP Office IPBX

AVAYA IP OF	AVAYA IP OFFICE IPBX – software versions							
Reference product	Software version	Certification ✓: Certified NS: No supported	Certified "Loads"	Restrictions				
AVAYA IP Office Select edition	Orange 11.0 SP1	✓	This load was built by Avaya for Orange. The exact load version is 11.0.0.1.24 build 2.	Due to an Avaya blocking defect, Business Talk service is not supported with this version. Only BTIP service is supported with this version.				
	Avaya 10.1 SP1	✓	10.1.0.1.0 build 3					
	Orange 10.0 SP3 + CP	✓	The critical patch (CP) for this load was built by Avaya for Orange. The exact load+CP version is 10.0.0.3.12 build 2.					

3.2 Avaya IP Office endpoints and applications

AVAYA IP OFFICE IPBX - Endpoints and applications						
	Reference product	Software version NA: not applicable	Certification ✓: Certified NS: No supported	IP Office version	Comments	
		2.4.3.5	✓	11.0 SP1		
	B179 SIP conference phones	0.4.4.5	√	10.1 SP1		
		2.4.1.5	V	10.0 SP3		
		1.25.2.26	✓	11.0 SP1		
	E129 SIP phones	1.25.2.26	✓	10.1 SP1		
		1.25.2.34	✓	10.0 SP3		
		3.0.0.0.20	✓	11.0 SP1		
	J129 SIP phones	1.1.0.0.15	✓	10.1 SP1		
Avaya		1.0.0.0.43	✓	10.0 SP3		
endpoints	J139/J169/J179 SIP phones	3.0.0.0.20	✓	11.0 SP1		
		NA	NS	10.1 SP1		
		NA	NS	10.0 SP3		
	1603, 1608, 1616 IP phones	1.350B	✓	11.0 SP1		
		1.350B	✓	10.1 SP1		
		1.390A	✓	10.0 SP3		
	9608, 9611G, 9621G, 9641G, 9641GS IP phones	6.6.6.04	✓	11.0 SP1		
		6.6.5.06	✓	10.1 SP1		
	304100 II priories	6.6.4.01	✓	10.0 SP3		
•		11.0.0.1.0 build 1	✓	11.0 SP1		
Avaya	IP Office Softconsole	10.1.0.1.0 build 5	✓	10.1 SP1		
Attendant		10.0.0.3.0 build 1	✓	10.0 SP3		
		2.1.4.0	✓	11.0 SP1		
	Avaya Communicator for	2.1.4.0.274	✓	10.1 SP1		
Avaya	Windows	2.1.3.0.237	✓	10.0 SP3		
Softphone			✓	11.0 SP1		
·	Avaya Communicator for iPad	2.0.6	✓	10.1 SP1		
			✓	10.0 SP3		



AVAYA IF	OFFICE IPBX - Endpo	ints and applicat			
F	Reference product	Software version NA: not applicable	Certification ✓: Certified NS: No supported	IP Office version	Comments
	Avaya 3730,3735 DECT	2.1.4	✓	11.0 SP1	
	phones	NA	NS	10.1 SP1	
	priories	NA	NS	10.0 SP3	
	Avaya 3720,3725,3740,3745		✓	11.0 SP1	
	DECT phones	4.3.32	✓	10.1 SP1	
	BEOT phones		✓	10.0 SP3	
Avaya		10.0.7	✓	11.0 SP1	
DECT	DECT R4 – IPBS1-IPBS2	10.0.5	✓	10.1 SP1	
DLCT		7.2.28	✓	10.0 SP3	
		4.5.1	✓	11.0 SP1	
	DECT R4 – AIWS2	4.5.1	✓	10.1 SP1	
		3.70A	✓	10.0 SP3	
			✓	11.0 SP1	
	DECT R4 – AIWS1	2.73	✓	10.1 SP1	
			✓	10.0 SP3	
Avere IDBV		11.0.0.1.0 build 8	✓	11.0 SP1	
Avaya IPBX	IP Office UC module	10.1.0.1.0 build 3	✓	10.1 SP1	
components		10.0.0.3.12 build 2	✓	10.0 SP3	
Avaya		11.0.0.1.24 build 2	✓	11.0 SP1	
Gateway	IP500v2	10.1.0.1.0 build 3	✓	10.1 SP1	
Galeway		10.0.0.3.12 build 2	✓	10.0 SP3	
		11.0.0.1.0 build 3	✓	11.0 SP1	
Voice Mail	Avaya VoiceMail Pro	10.1.0.1.0 build 6	✓	10.1 SP1	
		10.0.0.3.0 build 5	✓	10.0 SP3	
		11.0.0.1.0 build 38	✓	11.0 SP1	
	One-X Portal	10.1.120.30	✓	10.1 SP1	
Avaya		10.0.0.3.0 build 9	✓	10.0 SP3	
Unified	On a Malalla Danfarra I	10.0.0.5.220	✓	11.0 SP1	
Communicat	One-X Mobile Preferred	10.0.0.5.220	✓	10.1 SP1	
ions and Mobility	Edition for Android	10.0.0.3.201	✓	10.0 SP3	
	0 //// 5 /	4.1.12.769	✓	11.0 SP1	
	One-X Mobile Preferred	4.1.8.763	✓	10.1 SP1	
	Edition for iOS	4.1.8.763	✓	10.0 SP3	
Third-party				11.0 SP1	
endpoints	ISI COM Internet	7 4/0 4	./	10.1 SP1	Contact Cont
.	ISI-COM Interact	7.x/8.x	✓		Contact Center
applications				10.0 SP3	



4 SIP TRUNKING CONFIGURATION CHECKLIST FOR Orange 11.0 SP1 version (11.0.0.1.24 Build 2)

The checklist below presents all the steps of configuration required for interoperability between BTIP/BT and Orange IP Office IPBX 11.0 SP1 version (11.0.0.1.24 Build 2).

Trunk configuration - IP Office Server Edition

Access type: IP Office Web Manager page.

Platform	Configuration place	Configuration details				
Services						
Primary IPO	System	running services: IP Office Voicemail One-X Portal Web Manager Web License Manager Web Collaboration WebRTC Gateway Web Client				

Access type: IP Office Manager application.

Platform	Menu	Object	Tab	Parameter	Value				
	System configuration – Locale configuration								
Every platform in the solution ¹	System	-	System	Locale	France2 (French)				
		System c	onfiguration - DS	SCP configuration					
_				DSCP (Hex) / DSCP	B8 / 46				
Every platform in the	System	-	LAN1 -> VoIP	Video DSCP (Hex) / Video DSCP	88 / 34				
solution				SIG DSCP (Hex) / SIG DSCP	B8 / 46				
			DHCP configurat	tion offer					
				Start address	Start IP address				
Primary	System		LAN1 ->	Subnet Mask	Subnet Mask				
IPO	System -	-	DHCP Poll	DHCP Poll	Default Router	Router IP address			
				Pool size	DHCP pool size				
	Codec configuration								
Every	System	-	Telephony ->	Companding Law	A-Law				

¹ Every platform in the solution: primary IPO, secondary IPO (if used), expansion units (if used)

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platform in the	Line		Telephony	High Quality Conferencing	Checked
solution				Ignore DTMF Mismatch For Phones	Checked
			VolP	RFC2833 Default Payload	101
				Default Codec Selection -> Selected	G.722 64K G.711 ALAW 64K
		Call Admiss	sion Control & Lo	cation configuration ²	
				Location Name	Ex:RS140
				Subnet Address	6.201.40.0
				Subnet Mask	255.255.255.0
	Location Location		Parent Location for CAC	<none></none>	
Solution level		Location	Call Admission Control -> Total Maximum Calls	99	
				Call Admission Control -> External Maximum Calls	99
				Call Admission Control -> Internal Maximum Calls	99
Every platform in the solution	System	-	System	Location	Ex:HQ313
			Fallback config	uration ³	
Primary IPO	Location	Location	Location	Fallback System	Local GW's IP address
			SCN lines config	guration	
				Outgoing Group ID	99998
				Transport Type	Proprietary
Primary	Line	IP Office	Line	Networking Level	SCN
IPO ⁴	Line line ⁵	LINE .	Gateway -> Address	Backup IPO's IP address	
				Gateway -> Location	Location name

² For each physical site (Headquarter and Remote Sites) dedicated location has to be created, mainly for Call Admission Control and emergency calls management. This section provides example values.

³ For each location where local gateway should act as a backup system in case of primary server failure Fallback System should be defined.

⁴ Repeat the steps on primary IPO to create separate SCN line for each local gateway in the solution.

⁵ SCN Line to secondary server



				CON Decilier	
				SCN Resiliency Options -> Supports Resiliency	Checked
				- Backs up my IP Phones	Checked
				- Backs up my Hunt Groups	Checked
				- Backs up my Voicemail	Checked
				- Backs up my IP Dect Phones	Checked
				- Backs up my One- x Portal	Checked
			VoIP settings	Allow Direct Media Path	Checked
				Outgoing Group ID	99901 - 99930
				Transport Type	Proprietary
				Networking Level	SCN
	IP Office Line ⁶			Gateway -> Address	Local GW's IP address
				Gateway -> Location	Location name
		Line	SCN Resiliency Options -> Supports Resiliency	Checked	
				- Backs up my IP Phones	Unchecked
				- Back up my Hunt Groups	Unchecked
				- Back up my IP Dect Phones	Unchecked
			VoIP settings	Allow Direct Media Path	Checked
				Outgoing Group ID	99999
				Transport Type	Proprietary
			Networking Level	SCN	
Secondary IPO	IPO Line	IP Office line ⁸	Line	Gateway -> Address	Primary IPO's IP address
(if used) ⁷		iirie ~		Gateway -> Location	Location name
				SCN Resiliency Options -> Supports Resiliency	Checked

⁶ SCN Line to expansion gateway

 $^{^{7}}$ Repeat the steps on secondary IPO (if used) to create separate SCN line for each local gateway in the solution.

⁸ SCN Line to Primary server



				- Backs up my IP Phones	Checked
				- Backs up my Hunt Groups	Checked
				- Back up my Voicemail	Checked
				- Back up my IP Dect Phones	Checked
				- Back up my one-X Portal	Checked
			VoIP settings	Allow Direct Media Path	Checked
				Outgoing Group ID	99901 - 99930
				Transport Type	Proprietary
	IP Office Line ⁹			Networking Level	SCN
		IP Office	Line	Gateway -> Address	Local GW's IP address
				Gateway -> Location	Location name
				SCN Resiliency Options -> Supports Resiliency	Unchecked
			VoIP settings	Allow Direct Media Path	Checked
				Outgoing Group ID	99999
				Transport Type	Proprietary
			Line	Networking Level	SCN
Expansion Gateway	Line	IP Office line ¹¹		Gateway -> Address	Primary IPO's IP address
			Gateway -> Location	Location name	
			SCN Resiliency Options -> Supports Resiliency	Checked	
			VoIP Settings	Allow Direct Media Path	Checked

⁹ SCN line to expansion gateway

¹⁰ Redundant architecture only

¹¹ SCN line to Primary server



				Outgoing Group ID	99998
				Transport Type	Proprietary
				Networking Level	SCN
		IP Office Line ¹²	Line	Gateway -> Address	Backup IPO's IP address
				Gateway -> Location	Location name
				SCN Resiliency Options -> Supports Resiliency	Unchecked
			VoIP Settings	Allow Direct Media Path	Checked
SCN lines configuration				ocal PSTN access	
Expansion	Expansion	PRI 30 (Universal) ¹³	-	Incoming Group ID	3
Gateway	Line			Outgoing Group ID	3
SIP Trunks confi				- Global settings	
				SIP Trunks Enable	Checked
				SIP Registrar Enable	Checked
Primary IPO	System	-	LAN1 -> VoIP	Media Connection Preservation	Enabled
				Inhibit Off-Switch Forward/Transfer	Unchecked
				SIP Trunks Enable	Checked
Secondary				SIP Registrar Enable	Checked
IPO (if used)	System	-	LAN1 -> VoIP	Media Connection Preservation	Enabled
				Inhibit Off-Switch Forward/Transfer	Unchecked
		SIP 1	Trunks configurat	ion – SIP line	
				Line Number	10
Primary IPO	Line	SIP Line	SIP Line	Local Domain Name	Primary IPO's IP address
0			LAN1 -> VoIP	Location	Cloud

¹² SCN line to secondary server

 $^{^{\}rm 13}$ Line type depends on line type attached to Expansion Gateway



Prefix 0 National Prefix 00 Country Code 33 International Prefix 000 In service Checked Check OOS Checked Session Timers -> Refresh Nethod Session Timers -> Timer (seconds) Redirect and Transfer -> Outgoing Supervised REFER Redirect and Transfer -> Outgoing Supervised REFER ITSP Proxy Address Layer 4 Protocol UDP Network Configuration -> Use Network Topology Info Send Port 5060 Listen Port 5060 Incoming Group 10 Outgoing Group 10 Max Sessions Default=10 Range 1 - 250 Local URI -> Display Use Internal Data Local URI -> Display Use Internal Data Local URI -> Display Use Internal Data Local URI -> Field meaning -> Call Petails Call Details			0
Country Code 33 International Prefix O00 In service		National Profiv	i l
International Prefix In service Checked Check OOS Checked Session Timers -> Refiresh Nethod Session Timers -> Timer (seconds) Redirect and Transfer -> Incoming Supervised REFER Redirect and Transfer -> Outgoing Supervised REFER ITSP Proxy Address Layer 4 Protocol UDP Network Configuration -> Use Network Topology Info Send Port Send Port Send Port Incoming Group Incoming Group Outgoing Group Incoming Group Outgoing Group Incoming Group Outgoing Group UDP Max Sessions Default=10 Range 1 - 250 Local URI -> Display Use Internal Data Local URI -> Content Local URI -> Field meaning -> Coller		TVALIONAL LITERA	00
In service Checked Check OOS Checked Session Timers -> Refresh Nethod Session Timers -> Timer (seconds) Redirect and Transfer -> Incoming Supervised REFER Redirect and Transfer -> Outgoing Supervised REFER Redirect and Transfer -> Never ITSP Proxy Address primary SBC's IP address Layer 4 Protocol UDP Network Configuration -> Use Network Topology Info Send Port 5060 Listen Port 5060 Incoming Group 10 Outgoing Group 10 Max Sessions Default=10 Range 1 - 250 Local URI -> Display Local URI -> Content Local URI -> Contern Local URI -> Field meaning -> Coller		Country Code	33
Check OOS		International Prefix	000
Session Timers -> Refiresh Nethod Session Timers -> Timer (seconds) Redirect and Transfer -> Incoming Supervised REFER Redirect and Transfer -> Outgoing Supervised REFER Redirect and Transfer -> Outgoing Supervised REFER ITSP Proxy Address address Layer 4 Protocol UDP Network Configuration -> Use Network Topology Info Send Port 5060 Listen Port 5060 Incoming Group 10 Outgoing Group 10 Max Sessions Default=10 Range 1 - 250 Local URI -> Display Use Internal Data Local URI -> Contern Contern Coller Coller		In service	Checked
Refresh Nethod Session Timers -> Timer (seconds) Redirect and Transfer -> Incoming Supervised REFER Redirect and Transfer -> Outgoing Supervised REFER Redirect and Transfer -> Outgoing Supervised REFER ITSP Proxy Address Layer 4 Protocol UDP Network Configuration -> Use Network Topology Info Send Port 5060 Listen Port 5060 Listen Port 5060 Incoming Group 10 Outgoing Group 10 Max Sessions Default=10 Range 1 - 250 Local URI -> Display Use Internal Data Local URI -> Content Use Internal Data Local URI -> Content Use Internal Data College College Co		Check OOS	Checked
Timer (seconds) Redirect and Transfer -> Incoming Supervised REFER Redirect and Transfer -> Outgoing Supervised REFER Redirect and Transfer -> Outgoing Supervised REFER ITSP Proxy Address Layer 4 Protocol UDP Network Configuration -> Use Network Topology Info Send Port 5060 Listen Port 5060 Listen Port 5060 Incoming Group 10 Outgoing Group 10 Max Sessions Default=10 Range 1 - 250 Local URI -> Display Use Internal Data Local URI -> Content Use Internal Data Local URI -> Content Use Internal Data		 	Reinvite
Transfer -> Incoming Supervised REFER			14880
Transfer -> Outgoing Supervised REFER ITSP Proxy Address Layer 4 Protocol UDP Network Configuration -> Use Network Topology Info Send Port Listen Port Incoming Group Outgoing Group Max Sessions Default=10 Range 1 - 250 Local URI -> Display Use Internal Data Local URI -> Content Coal URI -> Fleld meaning -> Coller		Transfer -> Incoming	Never
Transport Layer 4 Protocol Network Configuration -> Use Network Topology Info Send Port Listen Port Incoming Group Outgoing Group Max Sessions Local URI -> Display Use Internal Data Local URI -> Fleld meaning -> Coller		Transfer -> Outgoing	Never
Transport Network Configuration -> Use Network Topology Info Send Port Listen Port Incoming Group Outgoing Group Max Sessions Default=10 Range 1 - 250 Local URI -> Display Use Internal Data Local URI -> Fleld meaning -> Coller		ITSP Proxy Address	
Transport Configuration -> Use Network Topology Info		Layer 4 Protocol	UDP
Listen Port 5060 Incoming Group 10 Outgoing Group 10 Max Sessions Default=10 Range 1 - 250 Local URI -> Display Use Internal Data Local URI -> Content Local URI -> Fleld meaning -> Caller	Transport	Configuration -> Use Network	None
Incoming Group Outgoing Group Max Sessions Default=10 Range 1 - 250 Local URI -> Display Use Internal Data Local URI -> Content Local URI -> Fleld meaning -> Caller		Send Port	5060
Outgoing Group Max Sessions Default=10 Range 1 - 250 Local URI -> Display Use Internal Data Local URI -> Content Local URI -> Fleld meaning -> Caller		Listen Port	5060
Max Sessions Default=10 Range 1 - 250 Local URI -> Display Use Internal Data Local URI -> Content Local URI -> Fleld meaning -> Caller		Incoming Group	10
Local URI -> Display Local URI -> Display Local URI -> Use Internal Data Local URI -> Content Local URI -> Fleld meaning -> Caller		Outgoing Group	10
Local URI -> Display Local URI -> Display Local URI -> Content Local URI -> Fleld meaning -> Coller		Max Sessions	
Content Local URI -> Fleld meaning -> Coller		Local URI -> Display	Use Internal Data
meaning -> Callor		 	Use Internal Data
g calls	Call Details	meaning -> Forwarding/Outgoin	Caller
Local URI -> Fleld meaning -> Forwarding/Twinnin g Original Caller		meaning -> Forwarding/Twinnin	Original Caller
Local URI -> Fleld meaning -> Forwarding/Incomin g calls Called		meaning -> Forwarding/Incomin	Called
		Contact-> Display	Use Internal Data



		Contact-> Content	Use Internal Data
		Contact -> Fleld meaning -> Outgoing calls	Caller
		Contact -> Fleld meaning -> Forwarding/Twinnin g	Original Caller
		Contact -> Fleld meaning -> Incoming calls	Called
		Diversion Header	Checked
		Diversion Header -> Display	Use Internal Data
		Diversion Header -> Content	Use Internal Data
		Diversion Header -> Fleld meaning -> Outgoing Calls	None
		Diversion Header -> Fleld meaning -> Forwarding/Twinnin g	Caller
		Diversion Header -> Fleld meaning -> Incoming Calls	None
	VolP	Codec Selection	Custom
		DTMF Support	RFC2833/RFC4733
		Local HOLD Music	Checked
		RE-ivite Supported	Checked
		Allow Direct Media Path	Checked
		Force direct media with phones	Checked
		PRACK/100rel Supported	Checked
		Use + for International	On/Off ¹⁴
		Caller ID from From Header	Checked
	SIP Advanced	Send From in Clear	Checked
	Advanced	Cache Auth Credentials	Unchecked
		Add UUI Header	Checked
		Add UUI Header to	Checked

 $^{^{14}}$ When set to On, outgoing international calls use E.164/International format with a '+' followed by the country code and then the directory number (optional).



			redirected calls	
			Media -> P-Early- Media Support	All
			Media -> Force Early Direct Media	Checked
			Media -> Media Connection Preservation	System
			Media -> Media Indicate HOLD	Checked
			Call Control -> Call Initiation Timeout (s)	18
			Call Control -> Call Queuing Timeout (m)	1
			Call Control -> Service Busy Response	503 – Service Unavailable
			Call Control -> on No User Responding Send	480-Temporarily Unavailable
			Call Control -> Action on CAC Location limit	Allow Voicemail / Reject Call
			Call Control -> Suppress Q.850 Reason Header	Checked
		Engineering	Custom String	SLIC_NO_USER_AV AIL=480
			Line Number	11
			Local Domain Name	Primary IPO's IP address
			Location	Cloud
			Prefix	0
			National Prefix	00
			Country Code	33
	OID I :	OID I :	International Prefix	000
	SIP Line	SIP Line	In service	Checked
			Check OOS	Checked
			Session Timers -> Refresh Nethod	Reinvite
			Session Timers -> Timer (seconds)	14880
			Redirect and Transfer -> Incoming Supervised REFER	Never



	Dadirant and	
	Redirect and Transfer -> Outgoing Supervised REFER	Never
	ITSP Proxy Address	backup SBC's IP address
	Layer 4 Protocol	UDP
Transport	Network Configuration -> Use Network Topology Info	None
	Send Port	5060
	Listen Port	5060
	Incoming Group	11
	Outgoing Group	11
	Max Sessions	Default=10 Range 1 - 250
	Local URI -> Display	Use Internal Data
	Local URI -> Content	Use Internal Data
	Local URI -> Fleld meaning -> Forwarding/Outgoin g calls	Caller
	Local URI -> Fleld meaning -> Forwarding/Twinnin g	Original Caller
Call Details	Local URI -> Fleld meaning -> Forwarding/Incomin g calls	Called
	Contact-> Display	Use Internal Data
	Contact-> Content	Use Internal Data
	Contact -> Fleld meaning -> Outgoing calls	Caller
	Contact -> Fleld meaning -> Forwarding/Twinnin g	Original Caller
	Contact -> Fleld meaning -> Incoming calls	Called
	Diversion Header	Checked
	Diversion Header -> Display	Use Internal Data
	Diversion Header -> Content	Use Internal Data



	Diversion Header ->	
	Fleld meaning -> Outgoing Calls	None
	Diversion Header -> Fleld meaning -> Forwarding/Twinnin g	Caller
	Diversion Header -> Fleld meaning -> Incoming Calls	None
	Codec Selection	Custom
	Codec Selected	G.722 64K G.711 ALAW 64K
	DTMF Support	RFC2833/RFC4733
	Local HOLD Music	Checked
VoIP	RE-ivite Supported	Checked
	Allow Direct Media Path	Checked
	Force direct media with phones	Checked
	PRACK/100rel Supported	Checked
	Use + for International	On/Off ¹⁵
	Caller ID from From Header	Checked
	Send From in Clear	Checked
	Cache Auth Credentials	Unchecked
	Add UUI Header	Checked
	Add UUI Header to redirected calls	Checked
SIP Advanced	Media -> P-Early- Media Support	All
Havariood	Media -> Force Early Direct Media	Checked
	Media -> Media Connection Preservation	System
	Media -> Indicate HOLD	Checked
	Call Control -> Call Initiation Timeout (s)	18
	Call Control -> Call Queuing	1

 $^{^{15}}$ When set to On, outgoing international calls use E.164/International format with a '+' followed by the country code and then the directory number (optional).



				Timeout (m)	
				Call Control -> Service Busy Response	503 – Service Unavailable
				Call Control -> on No User Responding Send	480-Temporarily Unavailable
				Call Control -> Action on CAC Location limit	Allow Voicemail / Reject Call ¹⁶
				Call Control -> Suppress Q.850 Reason Header	Checked
			Engineering	Custom String	SLIC_NO_USER_AV AIL=480
				Line Number	110
				Local Domain Name	Secondary IPO's IP address
Secondary IPO (if used) Line SIP Line		SIP Line	Location	Cloud	
			Prefix	0	
			National Prefix	00	
			Country Code	33	
			International Prefix	000	
			In service	Checked	
			Check OOS	Checked	
	0.5		Session Timers -> Refresh Method	Reinvite	
	SIP Line		Session Timers -> Timer (seconds)	14880	
			Redirect and Transfer -> Incoming Supervised REFER	Never	
			Redirect and Transfer -> Outgoing Supervised REFER	Never	
			ITSP Proxy Address	primary SBC's IP address	
			Transport	Layer 4 Protocol	UDP
			Transport	Network Configuration -> Use Network	None

¹⁶ Two options are possible, depending on the needs. If CAC is reached on Remote Site call can be rerouted to Voicemail located on main site or rejected with 503 message (configured above). If CAC is reached on the main site call will be always rejected, no matter what is configured in this field.



			Topology Info	
			Send Port	5060
			Listen Port	5060
			Incoming Group	110
			Outgoing Group	110
			Max Sessions	Default=10 Range 1 - 250
			Local URI -> Display	Use Internal Data
			Local URI -> Content	Use Internal Data
			Local URI -> Fleld meaning -> Forwarding/Outgoin g calls	Caller
			Local URI -> Fleld meaning -> Forwarding/Twinnin g	Original Caller
			Local URI -> Fleld meaning -> Forwarding/Incomin g calls	Called
			Contact-> Display	Use Internal Data
			Contact-> Content	Use Internal Data
	Call Details	Call Details	Contact -> Fleld meaning -> Outgoing calls	Caller
			Contact -> Fleld meaning -> Forwarding/Twinnin g	Original Caller
			Contact -> Fleld meaning -> Incoming calls	Called
			Diversion Header	Checked
			Diversion Header -> Display	Use Internal Data
			Diversion Header -> Content	Use Internal Data
			Diversion Header -> Fleld meaning -> Outgoing Calls	None
			Diversion Header -> Fleld meaning -> Forwarding/Twinnin g	Caller
			Diversion Header -> Fleld meaning -> Incoming Calls	None



		Codec Selection	Custom
	VolP	Codec Selected	G.722 64K G.711 ALAW 64K
		DTMF Support	RFC2833/RFC4733
		Local HOLD Music	Checked
		RE-ivite Supported	Checked
		Allow Direct Media Path	Checked
		Force direct media with phones	Checked
		PRACK/100rel Supported	Checked
		Use + for International	On/Off ¹⁷
		Caller ID from From Header	Checked
		Send From in Clear	Checked
		Cache Auth Credentials	Unchecked
		Add UUI Header	Checked
		Add UUI Header to redirected calls	Checked
		Media -> P-Early- Media Support	All
		Media -> Force Early Direct Media	Checked
	SIP Advanced	Media -> Media Connection Preservation	System
		Media -> Indicate HOLD	Checked
		Call Control -> Call Initiation Timeout (s)	18
		Call Control -> Call Queuing Timeout (m)	1
		Call Control -> Service Busy Response	503 – Service Unavailable
		Call Control -> on No User Responding Send	480-Temporarily Unavailable
		Call Control -> Action on CAC	Allow Voicemail / Reject Call ¹⁸

 $^{^{17}}$ When set to On, outgoing international calls use E.164/International format with a '+' followed by the country code and then the directory number (optional).



		Location limit	
		Call Control -> Suppress Q.850 Reason Header	Checked
	Engineering	Custom String	SLIC_NO_USER_AV AIL=480 ¹⁹
		Line Number	111
		Local Domain Name	Secondary IPO's IP address
		Location	Cloud
		Prefix	0
		National Prefix	00
		Country Code	33
		International Prefix	000
		In service	Checked
	SIP Line	Check OOS	Checked
		Session Timers -> Refresh Method	Reinvite
SIP Line		Session Timers -> Timer (seconds)	14880
		Redirect and Transfer -> Incoming Supervised REFER	Never
		Redirect and Transfer -> Outgoing Supervised REFER	Never
		ITSP Proxy Address	backup SBC's IP address
		Layer 4 Protocol	UDP
	Transport	Network Configuration -> Use Network Topology Info	None
		Send Port	5060
		Listen Port	5060
			444
	Call Details	Incoming Group	111
	SIP Line	SIP Line	Call Control -> Suppress Q.850 Reason Header Engineering Custom String Line Number Local Domain Name Location Prefix National Prefix Country Code International Prefix In service SIP Line SIP Line SIP Line SIP Line SIP Line Transfer -> Outgoing Supervised REFER Redirect and Transfer -> Outgoing Supervised REFER ITSP Proxy Address Layer 4 Protocol Network Configuration -> Use Network Topology Info Send Port

¹⁸ Two options are possible, depending on the needs. If CAC is reached on Remote Site call can be rerouted to Voicemail located on main site or rejected with 503 message (configured above). If CAC is reached on the main site call will be always rejected, no matter what is configured in this field.

¹⁹ This Custom String is required for triggering DTO option, for an unregistered/unplugged phone located on a remote site without media gateway.



		Max Sessions	Default=10
			Range 1 - 250
		Local URI -> Display	Use Internal Data
		Local URI -> Content	Use Internal Data
		Local URI -> Fleld meaning -> Forwarding/Outgoin g calls	Caller
		Local URI -> Fleld meaning -> Forwarding/Twinnin g	Original Caller
		Local URI -> Fleld meaning -> Forwarding/Incomin g calls	Called
		Contact-> Display	Use Internal Data
		Contact-> Content	Use Internal Data
		Contact -> Fleld meaning -> Outgoing calls	Caller
		Contact -> Fleld meaning -> Forwarding/Twinnin g	Original Caller
		Contact -> Fleld meaning -> Incoming calls	Called
		Diversion Header	Checked
		Diversion Header -> Display	Use Internal Data
		Diversion Header -> Content	Use Internal Data
		Diversion Header -> Fleld meaning -> Outgoing Calls	None
		Diversion Header -> Fleld meaning -> Forwarding/Twinnin g	Caller
		Diversion Header -> Fleld meaning -> Incoming Calls	None
		Codec Selection	Custom
		Codec Selected	G.722 64K G.711 ALAW 64K
	VoIP	DTMF Support	RFC2833/RFC4733
		Local HOLD Music	Checked
		RE-ivite Supported	Checked



		All D' 'AA ''	
		Allow Direct Media Path	Checked
		Force direct media with phones	Checked
		PRACK/100rel Supported	Checked
		Use + for International	On/Off ²⁰
		Caller ID from From Header	Checked
		Send From in Clear	Checked
		Cache Auth Credentials	Unchecked
		Add UUI Header	Checked
		Add UUI Header to redirected calls	Checked
	Media Support Media -> Force	Media -> P-Early- Media Support	All
		Media -> Force Early Direct Media	Checked
		Connection	System
	SIP Advanced	Media Indicate HOLD	Checked
		Call Control -> Call Initiation Timeout (s)	18
		Call Control -> Call Queuing Timeout (m)	1
		Call Control -> Service Busy Response	503 – Service Unavailable
		Call Control -> on No User Responding Send	480-Temporarily Unavailable
		Call Control -> Action on CAC Location limit	Allow Voicemail / Reject Call ²¹
		Call Control -> Suppress Q.850 Reason Header	Checked

²⁰ When set to On, outgoing international calls use E.164/International format with a '+' followed by the country code and then the directory number (optional).

²¹ Two options are possible, depending on the needs. If CAC is reached on Remote Site call can be rerouted to Voicemail located on main site or rejected with 503 message (configured above). If CAC is reached on the main site call will be always rejected, no matter what is configured in this field.



		DECT line config	guration								
			DECT line configuration								
			Enable Provisioning	Checked							
			SARI/PARK	PARK license key ²²							
		Gateway	Subscriptions	Auto-Create / Preconfigured							
			Authentication Code	1234 ²³							
, line	P DECT _ine		Enable Resiliency	Checked							
IPO L	ıne -		Gateway IP Address	DECT IPBS's IP address							
		VoIP	Allow Direct Media Path	Checked							
			Codec Selection	Custom							
			Codec Selected	G.711 ALAW 64K							
	Se	curity settings fo	r IP DECT								
S	Services	HTTP -> Service details	Service Security Level	Unsecure + Secure							
	Right Group	IPDECT Group -> HTTP	DECT R4 Provisioning	Checked							
Primary Security	uritv	IPDECTServi ce -> Service User Details	Name	IPDECTService							
IPO Occurry			Password	password							
	Service		Account status	Enabled							
	Jsers		Account Expiry	No Account Expiry							
			Right Group Membership	IPDECT Group							
Dial Plan configuration ²⁴											
Dial Plan – General dialing configuration											
			Dial Delay Time (secs)	10							
Primary System -	-	Telephony ->Telephony	Dial Delay Count	0							
		. ,	Default No Answer Time	15							
Secondary IPO System -	-	Telephony	Dial Delay Time (secs)	10							
(if used)		->Telephony	Dial Delay Count	0							

²² License number has to match the one configured on DECT IPBS line under SARI

²³ Authentication code has to match the one configured on DECT IPBS under DECT-> System

 $^{^{24}}$ This is common configuration. It may be required to adjust dial plan configuration per particular system.



Dial Plan - Short Codes and ARS configuration when local PSTN access is not used ARS Route Name Main Code N Feature Dial Telephone Number N Line Group ID 10 Code N Feature Dial Telephone Number N Line Group ID 11 Code 002xxxxxxxxx25 Feature Dial Telephone Number 02N Line Group ID 50: Main Code 000N; Feature Dial Telephone Number 00N Line Group ID 50: Main Telephone Number 00N Line Group ID 50: Main Route Name Main Add Feature Dial Telephone Number N					Default No Answer Time	15
Primary IPO ARS ARS1 ARS1 Code N Primary IPO Add Telephone Number N N Primary IPO Code N Primary IPO Feature Dial Telephone Number N N Line Group ID 11 Code 002xxxxxxxxx²5 Feature Dial Telephone Number O2N Line Group ID 50: Main Code 000N; Feature Dial Telephone Number O0N Line Group ID 50: Main Telephone Number O0N Line Group ID 50: Main ARS Route Name Main Add Feature Dial	Dial	Plan – Short	Codes and A	ARS configuration	n when local PSTN acc	ess is not used
Primary IPO ARS ARS1 Add Feature Feature Feature Feature N IDO N Primary IPO Add Code N Feature Dial Telephone Number N IDO N Primary IPO Short Code Code O02xxxxxxx25 Short Code Feature Dial Telephone Number O2N IDO Dial Telephone Number O2N IDO Line Group ID S0: Main Code O00N; Feature Dial Telephone Number O0N IDO Dial Telephone Number O0N IDO Line Group ID S0: Main Main Code Number O0N IDO So: Main Main Main IDO Feature Dial IDO Add Add Feature Dial IDO Feature Dial IDO				ARS	Route Name	Main
ARS ARS1 ARS1 Telephone Number N Line Group ID 10 Code N Feature Dial Telephone Number N Line Group ID 11 Code O02XXXXXX25 Feature Dial Telephone Number Dial Telephone Number O2N Line Group ID 50: Main Code Short Code Short Code Short Code ARS ARS ARS ARS Route Name Main Add					Code	N
ARS ARS1				٨ -١ -١	Feature	Dial
Primary Prim				Add	Telephone Number	N
Primary IPO Add Feature Telephone Number N Telephone Number N Line Group ID N Telephone Number N Telephone Number Dial Short Code Short Code Code 002xxxxxxx25 Feature Dial Telephone Number Dial 50: Main Code 000N; Feature Dial Telephone Number Dial Telephone Number Dial		ARS	ARS1		Line Group ID	10
Primary IPO Add Telephone Number N Line Group ID 11 Code 002XXXXX25 Feature Dial Telephone Number 02N Line Group ID 50: Main Code 000XXXXX25 Feature Dial Telephone Number 02N Line Group ID 50: Main Code 000N; Feature Dial Telephone Number Dial ARS Route Name Main Code N Feature Dial					Code	N
Primary IPO Representation of the property of				۸dd	Feature	Dial
Short Code Sho				Add	Telephone Number	N
Short Code Eature Dial Short Code Telephone Number 02N Line Group ID 50: Main Code 000N; Feature Dial Telephone Number Dial Telephone Number 00N Line Group ID 50: Main Line Group ID 50: Main Code N Add Feature Dial					Line Group ID	11
Short Code - Telephone Number 02N	0				Code	002XXXXXXXXX ²⁵
Short Code C				-	Feature	Dial
Code					Telephone Number	02N
Short Feature Dial Telephone Number OON Line Group ID 50: Main ARS Route Name Main Code N Feature Dial Code N		Short			Line Group ID	50: Main
Telephone Number		Code	<u> </u>	-	Code	000N;
Code					Feature	Dial
ARS Route Name Main Code N Feature Dial					Telephone Number	00N
Code N Feature Dial					Line Group ID	50: Main
Add Feature Dial				ARS	Route Name	Main
Add					Code	N
				Add	Feature	Dial
1212-1111111111111111111111111111111111					Telephone Number	N
Secondary ARS ARS1 Line Group ID 110	Secondary	ARS	ARS1		Line Group ID	110
IPO Code N	IPO				Code	N
(if used) Feature Dial	(if used)			Add	Feature	Dial
Telephone Number N				Auu	Telephone Number	N
Line Group ID 111					Line Group ID	111
Short Short Code 002XXXXXX ²⁶		Short	Short		Code	002XXXXXXXX ²⁶
Code Code Feature Dial				-	Feature	Dial

 $^{^{25}}$ It is not possible to add one global entry for immediate national numbers, so such configuration should be repeated for each national numbering pattern 00ZABPQMCDU, where Z is a digit from range 1-9.

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 $^{^{26}}$ It is not possible to add one global entry for immediate national numbers, so such configuration should be repeated for each national numbering pattern 00ZABPQMCDU, where Z is a digit from range 1-9.



				Telephone Number	02N
				Line Group ID	50: Main
				Code	000N;
		Short		Feature	Dial
		Code	-	Telephone Number	00N
				Line Group ID	50: Main
D	ial Plan – Sho	rt Codes and	ARS configuration	on when local PSTN ac	cess is used ²⁷
			ARS	Route Name	PSTN_for_HQ313
				Code	N
		ARS2 ²⁸		Feature	Dial
			Add	Telephone Number	9N
				Line Group ID	99901
		ARS	ARS	Route Name	HQ313
ARS				Alternate Route	PSTN_for_HQ313
	ARS		Add	Code	N
				Feature	Dial
		ARS1		Telephone Number	N
				Line Group ID	10
Primary			Add	Code	N
IPO				Feature	Dial
				Telephone Number	N
				Line Group ID	11
			User	Name	RS140
				-	Apply User Rights value
	User Rights	User Rights	0.10.1 000.00	Short Code table (Code, Telephone Number, Feature, Line Group ID)	Please refer to next section
			User Rights Membership	Member of this User Rights	All RS140 users
	Short	Short		Code	002XXXXXX ²⁹
	Code	Code	-	Feature	Dial

²⁷ Below configuration should be repeated for each location using local PSTN access.

²⁸ Repeat the configuration steps for all Expansion Units within the IPO solution that will be used for local PSTN access.

 $^{^{29}}$ It is not possible to add one global entry for immediate national numbers, so such configuration should be repeated for each national numbering pattern 00ZABPQMCDU, where Z is a digit from range 1-9.



				Telephone Number	02N		
				Line Group ID	54: RS140		
				Code	000N;		
		Short		Feature	Dial		
		Code	-	Telephone Number	00N		
				Line Group ID	54: RS140		
Note: Before configuring ARS tables on secondary IPO it is necessary to save ARS tables from primary IPO as a templates. This approach is necessary if we are using User Rights (described in next section) as it's not possible to modify ARS number.							
Primary IPO	ARS	Select first ARS table created in previous steps and click Export as Template (Binary) in top-right window menu. Repeat this action for all other ARS tables created on primary IPO.					
Secondary IPO (if used)	ARS	 Chose New from Template (Binary) and select from the list saved ARS table³⁰. Double-click on the Short Code entry within added ARS table and modify Line Group ID with the equivalent number configured on secondary IPO. Repeat the steps above for each ARS table copied from primary IPO. 					
				Code	9N		
Expansion	Short	Short		Feature	Dial		
Gateway	Code	Code		Telephone Number	NS225374380 ³¹		
				Line Group ID	3		
[Dial Plan – Ind	coming Call F	Route configuration	on - Incoming call to ph	none user ³²		
			Ctondord	Line Group ID	10		
		Incoming Call	Standard	Incoming Number	+33296084361		
	Route 10	Destinations	Destination -> Default Value	4701001 Extn4701001			
	Incoming		Ctondord	Line Group ID	11		
- Call Route	Incoming Call	Standard	Incoming Number	+33296084361			
		Route 11	Destinations	Destination -> Default Value	4701001 Extn4701001		
		Incoming	Standard	Line Group ID	3		
		Call	Jiai iuai u	Incoming Number	225374381 ³⁴		

 $^{^{30}}$ It is important to add all ARS tables for local PSTN access first, otherwise it will be required to manually select Alternate Route table later.

³¹ Sxxxxxxxx means that provided number is used for CLI in outgoing calls via local PSTN line

 $^{^{32}}$ Each user has to have DID number assigned. To route incoming BTIP calls it is required to have SIP URI tab on primary and backup SIP trunk, configuration of which is described in section: SIP trunks configuration.

³⁴ This field can be used to match the called public number with private one.



		Route 3 ³³	Destinations	Destination -> Default Value	4701001 Extn4701001 ³⁵
Dial Plan -	- Incoming Ca		iguration - Incor i.e. voicemail, hu	ning call to destination o	other than phone user
			101 10100111011, 110	Incoming Group	10
				Outgoing Group	10
				Max Sessions	Default=10 Range 1 - 250
				Local URI -> Display	Auto
				Local URI -> Content	Auto
				Local URI -> Fleld meaning -> Forwarding/Outgoin g calls	Caller
		SIP Line		Local URI -> Fleld meaning -> Forwarding/Twinnin g	Original Caller
	10	Call Details	Local URI -> Fleld meaning -> Forwarding/Incomin g calls	Called	
				Contact-> Display	Auto
Primary	Line			Contact-> Content	Auto
IPO				Contact -> Fleld meaning -> Outgoing calls	Caller
			Contact -> Fleld meaning -> Forwarding/Twinnin g	Original Caller	
			Contact -> Fleld meaning -> Incoming calls	Called	
			Incoming Group	11	
				Outgoing Group	11
				Max Sessions	Default=10 Range 1 - 250
		SIP Line	Call Details	Local URI -> Display	Auto
		11		Local URI -> Content	Auto
			Local URI -> Fleld meaning -> Forwarding/Outgoin g calls	Caller	

³³ Dedicated for local PSTN access (optional)

 $^{^{\}rm 35}$ Binds public DID with the private extension.



				Local URI -> Fleld meaning -> Forwarding/Twinnin g Local URI -> Fleld	Original Caller
				meaning -> Forwarding/Incomin g calls	Called
				Contact-> Display	Auto
				Contact-> Content	Auto
				Contact -> Fleld meaning -> Outgoing calls	Caller
				Contact -> Fleld meaning -> Forwarding/Twinnin g	Original Caller
				Contact -> Fleld meaning -> Incoming calls	Called
		Incoming Call Route 10	Standard	Line Group ID	10
			Staridard	Incoming Number	+33296084362
_	Incoming Call		Destinations	Destination -> Default Value	Voicemail / Hunt group / etc.
	Route	Incoming Call Route 11	Standard	Line Group ID	11
				Incoming Number	+33296084362
			Destinations	Destination -> Default Value	Voicemail / Hunt group / etc.
		Dial Plan	configuration for	r Emergency calls	
	Dial Plan cor	nfiguration for	Emergency call	s - Short Code: Dial Em	nergency ³⁶
				Code	112
	Short	Short		Feature	Dial Emergency
	Code	Code		Telephone Number	112
				Line Group ID	Blank
Primary IPO			ADC	Route Name	HQ313-Emergency
			ARS	Alternate Route	PSTN_for_HQ313
	ADC	ADC		Code	N
	ARS	ARS	A -1-1	Feature	Dial
			Add	Telephone Number	N
				Line Group ID	20 ³⁷

 $^{^{\}rm 36}$ If the system uses prefixes for external dialing, the dialing of emergency numbers with and without the prefix should be allowed.

 $^{^{}m 37}$ This value must be different than the one used for standard calls.



				Code	N
			A -1-1	Feature	Dial
			Add	Telephone Number	N
				Line Group ID	21 ³⁸
	Location	Location	Location	Emergency ARS	HQ313-Emergency
				Incoming Group	0
				Outgoing Group	20 ³⁹
				Max session	Default=10 Range 1 - 250
				Local URI -> Display	Example: +33296083900
				Local URI -> Content	Example: +33296083900
		SIP Line 10 Line	Call Details	Contact -> Fleld meaning -> Outgoing Call	Explicit
				Local URI -> Fleld meaning -> Forwarding/Twinnin g	Original Caller
	Line			Local URI -> Fleld meaning -> Forwarding/Incomin g calls	Called
				Contact-> Display	Example: +33296083900
				Contact-> Content	Example: +33296083900
				Contact -> Fleld meaning -> Outgoing Call	Explicit
			Contact -> Fleld meaning -> Forwarding/Twinnin g	Original Caller	
				Contact -> Fleld meaning -> Incoming calls	Called
				Incoming Group	0
		SIP Line	Call Details	Outgoing Group	21 ⁴⁰
		11		Max session	Default=10 Range 1 - 250

 $^{^{\}rm 38}$ This value must be different than the one used for standard calls.

³⁹ This value must equal the one configured under emergency ARS on first position!

⁴⁰ This value must equal the one configured under emergency ARS on second position!



				Local URI -> Display	Example: +33296083900
				Local URI -> Content	Example: +33296083900
			Contact -> Fleld meaning -> Outgoing Call	Explicit	
				Local URI -> Fleld meaning -> Forwarding/Twinnin g	Original Caller
				Local URI -> Fleld meaning -> Forwarding/Incomin g calls	Called
				Contact-> Display	Example: +33296083900
				Contact-> Content	Example: +33296083900
			Contact -> Fleld meaning -> Outgoing Call	Explicit	
				Contact -> Fleld meaning -> Forwarding/Twinnin g	Original Caller
			Contact -> Fleld meaning -> Incoming calls	Called	
				Code	112
	Short	Short		Feature	Dial Emergency
	Code	Code	-	Telephone Number	112
				Line Group ID	Blank
			ADO	Route Name	HQ313-Emergency
Secondary	Secondary		ARS	Alternate Route	PSTN_for_HQ313
IPO (if used)				Code	N
			Add	Feature	Dial
	ARS	ARS	Auu	Telephone Number	N
				Line Group ID	120 ⁴¹
				Code	N
			Add	Feature	Dial
				Telephone Number	N

 $^{^{\}rm 41}$ This value must be different than the one used for standard calls.



				Line Group ID	121 ⁴²
	Location	Location	Location	Emergency ARS	HQ313-Emergency
				Incoming Group	0
				Outgoing Group	120 ⁴³
				Max session	Default=10 Range 1 - 250
				Local URI -> Display	Example: +33296083900
				Local URI -> Content	Example: +33296083900
				Contact -> Fleld meaning -> Outgoing Call	Explicit
				Local URI -> Fleld meaning -> Forwarding/Twinnin g	Original Caller
		SIP Line 110	Call Details	Local URI -> Fleld meaning -> Forwarding/Incomin g calls	Called
	Line			Contact-> Display	Example: +33296083900
				Contact-> Content	Example: +33296083900
				Contact -> Fleld meaning -> Outgoing Call	Explicit
				Contact -> Fleld meaning -> Forwarding/Twinnin g	Original Caller
				Contact -> Fleld meaning -> Incoming calls	Called
				Incoming Group	0
				Outgoing Group	121 ⁴⁴
		SIP Line	Call Details	Max Session	Default=10 Range 1 - 250
		111		Local URI -> Display	Example: +33296083900
				Local URI -> Content	Example: +33296083900

⁴² This value must be different than the one used for standard calls.

⁴³ This value must equal the one configured under emergency ARS on first position!

⁴⁴ This value must equal the one configured under emergency ARS on second position!



				Contact -> Fleld meaning -> Outgoing Call	Explicit
				Local URI -> Fleld meaning -> Forwarding/Twinnin g	Original Caller
				Local URI -> Fleld meaning -> Forwarding/Incomin g calls	Called
				Contact-> Display	Example: +33296083900
				Contact-> Content	Example: +33296083900
				Contact -> Fleld meaning -> Outgoing Call	Explicit
				Contact -> Fleld meaning -> Forwarding/Twinnin g	Original Caller
				Contact -> Fleld meaning -> Incoming calls	Called
	U	ser / Extension	on creation – ma	nual for IP endpoints ⁴⁵	
				Name	Extn3130001
				Password	password ⁴⁶
			User	Audio Conference PIN	PIN
	User	User		Extension	3130001
Primary IPO				Profile	Basic User / Power User ⁴⁷
			Telephony -> Supervisor Settings	Login Code	login code ⁴⁸
	Extension	H.323 / SIP Extension		utomatically prompt for saving User part and wi mation.	

 $^{^{45}}$ Below values are an examples and should be treated only as a common guidelines for new user creation

 $^{^{46}}$ Password provided here will be used only for user login to applications like One-X Portal or One-X Mobile.

 $^{^{47}}$ Power user allows to use additional features like Softphone or Telecommuter mode. Separate license is required.

⁴⁸ Login code provided here will be used for phone's registration. Not obligatory.



		-	Extn	Phone Password	Password ⁴⁹		
User / Exte	User / Extension creation - Public numbers assignment: NDI number declaration for non-DID users						
				SIP Name	Example: +33296084360		
Primary IPO	User	User	SIP	SIP Display Name (Alias)	Example: +33296084360		
				Contact	Example: +33296084360		
User / Ext	User / Extension creation - Public numbers assignment: NDI number declaration for DID users ⁵⁰						
			SIP	SIP Name	Example: +33296084361		
Primary IPO	User	User User		SIP Display Name (Alias)	Example: +33296084361		
				Contact	Example: +33296084361		
	Us	er / Extensior	creation - The	e "NoUser" configuration	1		
Primary IPO	User	NoUser	Source Numbers	Source Number	MEDIA_DISABLE_RF C2833_ON_IPO ⁵¹		
Secondary IPO (if used)	User	NoUser	Source Numbers	Source Number	MEDIA_DISABLE_RF C2833_ON_IPO ⁵²		
Expansion Gateway (if used)	User	NoUser	Source Numbers	Source Number	MEDIA_DISABLE_RF C2833_ON_IPO ⁵³		

⁴⁹ This code will be used by H.323 phone users to login

⁵⁰ Each user has to have DID number assigned, so configuration should be repeated for each user.

⁵¹ Configuration of NUSN is mandatory to have direct media for H.323 and DECT users registered to local gateways

⁵² Configuration of NUSN is mandatory to have direct media for H.323 and DECT users registered to local gateways

 $^{^{53}}$ Configuration of NUSN is mandatory to have direct media for H.323 and DECT users registered to local gateways



5 SIP TRUNKING CONFIGURATION CHECKLIST FOR Orange 10.1 SP1 (10.1.0.1.0 Build 3) and Orange 10.0 SP3 + CP (10.0.0.3.12 build 2)

The checklist below presents all the steps of configuration required for interoperability between BTIP/BT and Orange IP Office 10.1 SP1 (10.1.0.1.0 Build 3) and Orange IP Office 10.0 SP3 + CP (10.0.0.3.12 build 2).

Trunk configuration - IP Office Server Edition

Access type: IP Office Web Manager page.

Platform	Configuration place	Configuration details	
	Services		
Primary IPO	System	running services: IP Office Voicemail One-X Portal Web Manager Web License Manager	

Access type: IP Office Manager application.

Platform	Menu	Object	Tab	Parameter	Value	
	System configuration – Locale configuration					
Every platform in the solution ⁵⁴	System	-	System	Locale	France2 (French)	
		System c	onfiguration – DS	SCP configuration		
F		System -		DSCP (Hex) / DSCP	B8 / 46	
Every platform in the	System		LAN1 -> VoIP	Video DSCP (Hex) / Video DSCP	88 / 34	
solution				SIG DSCP (Hex) / SIG DSCP	B8 / 46	
			DHCP configurat	tion offer		
				Start address	Start IP address	
Primary	Cyctom		LAN1 ->	Subnet Mask	Subnet Mask	
IPO	System	-	DHCP Poll	Default Router	Router IP address	
				Pool size	DHCP pool size	
			Codec configu	ıration		

⁵⁴ Every platform in the solution: primary IPO, secondary IPO (if used), expansion units (if used)



			T.	Companding Law	A-Law
		Telephony -> Telephony	High Quality Conferencing	Checked	
Every platform in the	System Line	-		Ignore DTMF Mismatch For Phones	Checked
solution	EliTo		VoIP	RFC2833 Default Payload	101
				Default Codec Selection -> Selected	G.722 64K G.711 ALAW 64K
		Call Admitic	on Control & Loc	ation configuration ⁵⁵	
				Location Name	Ex:RS140
				Subnet Address	6.201.40.0
				Subnet Mask	255.255.255.0
			Location	Parent Location for CAC	<none></none>
Solution level	Location	Location		Call Admission Control -> Total Maximum Calls	99
				Call Admission Control -> External Maximum Calls	99
				Call Admission Control -> Internal Maximum Calls	99
Every platform in the solution	System	-	System	Location	Ex:HQ313
			Fallback configu	ıration ⁵⁶	
Primary IPO	Location	Location	Location	Fallback System	Local GW's IP address
			SCN lines confi	guration	
				Outgoing Group ID	99998
Primary		IP Office		Transport Type	Proprietary
IPO ⁵⁷	Line	line ⁵⁸	Line	Networking Level	SCN
				Gateway -> Address	Backup IPO's IP address

⁵⁵ For each physical site (Headquarter and Remote Sites) dedicated location has to be created, mainly for Call Admission Control and emergency calls management. This section provides example values.

⁵⁶ For each location where local gateway should act as a backup system in case of primary server failure Fallback System should be defined.

⁵⁷ Repeat the steps on primary IPO to create separate SCN line for each local gateway in the solution.

⁵⁸ SCN Line to secondary server



				Gateway -> Location	Location name
				SCN Resiliency Options -> Supports Resiliency	Checked
				- Backs up my IP Phones	Checked
				- Backs up my Hunt Groups	Checked
				- Backs up my Voicemail	Checked
				- Backs up my IP Dect Phones	Checked
				- Backs up my One- x Portal	Checked
			VoIP settings	Allow Direct Media Path	Checked
				Outgoing Group ID	99901 - 99930
			Line	Transport Type	Proprietary
				Networking Level	SCN
				Gateway -> Address	Local GW's IP address
				Gateway -> Location	Location name
		IP Office Line ⁵⁹		SCN Resiliency Options -> Supports Resiliency	Checked
				- Backs up my IP Phones	Unchecked
				- Back up my Hunt Groups	Unchecked
				- Back up my IP Dect Phones	Unchecked
			VoIP settings	Allow Direct Media Path	Checked
				Outgoing Group ID	99999
				Transport Type	Proprietary
Secondary IPO Line	Line	ine IP Office line ⁶¹	Line	Networking Level	SCN
(if used) ⁶⁰	LII I⊡		LITIC	Gateway -> Address	Primary IPO's IP address
				Gateway -> Location	Location name

⁵⁹ SCN Line to expansion gateway

 $^{^{60}}$ Repeat the steps on secondary IPO (if used) to create separate SCN line for each local gateway in the solution.

⁶¹ SCN Line to Primary server



				SCN Resiliency Options -> Supports Resiliency	Checked
				- Backs up my IP Phones	Checked
				- Backs up my Hunt Groups	Checked
				- Back up my Voicemail	Checked
				- Back up my IP Dect Phones	Checked
				- Back up my one-X Portal	Checked
			VoIP settings	Allow Direct Media Path	Checked
				Outgoing Group ID	99901 - 99930
				Transport Type	Proprietary
			Line	Networking Level	SCN
		IP Office Line ⁶²		Gateway -> Address	Local GW's IP address
				Gateway -> Location	Location name
				SCN Resiliency Options -> Supports Resiliency	Unchecked
			VoIP settings	Allow Direct Media Path	Checked
				Outgoing Group ID	99999
				Transport Type	Proprietary
Expansion	63	IP Office		Networking Level	SCN
Gateway	Line ⁶³	line ⁶⁴	Line	Gateway -> Address	Primary IPO's IP address
				Gateway -> Location	Location name
				SCN Resiliency Options -> Supports Resiliency	Checked

⁶² SCN line to expansion gateway

⁶³ Redundant architecture only

⁶⁴ SCN line to Primary server



			VoIP Settings	Allow Direct Media Path	Checked
				Outgoing Group ID	99998
				Transport Type	Proprietary
			Line	Networking Level	SCN
		IP Office Line ⁶⁵	ше	Gateway -> Address	Backup IPO's IP address
				Gateway -> Location	Location name
				SCN Resiliency Options -> Supports Resiliency	Unchecked
			VoIP Settings	Allow Direct Media Path	Checked
		SCN lines	configuration – l	ocal PSTN access	
Expansion	Line	PRI 30 (Universal	PRI line	Incoming Group ID	3
Gateway	LIIIC	ine (Oniversal	1 1 11 11110	Outgoing Group ID	3
		SIP Trun	ks configuration	- Global settings	
				SIP Trunks Enable	Checked
				SIP Registrar Enable	Checked
Primary IPO	System	-	LAN1 -> VoIP	Media Connection Preservation	Enabled
				Inhibit Off-Switch Forward/Transfer	Unchecked
				SIP Trunks Enable	Checked
Secondary			LANI	SIP Registrar Enable	Checked
IPO (if used)	System	-	LAN1 -> VoIP	Media Connection Preservation	Enabled
				Inhibit Off-Switch Forward/Transfer	Unchecked
		SIP 1	runks configurat	ion – SIP line	
	Line	SIP Line	SIP Line	Line Number	10

⁶⁵ SCN line to secondary server

⁶⁶ Line type depends on line type attached to Expansion Gateway



Primary IPO			Local Domain Name	Primary IPO's IP address
			Location	Cloud
		Prefix	0	
		National Prefix	00	
			Country Code	33
			International Prefix	000
			In service	Checked
			Check OOS	Checked
			Session Timers -> Refresh Nethod	Reinvite
			Session Timers -> Timer (seconds)	14880
			Redirect and Transfer -> Incoming Supervised REFER	Never
			Redirect and Transfer -> Outgoing Supervised REFER	Never
		Transport	ITSP Proxy Address	primary SBC's IP address
			Layer 4 Protocol	UDP
			Network Configuration -> Use Network Topology Info	None
			Send Port	5060
			Listen Port	5060
			Local URI	Use Internal Data
			Contact	Use Internal Data
			Display Name	Use Internal Data
			Identity->Identity	None
			Identity->Header	P Asserted ID
		SIP URI	Forwarding and Twinning -> Send Caller ID	Diversion Header
			Diversion Header	None
			Incoming Group	10
			Outgoing Group	10
			Max Sessions	Default=10 Range 1 - 250
1		VoIP	Codec Selection	Custom



	DTMF Support	RFC2833/RFC4733
	Local HOLD Music	Checked
	RE-ivite Supported	Checked
	Allow Direct Media Path	Checked
	Force direct media with phones	Checked
	PRACK/100rel Supported	Checked
	Use + for International	On/Off ⁶⁷
	Caller ID from From Header	Checked
	Send From in Clear	Checked
	Cache Auth Credentials	Unchecked
	Add UUI Header	Checked
	Add UUI Header to redirected calls	Checked
	Media -> P-Early- Media Support	All
	Media -> Force Early Direct Media	Checked
	Media -> Media Connection Preservation	System
SIP Advanced	Media -> Media Indicate HOLD	Checked
	Call Control -> Call Initiation Timeout (s)	18
	Call Control -> Call Queuing Timeout (m)	1
	Call Control -> Service Busy Response	503 – Service Unavailable
	Call Control -> on No User Responding Send	480-Temporarily Unavailable
	Call Control -> Action on CAC Location limit	Allow Voicemail / Reject Call
	Call Control -> Suppress Q.850	Checked

 $^{^{67}}$ When set to On, outgoing international calls use E.164/International format with a '+' followed by the country code and then the directory number (optional).



		Reason Header	
	Engineering	Custom String	SLIC_NO_USER_AV AIL=480
		Line Number	11
		Local Domain Name	Primary IPO's IP address
		Location	Cloud
		Prefix	0
		National Prefix	00
		Country Code	33
		International Prefix	000
		In service	Checked
	SIP Line	Check OOS	Checked
		Session Timers -> Refresh Nethod	Reinvite
		Session Timers -> Timer (seconds)	14880
		Redirect and Transfer -> Incoming Supervised REFER	Never
SIP Line		Redirect and Transfer -> Outgoing Supervised REFER	Never
		ITSP Proxy Address	backup SBC's IP address
		Layer 4 Protocol	UDP
	Transport	Network Configuration -> Use Network Topology Info	None
		Send Port	5060
		Listen Port	5060
		Local URI	Use Internal Data
		Contact	Use Internal Data
		Display Name	Use Internal Data
		Identity->Identity	None
	SIP URI	Identity->Header	P Asserted ID
		Forwarding and Twinning -> Send Caller ID	Diversion Header
		Diversion Header	None
		Incoming Group	11



	Outgoing Group	11
	Max Sessions	Default=10 Range 1 - 250
	Codec Selection	Custom
	Codec Selected	G.722 64K G.711 ALAW 64K
	DTMF Support	RFC2833/RFC4733
	Local HOLD Music	Checked
VoIP	RE-ivite Supported	Checked
	Allow Direct Media Path	Checked
	Force direct media with phones	Checked
	PRACK/100rel Supported	Checked
	Use + for International	On/Off ⁶⁸
	Caller ID from From Header	Checked
	Send From in Clear	Checked
	Cache Auth Credentials	Unchecked
	Add UUI Header	Checked
	Add UUI Header to redirected calls	Checked
	Media -> P-Early- Media Support	All
SIP	Media -> Force Early Direct Media	Checked
Advanced	Media -> Media Connection Preservation	System
	Media -> Indicate HOLD	Checked
	Call Control -> Call Initiation Timeout (s)	18
	Call Control -> Call Queuing Timeout (m)	1
	Call Control -> Service Busy Response	503 – Service Unavailable
	Call Control ->	480-Temporarily

 $^{^{68}}$ When set to On, outgoing international calls use E.164/International format with a '+' followed by the country code and then the directory number (optional).



			<u> </u>		
				on No User Responding Send	Unavailable
				Call Control -> Action on CAC Location limit	Allow Voicemail / Reject Call ⁶⁹
				Call Control -> Suppress Q.850 Reason Header	Checked
			Engineering	Custom String	SLIC_NO_USER_AV AIL=480
				Line Number	110
				Local Domain Name	Secondary IPO's IP address
				Location	Cloud
				Prefix	0
				National Prefix	00
				Country Code	33
				International Prefix	000
				In service	Checked
			SIP Line	Check OOS	Checked
				Session Timers -> Refresh Method	Reinvite
				Session Timers -> Timer (seconds)	14880
Secondary IPO (if used)	Line	SIP Line		Redirect and Transfer -> Incoming Supervised REFER	Never
				Redirect and Transfer -> Outgoing Supervised REFER	Never
				ITSP Proxy Address	primary SBC's IP address
				Layer 4 Protocol	UDP
			Transport	Network Configuration -> Use Network Topology Info	None
				Send Port	5060
				Listen Port	5060
			SIP URI	Local URI	Use Internal Data

⁶⁹ Two options are possible, depending on the needs. If CAC is reached on Remote Site call can be rerouted to Voicemail located on main site or rejected with 503 message (configured above). If CAC is reached on the main site call will be always rejected, no matter what is configured in this field.



		Contact	Use Internal Data
		Display Name	Use Internal Data
		Identity->Identity	None
		Identity->Header	P Asserted ID
		Forwarding and Twinning -> Send Caller ID	Diversion Header
		Diversion Header	None
		Incoming Group	110
		Outgoing Group	110
		Max Sessions	Default=10 Range 1 - 250
		Codec Selection	Custom
		Codec Selected	G.722 64K G.711 ALAW 64K
		DTMF Support	RFC2833/RFC4733
		Local HOLD Music	Checked
	VoIP	RE-ivite Supported	Checked
		Allow Direct Media Path	Checked
		Force direct media with phones	Checked
		PRACK/100rel Supported	Checked
		Use + for International	On/Off ⁷⁰
		Caller ID from From Header	Checked
		Send From in Clear	Checked
		Cache Auth Credentials	Unchecked
		Add UUI Header	Checked
	SIP Advanced	Add UUI Header to redirected calls	Checked
		Media -> P-Early- Media Support	All
		Media -> Force	Checked
		Early Direct Media	
		Early Direct Media Media -> Media Connection Preservation	System

 $^{^{70}}$ When set to On, outgoing international calls use E.164/International format with a '+' followed by the country code and then the directory number (optional).



		HOLD	
		Call Control -> Call Initiation Timeout (s)	18
		Call Control -> Call Queuing Timeout (m)	1
		Call Control -> Service Busy Response	503 – Service Unavailable
		Call Control -> on No User Responding Send	480-Temporarily Unavailable
		Call Control -> Action on CAC Location limit	Allow Voicemail / Reject Call ⁷¹
		Call Control -> Suppress Q.850 Reason Header	Checked
	Engineering	Custom String	SLIC_NO_USER_AV AIL=480 ⁷²
		Line Number	111
		Local Domain Name	Secondary IPO's IP address
		Location	Cloud
		Prefix	0
		National Prefix	00
		Country Code	33
		International Prefix	000
SIP Line	SIP Line	In service	Checked
		Check OOS	Checked
		Session Timers -> Refresh Method	Reinvite
		Session Timers -> Timer (seconds)	14880
		Redirect and Transfer -> Incoming Supervised REFER	Never
		Redirect and Transfer ->	Never

⁷¹ Two options are possible, depending on the needs. If CAC is reached on Remote Site call can be rerouted to Voicemail located on main site or rejected with 503 message (configured above). If CAC is reached on the main site call will be always rejected, no matter what is configured in this field.

⁷² This Custom String is required for triggering DTO option, for an unregistered/unplugged phone located on a remote site without media gateway.



		Outgoing Supervised REFER	
		ITSP Proxy Address	backup SBC's IP address
		Layer 4 Protocol	UDP
	Transport	Network Configuration -> Use Network Topology Info	None
		Send Port	5060
		Listen Port	5060
		Local URI	Use Internal Data
		Contact	Use Internal Data
		Display Name	Use Internal Data
		Identity->Identity	None
		Identity->Header	P Asserted ID
	SIP URI	Forwarding and Twinning -> Send Caller ID	Diversion Header
		Diversion Header	None
		Incoming Group	111
		Outgoing Group	111
		Max Sessions	Default=10 Range 1 - 250
		Codec Selection	Custom
		Codec Selected	G.722 64K G.711 ALAW 64K
		DTMF Support	RFC2833/RFC4733
		Local HOLD Music	Checked
	VoIP	RE-ivite Supported	Checked
		Allow Direct Media Path	Checked
		Force direct media with phones	Checked
		PRACK/100rel Supported	Checked
		Use + for International	On/Off ⁷³
	SIP Advanced	Caller ID from From Header	Checked
		Send From in Clear	Checked

 $^{^{73}}$ When set to On, outgoing international calls use E.164/International format with a '+' followed by the country code and then the directory number (optional).

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Cache Auth Credentials Add UUI Header Checked	1	1	1		
Add UUI Header to redirected calls Media > P-Early-Media Support All					Unchecked
Redirected calls Media >> P-Early- Media Support				Add UUI Header	Checked
Media Support Media - Prore Early Direct Media - Prore Early Direct Media - Prore Early Direct Media -					Checked
Early Direct Media					All
Connection Preservation Preser					Checked
HOLD Checked				Connection	System
Call Initiation Timeout (s) Call Control -> Call Queuing Timeout (m) Call Control -> Call Queuing Timeout (m) Call Control -> Service Busy Response Call Control -> On No User Responding Send Call Control -> On No User Responding Send Call Control -> Action on CAC Location limit Call Control -> Allow Voicemail / Reject Call Papers Queston Header Call Control -> Checked					Checked
Call Queuing Timeout (m)				Call Initiation	18
Service Busy Response Call Control -> on No User Responding Send Call Control -> Action on CAC Location limit Call Control -> Suppress Q.850 Reason Header Engineering Custom String Checked Primary IPO Line IP DECT Line Gateway Service Busy Responds 480-Temporarily Unavailable Allow Voicemail / Reject Call ⁷⁴ Checked SLIC_NO_USER_AV AIL=480 Checked SLIC_NO_USER_AV AIL=480 SARI/PARK PARK license key ⁷⁵ Subscriptions Auto-Create / Preconfigured Authentication Code 1234 ⁷⁶				Call Queuing	1
on No User Responding Send Call Control -> Action on CAC Location limit Call Control -> Suppress Q.850 Reason Header Engineering Custom String SLIC_NO_USER_AV AIL=480 Primary IPO Line IP DECT Line Gateway Gateway Gateway Gateway Allow Voicemail / Reject Call ⁷⁴ Checked SLIC_NO_USER_AV AIL=480 Checked SARI/PARK PARK license key ⁷⁵ Subscriptions Auto-Create / Preconfigured Authentication Code 1234 ⁷⁶				Service Busy	
Action on CAC Location limit Call Control -> Suppress Q.850 Reason Header Engineering Custom String SLIC_NO_USER_AV AlL=480 DECT line configuration Enable Provisioning Checked SARI/PARK PARK license key ⁷⁵ Subscriptions Auto-Create / Preconfigured Authentication Code 1234 ⁷⁶				on No User	
Suppress Q.850 Reason Header Engineering Custom String SLIC_NO_USER_AV AlL=480 DECT line configuration Enable Provisioning Checked SARI/PARK PARK license key ⁷⁵ Subscriptions Auto-Create / Preconfigured Authentication Code 1234 ⁷⁶				Action on CAC	
Primary IPO Line IP DECT Line Gateway Gateway Custom String AlL=480 AlL=480 Enable Provisioning SARI/PARK PARK license key ⁷⁵ Subscriptions Auto-Create / Preconfigured Authentication Code 1234 ⁷⁶				Suppress Q.850	Checked
Primary IPO Line Line Line Line Line Line Line Line Line Enable Provisioning Checked SARI/PARK PARK license key ⁷⁵ Subscriptions Auto-Create / Preconfigured Authentication Code 1234 ⁷⁶			Engineering	Custom String	
Primary IPO Line Line Line Line SARI/PARK PARK license key ⁷⁵ Subscriptions Auto-Create / Preconfigured Authentication Code 1234 ⁷⁶			DECT line config	guration	
Primary IPO Line Line IP DECT Line Gateway Subscriptions Auto-Create / Preconfigured Authentication Code 1234 ⁷⁶				Enable Provisioning	Checked
IPO Line Line Gateway Subscriptions Preconfigured Authentication Code 1234 ⁷⁶				SARI/PARK	PARK license key ⁷⁵
	Line		Gateway	Subscriptions	
Enable Resiliency Checked				Authentication Code	1234 ⁷⁶
				Enable Resiliency	Checked

⁷⁴ Two options are possible, depending on the needs. If CAC is reached on Remote Site call can be rerouted to Voicemail located on main site or rejected with 503 message (configured above). If CAC is reached on the main site call will be always rejected, no matter what is configured in this field.

⁷⁵ License number has to match the one configured on DECT IPBS line under SARI

⁷⁶ Authentication code has to match the one configured on DECT IPBS under DECT-> System



				Gateway IP Address	DECT IPBS's IP
				Allow Direct Media	address
		V	VoIP	Path	Checked
				Codec Selection	Custom
				Codec Selected	G.711 ALAW 64K
		Se	ecurity settings fo	or IP DECT	
		Services	HTTP -> Service details	Service Security Level	Unsecure + Secure
		Right Group	IPDECT Group -> HTTP	DECT R4 Provisioning	Checked
Primary IPO	Security			Name	IPDECTService
IF U			IDDEOTS :	Password	password
		Service	IPDECTServi ce -> Service	Account status	Enabled
		Users	User Details	Account Expiry	No Account Expiry
				Right Group Membership	IPDECT Group
			Dial Plan configu	ıration ⁷⁷	
		Dial Pla	n – General dialir	ng configuration	
		ystem -	Telephony ->Telephony	Dial Delay Time (secs)	10
Primary IPO	System			Dial Delay Count	0
				Default No Answer Time	15
Secondary				Dial Delay Time (secs)	10
IPO	System	-	Telephony ->Telephony	Dial Delay Count	0
(if used)			, i elektrieri	Default No Answer Time	15
Dial	Plan – Short	Codes and A	ARS configuration	when local PSTN acce	ess is not used
			ARS	Route Name	Main
				Code	N
			Add	Feature	Dial
Primary ,	ARS	ARS1	Auu	Telephone Number	N
IPO	ANO	ANOI		Line Group ID	10
				Code	N
			Add	Feature	Dial
				Telephone Number	N
					

 $^{^{77}}$ This is common configuration. It may be required to adjust dial plan configuration per particular system.

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				Line Group ID	11
				Code	002XXXXXXXX ⁷⁸
		Short		Feature	Dial
		Code	-	Telephone Number	02N
	Short			Line Group ID	50: Main
	Code			Code	000N;
		Short		Feature	Dial
		Code	-	Telephone Number	00N
				Line Group ID	50: Main
			ARS	Route Name	Main
				Code	N
		ARS1	Add	Feature	Dial
				Telephone Number	N
	ARS			Line Group ID	110
				Code	N
				Feature	Dial
Secondary				Telephone Number	N
IPO				Line Group ID	111
(if used)				Code	002XXXXXXX ⁷⁹
		Short		Feature	Dial
		Code	-	Telephone Number	02N
	Short			Line Group ID	50: Main
	Code			Code	000N;
		Short		Feature	Dial
		Code	=	Telephone Number	00N
				Line Group ID	50: Main
Dia	al Plan – Shor	t Codes and	ARS configuration	on when local PSTN acc	cess is used ⁸⁰
Primary	ARS	ARS2 ⁸¹	ARS	Route Name	PSTN_for_HQ313

 $^{^{78}}$ It is not possible to add one global entry for immediate national numbers, so such configuration should be repeated for each national numbering pattern 00ZABPQMCDU, where Z is a digit from range 1-9.

 $^{^{79}}$ It is not possible to add one global entry for immediate national numbers, so such configuration should be repeated for each national numbering pattern 00ZABPQMCDU, where Z is a digit from range 1-9.

⁸⁰ Below configuration should be repeated for each location using local PSTN access.

 $^{^{81}}$ Repeat the configuration steps for all Expansion Units within the IPO solution that will be used for local PSTN access.



IPO				Code	N	
				Feature	Dial	
			Add	Telephone Number	9N	
				Line Group ID	99901	
				Route Name	HQ313	
			ARS	Alternate Route	PSTN_for_HQ313	
				Code	N	
				Feature	Dial	
		4504	Add	Telephone Number	N	
		ARS1		Line Group ID	10	
				Code	N	
			A alal	Feature	Dial	
			Add	Telephone Number	N	
				Line Group ID	11	
		User Rights	User	Name	RS140	
			Short Codes	-	Apply User Rights value	
	User Rights			Short Code table (Code, Telephone Number, Feature, Line Group ID)	Please refer to next section	
			User Rights Membership	Member of this User Rights	All RS140 users	
				Code	002XXXXXXX 82	
		Short		Feature	Dial	
		Code	-	Telephone Number	02N	
	Short			Line Group ID	54: RS140	
	Code			Code	000N;	
		Short		Feature	Dial	
		Code	-	Telephone Number	00N	
				Line Group ID	54: RS140	
orimary IPO	as a templat	es. This appr) it is necessary to save y if we are using User F er.		
Primary PO	ARS	3. Select Templ	 Select first ARS table created in previous steps and click Exp Template (Binary) in top-right window menu. Repeat this action for all other ARS tables created on primary 			

 $^{^{82}}$ It is not possible to add one global entry for immediate national numbers, so such configuration should be repeated for each national numbering pattern 00ZABPQMCDU, where Z is a digit from range 1-9.



Secondary IPO (if used)	ARS	 Chose New from Template (Binary) and select from the list saved ARS table⁸³. Double-click on the Short Code entry within added ARS table and modify Line Group ID with the equivalent number configured on secondary IPO. Repeat the steps above for each ARS table copied from primary IPO. 							
				Code	9N				
Expansion	Short	Short		Feature	Dial				
Gateway	Code	Code	-	Telephone Number	NS225374380 ⁸⁴				
				Line Group ID	3				
Γ	Dial Plan – Ind	coming Call F	Route configurati	on - Incoming call to pl	none user ⁸⁵				
			Standard	Line Group ID	10				
		Incoming Call	Stariuaru	Incoming Number	+33296084361				
		Route 10	Destinations	Destination -> Default Value	4701001 Extn4701001				
		Incoming Call Route 11	Ottom aloual	Line Group ID	11				
-	Incoming Call		Standard	Incoming Number	+33296084361				
	Route		Destinations	Destination -> Default Value	4701001 Extn4701001				
			Standard	Line Group ID	3				
		Incoming Call		Incoming Number	225374381 ⁸⁷				
		Route 3 ⁸⁶	Destinations	Destination -> Default Value	4701001 Extn4701001 ⁸⁸				
Dial Plan –	Dial Plan – Incoming Call Route configuration - Incoming call to destination other than phone user (i.e. voicemail, hunt group)								
	Option	1: Configure	separate entry d	edicated to particular se	ervice				
				Local URI	+33296084362				
				Contact	+33296084362				
Primary IPO	Line	SIP Line 10	SIP URI	Display Name	+33296084362				
				Identity -> Identity	None				
				Identity->Header	P Asserted ID				

⁸³ It is important to add all ARS tables for local PSTN access first, otherwise it will be required to manually select Alternate Route table later.

⁸⁴ Sxxxxxxxx means that provided number is used for CLI in outgoing calls via local PSTN line

⁸⁵ Each user has to have DID number assigned. To route incoming BTIP calls it is required to have SIP URI tab on primary and backup SIP trunk, configuration of which is described in section: SIP trunks configuration.

⁸⁶ Dedicated for local PSTN access (optional)

 $^{^{87}}$ This field can be used to match the called public number with private one.

⁸⁸ Binds public DID with the private extension.



				Forwarding and Twinning -> Send Caller ID	Diversion Header
				Diversion Header	None
				Incoming Group	10
				Max Session	Default=10 Range 1 - 250
				Local URI	+33296084362
				Contact	+33296084362
				Display Name	+33296084362
				Identity -> Identity	None
		SIP Line		Identity->Header	P Asserted ID
		11	SIP URI	Forwarding and Twinning -> Send Caller ID	Diversion Header
				Diversion Header	None
				Incoming Group	11
				Max Session	Default=10 Range 1 - 250
		Incoming Call Route 10	Standard	Line Group ID	10
				Incoming Number	+33296084362
	Incoming Call		Destinations	Destination -> Default Value	Voicemail / Hunt group / etc.
	Route	Incoming Call Route 11	Standard	Line Group ID	11
				Incoming Number	+33296084362
			Destinations	Destination -> Default Value	Voicemail / Hunt group / etc.
		Option2: 0	Configure commo	on entry using Auto	
				Local URI	Auto
				Contact	Auto
				Display Name	Auto
				Identity -> Identity	None
		SIP Line		Identity->Header	P Asserted ID
Primary Lin	Line	10	SIP URI	Forwarding and Twinning -> Send Caller ID	Diversion Header
				Diversion Header	None
				Incoming Group	10
				Max session	Default=10 Range 1 - 250
		SIP Line	SIP URI	Local URI	Auto
		11	OIF UNI	Contact	Auto



				Display Name	Auto
				Identity -> Identity	None
				Identity->Header	P Asserted ID
				Forwarding and Twinning -> Send Caller ID	Diversion Header
				Diversion Header	None
				Incoming Group	11
				Max Session	Default=10 Range 1 - 250
			Standard	Line Group ID	10
		Incoming Call	Stanuaru	Incoming Number	+33296084362
_	Incoming Call	Route 10	Destinations	Destination -> Default Value	Voicemail / Hunt group / etc.
	Route		Standard	Line Group ID	11
		Incoming Call Route 11	Standard	Incoming Number	+33296084362
			Destinations	Destination -> Default Value	Voicemail / Hunt group / etc.
		Dial Plan	configuration fo	r Emergency calls	
	Dial Plan cor	nfiguration for	Emergency call	s – Short Code: Dial En	nergency ⁸⁹
				Code	112
	Short	Short		Feature	
Code			l _	T eature	Dial Emergency
	Code	Code	-	Telephone Number	Dial Emergency 112
	Code	Code	-		
	Code	Code	ADS	Telephone Number	112
	Code	Code	ARS	Telephone Number Line Group ID	112 Blank
Primary	Code	Code	ARS	Telephone Number Line Group ID Route Name	112 Blank HQ313-Emergency
Primary IPO	Code	Code		Telephone Number Line Group ID Route Name Alternate Route	Blank HQ313-Emergency PSTN_for_HQ313
			ARS Add	Telephone Number Line Group ID Route Name Alternate Route Code	Blank HQ313-Emergency PSTN_for_HQ313 N
	Code	Code		Telephone Number Line Group ID Route Name Alternate Route Code Feature	112 Blank HQ313-Emergency PSTN_for_HQ313 N Dial
				Telephone Number Line Group ID Route Name Alternate Route Code Feature Telephone Number	Blank HQ313-Emergency PSTN_for_HQ313 N Dial N
			Add	Telephone Number Line Group ID Route Name Alternate Route Code Feature Telephone Number Line Group ID	112 Blank HQ313-Emergency PSTN_for_HQ313 N Dial N 2090
				Telephone Number Line Group ID Route Name Alternate Route Code Feature Telephone Number Line Group ID Code	112 Blank HQ313-Emergency PSTN_for_HQ313 N Dial N 2090 N

 $^{^{89}}$ If the system uses prefixes for external dialing, the dialing of emergency numbers with and without the prefix should be allowed.

⁹⁰ This value must be different than the one used for standard calls.

⁹¹ This value must be different than the one used for standard calls.



Line	Location	Location	Location	Emergency ARS	HQ313-Emergency
SIP Line				Local URI	· ·
SIP Line				Contact	'
SIP Line 10 SIP URI Identity->Header P Asserted ID				Display Name	· ·
Line				Identity->Identity	None
Line		SIP Line	QID I IDI	Identity->Header	P Asserted ID
Line		10	SIF UNI	Twinning -> Send	Diversion Header
Line Line				Diversion Header	None
Line Line Line Line Line Local URI Example:				Incoming Group	0
Line Line Line Line Local URI Example:				Outgoing Group	20 ⁹²
Local URI Example:	Lina			Max session	
SIP Line SIP Line SIP URI SIP URI SIP URI SIP URI Example:	LINE	<u> </u>		Local URI	· ·
SIP Line				Contact	· ·
SIP Line 11 SIP URI Identity->Header P Asserted ID Forwarding and Twinning -> Send Caller ID Diversion Header None Incoming Group Outgoing Group 2193 Max Session Default=10 Range 1 - 250 Code Secondary IPO (if used) SIP URI Identity->Header P Asserted ID Diversion Header None Incoming Group Outgoing Group Telephone Number				Display Name	1
Secondary IPO (if used) SIP URI Forwarding and Twinning -> Send Caller ID Diversion Header None Incoming Group Outgoing Group 21 ⁹³ Max Session Default=10 Range 1 - 250 Code Code Telephone Number 112 Line Group ID Blank Route Name HQ313-Emergency				Identity->Identity	None
Secondary IPO (if used) Forwarding and Twinning -> Send Caller ID Diversion Header None Incoming Group Outgoing Group Code Code Short Code Secondary IPO (if used) None Incoming Group Outgoing Group Code Code 112 Feature Dial Emergency Telephone Number 112 Line Group ID Blank Route Name HQ313-Emergency			QID I IDI	Identity->Header	P Asserted ID
Secondary IPO (if used) Incoming Group 0 Outgoing Group 21 ⁹³ Max Session Default=10 Range 1 - 250 Code 112 Feature Dial Emergency Telephone Number 112 Line Group ID Blank Route Name HQ313-Emergency			Sii Orti	Twinning -> Send	Diversion Header
Secondary IPO (if used) Secondary ARS ARS Outgoing Group 2193 Outgoing Group 2193 Code 112 Feature Dial Emergency Telephone Number 112 Line Group ID Blank Route Name HQ313-Emergency				Diversion Header	None
Secondary IPO (if used) Short Code Short Code Short Code ARS ARS ARS Max Session Default=10 Range 1 - 250 Code 112 Feature Dial Emergency Telephone Number 112 Line Group ID Blank Route Name HQ313-Emergency				Incoming Group	0
Secondary IPO (if used) Short Code Short Code Short Code Short Code ARS ARS ARS Range 1 - 250 Code 112 Feature Dial Emergency Telephone Number 112 Line Group ID Blank Route Name HQ313-Emergency				Outgoing Group	21 ⁹³
Secondary IPO (if used) Short Code Short Code Short Code Short Code Feature Telephone Number Line Group ID Blank Route Name HQ313-Emergency				Max Session	
Secondary IPO (if used) Secondary IPO ARS ARS SHOTT Code Code Telephone Number Line Group ID Blank Route Name HQ313-Emergency				Code	112
IPO (if used) Line Group ID Blank Route Name HQ313-Emergency		Short	_	Feature	Dial Emergency
(if used) Line Group ID Blank Route Name HQ313-Emergency	Code	Code	-	Telephone Number	112
ARS ARS ARS				Line Group ID	Blank
Alternate Route PSTN_for_HQ313	ARS	ARS	ΔRS	Route Name	HQ313-Emergency
	AUO	AHO	AI IO	Alternate Route	PSTN_for_HQ313

⁹² This value must equal the one configured under emergency ARS on first position!

⁹³ This value must equal the one configured under emergency ARS on second position!



				Code	N
					N
			Add	Feature	Dial
				Telephone Number	N
				Line Group ID	120 ⁹⁴
				Code	N
			Add	Feature	Dial
			Add	Telephone Number	N
				Line Group ID	121 ⁹⁵
	Location	Location	Location	Emergency ARS	HQ313-Emergency
				Local URI	Example: +33296083900
				Contact	Example: +33296083900
		SIP Line 10	SIP URI	Display Name	Example: +33296083900
	Line			Identity->Identity	None
				Identity->Header	P Asserted ID
				Forwarding and Twinning -> Send Caller ID	Diversion Header
				Diversion Header	None
				Incoming Group	0
				Outgoing Group	120 ⁹⁶
				Max session	Default=10 Range 1 - 250
				Local URI	Example: +33296083900
				Contact	Example: +33296083900
				Display Name	Example: +33296083900
		SIP Line	SIP URI	Identity->Identity	None
		11		Identity->Header	P Asserted ID
				Forwarding and Twinning -> Send Caller ID	Diversion Header
				Diversion Header	None
				Incoming Group	0
	1	1	1	<u> </u>	1

 $^{^{94}}$ This value must be different than the one used for standard calls.

⁹⁵ This value must be different than the one used for standard calls.

 $^{^{96}}$ This value must equal the one configured under emergency ARS on first position!



				Outgoing Group	121 ⁹⁷
				Max Session	Default=10 Range 1 - 250
	U	lser / Extensi	on creation – ma	nual for IP endpoints ⁹⁸	
				Name	Extn3130001
				Password	password ⁹⁹
			User	Audio Conference PIN	PIN
	User	User		Extension	3130001
Primary IPO				Profile	Basic User / Power User ¹⁰⁰
0			Telephony -> Supervisor Settings	Login Code	login code ¹⁰¹
	Extension	H.323 / SIP Extension	Manager will automatically prompt for new VoIP extension creation when saving User part and will be filled with all necessary information.		
		-	Extn	Phone Password	Password ¹⁰²
User / Exte	nsion creatio	n - Public nu	mbers assignme	nt: NDI number declara	tion for non-DID users
				SIP Name	Example: +33296084360
Primary IPO	User	User	SIP	SIP Display Name (Alias)	Example: +33296084360
				Contact	Example: +33296084360
User / Exte	ension creation	on - Public nu	umbers assignme	ent: NDI number declara	ation for DID users ¹⁰³
				SIP Name	Example: +33296084361
Primary IPO	User	User	SIP	SIP Display Name (Alias)	Example: +33296084361
				Contact	Example: +33296084361
	Use	er / Extensior	creation - The	e "NoUser" configuration	n

⁹⁷ This value must equal the one configured under emergency ARS on second position!

 $^{^{\}rm 98}$ Below values are an examples and should be treated only as a common guidelines for new user creation

 $^{^{\}rm 99}$ Password provided here will be used only for user login to applications like One-X Portal or One-X Mobile.

 $^{^{\}rm 100}$ Power user allows to use additional features like Softphone or Telecommuter mode. Separate license is required.

¹⁰¹ Login code provided here will be used for phone's registration. Not obligatory.

¹⁰² This code will be used by H.323 phone users to login

¹⁰³ Each user has to have DID number assigned, so configuration should be repeated for each user.



Primary IPO	User	NoUser	Source Numbers	Source Number	MEDIA_DISABLE_RF C2833_ON_IPO ¹⁰⁴
Secondary IPO (if used)	User	NoUser	Source Numbers	Source Number	MEDIA_DISABLE_RF C2833_ON_IPO ¹⁰⁵
Expansion Gateway (if used)	User	NoUser	Source Numbers	Source Number	MEDIA_DISABLE_RF C2833_ON_IPO ¹⁰⁶

 $^{^{104}}$ Configuration of NUSN is mandatory to have direct media for H.323 and DECT users registered to local gateways

¹⁰⁵ Configuration of NUSN is mandatory to have direct media for H.323 and DECT users registered to local gateways

 $^{^{106}}$ Configuration of NUSN is mandatory to have direct media for H.323 and DECT users registered to local gateways



6 ECOSYSTEM AND ENDPOINTS CONFIGURATION

6.1 Avaya Communicator for Windows

Access type: application.

Avaya Communicator for Windows					
		Server address	Primary FQDN		
Communi	ammuni Carasa	Server port	5060		
cator for	Server	Transport type	TCP		
windows		Domain	IPO's Domain Name		
	Conference	Conference server address	Example 6.3.13.1		

6.2 Avaya B179 Conference Station

Access type: B179 Conference Station's Administration web page.

Menu	Tab	Parameter	
	Codec configuration	n – G.722	
Settings	Media	Codec priorities: G722: 4 – High G711 Alaw: 3 G711 Ulaw: 0 – Disabled G729: 0 – Disabled	
	SIP setting	s	
	Enable account	YES	
	Account name	Extn3133102	
	User	3133102	
Primary Account	Registrar	Primary IPO IP address	
	Realm	*	
	Autentication name	3133102	
	Password	Password	
	Enable account	YES	
	Account name	Extn3133102	
Fallback Account	User	3133102	
	Registrar	Secondary IPO IP address or Local GW IP address	
	Realm	*	



Autentication name	3133102
Password	Password

6.3 Avaya DECT IP Base Station

Access type: DECT IP Base Station Administration web page.

Menu	Tab	Parameter	Value			
	LAN configuration					
	DHCP	Mode	disabled			
		IP Address	IPBS static IP address			
LAN	IP	Network Mask	255.255.255.0			
	"	Default Gateway	default gateway's IP address			
		DECT configuration				
	Master	Mode	Active * restart required			
		Name	IPBS			
	Radio	Password	password			
		Master IP Address	127.0.0.1			
		Authentication Code	1234 ¹⁰⁷			
	Air Sync	Sync Mode	Master * restart required			
	System	System Name	DECT			
DECT		Password	password ¹⁰⁸			
		Confirm password	password			
		Subscriptions	With User AC			
	Master	PBX	IPO			
	iviasiei	Protocol	H.323/XMobile			
		Name	Trunk1 (default)			
	Trunks	Local Port	1720 (default)			
	Truins	CS IP Address	primary IPO's IP address			
		CS Port	1720 (default)			
	SARI	SARI	license number ¹⁰⁹			
	PF	ROVISIONING configuration				
Services	Provisioning	Current view	Primary			
OCI VICCO	i rovisioriirig	Enable	Checked			

 $^{^{107}}$ Authentication code has to match the one configured on primary IPO for DECT line under Authentication Code

¹⁰⁸ The same password has to be configured as in Master tab

¹⁰⁹ License number has to match the one configured on primary IPO for DECT line under SARI/PARK



		PBX IP Address	IP address Primary IPO			
		User Name	IPDECTService ¹¹⁰			
		Password	Password ¹¹¹			
		Fassword	reset required			
	DECT configuration for AIWS					
UNITE	Device Management	Unite IP Address	AIWS' IP address			
		HTTP Client configuration				
Services	HTTP Client	Password	Password ¹¹²			
	Sw	ritch Resilience configuration				
	Provisioning	Current view	Redundant			
		Enable	Checked			
Services		PBX IP Address	IP address Backup IPO			
Oel vices		User Name	IPDECTService ¹¹³			
		Password	Password ¹¹⁴			
			reset required			
	Master	PBX Resiliency	Checked			
		Status Inquiry period	30 ¹¹⁵			
		Supervision timeout	120 ¹¹⁶			
DECT	Trunks	Redundant Trunks -> Name	Trunk2 (default)			
	TTUTINS	Local Port	1720 (default)			
		CS IP Address	backup IPO's IP address			
		CS Port	1720 (default)			

6.4 Avaya One-X Portal

Access type: IP Office Manager application.

Menu	Submenu	Parameter	Value

¹¹⁰ "User Name" must be the same as in settings on IPO Manager – go to Security Settings -> Service Users -> IPDECTService

^{111 &}quot;Password" must be the same as in settings on IPO Manager – go to Security Settings -> Service Users -> IPDECTService

¹¹² Password the same as for Provisioning

¹¹³ "User Name" must be the same as in settings on IPO Manager for backup server – go to Security Settings -> Service Users -> IPDECTService

^{114 &}quot;Password" must be the same as in settings on IPO Manager for backup server – go to Security Settings -> Service Users -> IPDECTService

¹¹⁵ Value for "Status Inquiry period" should be the same as in settings on IPO – go to IP DECT Line.

¹¹⁶ Value for "Supervision timeou"t should be the same as in settings on IPO – go to IP DECT Line.



Drimon IDO	Primary IPO LAN1 -> VOIP	SIP Registrar FQDN	Primary FQDN
Fillinary IFO		SIP Domain Name	IPO's Domain Name
Secondary IPO	I AN1 -> VOIP	SIP Registrar FQDN	Secondary FQDN
	LAINT -> VOIP	SIP Domain Name	IPO's Domain Name

Access type: One-X Portal Administration web page.

Menu	Submenu	Parameter	Value
		IM/Presence Server -> XMPP Domain Name	IPO's Domain Name
		Resiliency -> Failover	Enebled
Pimary	Configuration	Resiliency -> Failover Detection Time	3
One-x Portal	Configuration	Resiliency -> Failback	Automatic
		HOST Domain Name -> Primary HOST Domain Name	Primary FQDN
		HOST Domain Name -> Secondary HOST Domain Name	Secondary FQDN
	Configuration	IM/Presence Server -> XMPP Domain Name	IPO's Domain Name
		Resiliency -> Failover	Enebled
Secondary		Resiliency -> Failover Detection Time	3
One-x Portal		Resiliency -> Failback	Automatic
		HOST Domain Name -> Primary HOST Domain Name	Primary FQDN
		HOST Domain Name -> Secondary HOST Domain Name	Secondary FQDN



6.5 Avaya One-X Mobile

Access type: One-X Mobile Preferred for Android application installed on mobile device.

Menu	Submenu	Parameter	Value
O. Win va		Server ID	IPO Domain Name (example: ipo.labobs.com)
	Server ID and user account	Username	Extn3130001
Settings		Password	password ¹¹⁷
	Voice Over IP	Voice Over IP	Checked

Orange SA au capital de 10 595 541 532 € 78 rue Olivier de Serres 75505 Paris Cedex 15 380 129 866 RDC Paris

¹¹⁷ Password used to login.