TECHNICAL GUIDE to access Business Talk IP SIP IPBX Avaya IP Office

Versions addressed in this guide : 10.1 SP1

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

Version of 14/09/2018

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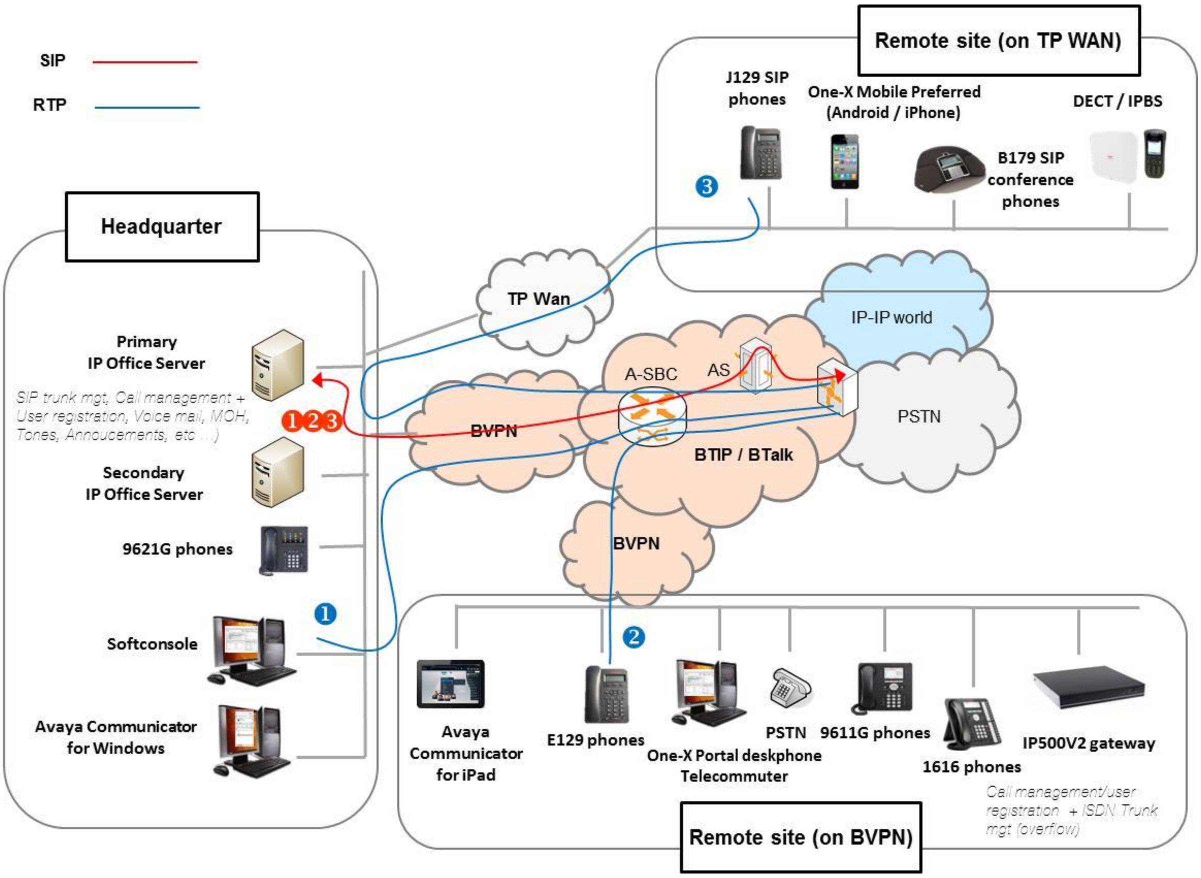
Goal of this document

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:
    - 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 **1** in RS	0 in HQ **1** in RS	0 in HQ **1** in RS
1 offnet call from/to a remote site (RS) on TP Wan	0 in HQ **1** in RS	**1** 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 RS	**1** in HQ **1** in RS	0 in HQ **1** in RS
1 offnet call from/to a remote site **after transfer/forward to BTIP**	0 in HQ **0** in RS	0 in HQ **0** in RS	0 in HQ **2** in RS
1 **forced onnet** call from Headquarter to a remote site (= through Business Talk IP infrastructure)	**2** in HQ **2** in RS	**1** in HQ **1** in RS	**0** in HQ **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.

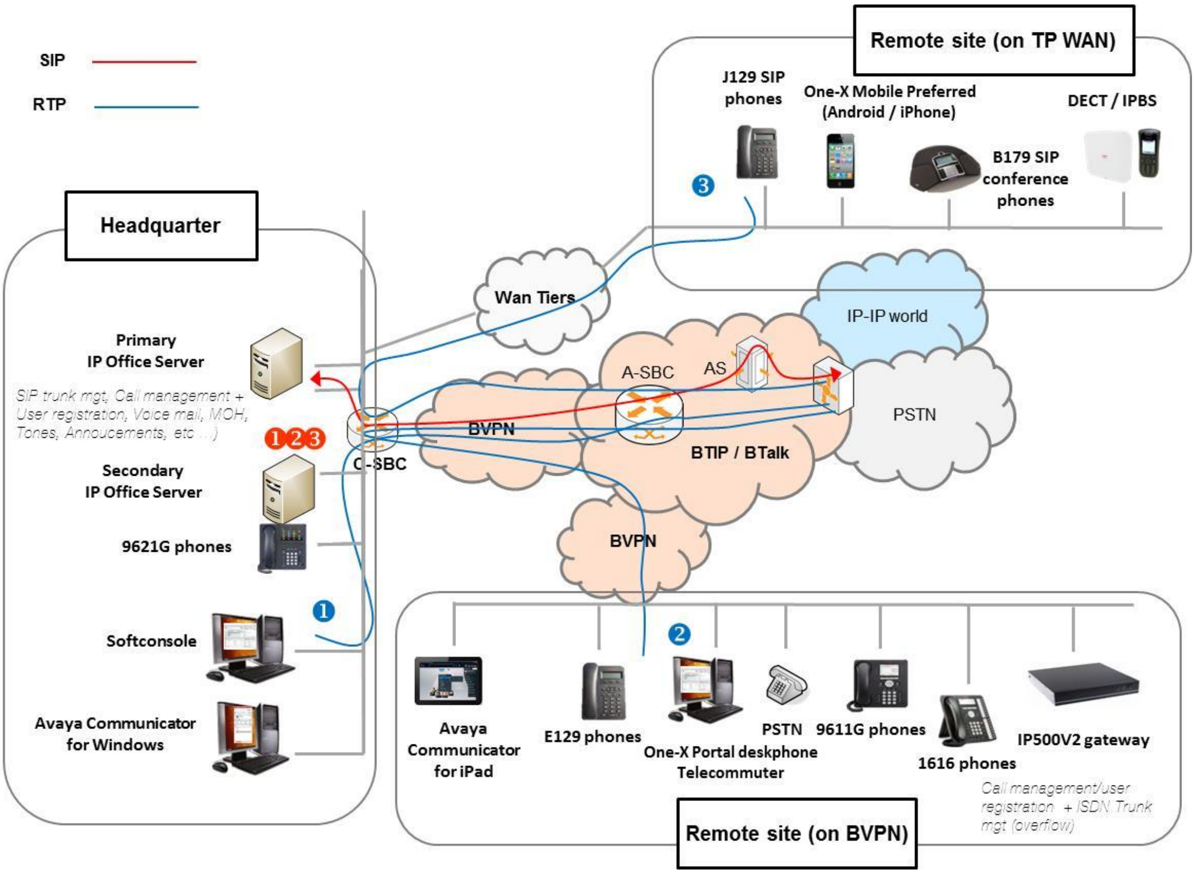
All users are registered initially to a nominal central server. Then in case of failure of the primary server:

- HQ users register to the backup server located near the nominal server or distant from the nominal server
- 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
-

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":
  - for the Headquarter site, media flows are routed through the SBC and the main BVPN connection
  - 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		
	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 **1** in RS	2 in HQ **1** in RS	0 in HQ **1** in RS
1 offnet call from/to a remote site (RS) on TP Wan	0 in HQ **1** in RS	**1** 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 RS	**3** in HQ **1** in RS	0 in HQ **1** in RS
1 offnet call from/to a remote site **after transfer/forward to BTIP**	0 in HQ **0** in RS	0 in HQ*/**3** in HQ** **0** in RS	0 in HQ **2** in RS
1 **forced onnet** call from Headquarter to a remote site (= through Business Talk IP infrastructure)	**2** in HQ **2** in RS	**3** in HQ **1** in RS	**0** in HQ **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) architecture without Customer SBC	Level of Service	Customer IP addresses used by the service	
		Nominal	Backup
1 IPO Server (Call Server)	No redundancy 1 single call server Note : media gateway optional - used for local PSTN access only	IPO IP@	N/A
Redundancy	Local redundancy: 2 call servers (active/standby) hosted by one physical site OR Site redundancy: 2 call servers (active/standby) hosted by 2 different physical sites, but connected to same Orange a-sbc	IPO1 IP@	IPO2 IP@
Remote Site (RS) architecture without Customer SBC	**Level of Service**	**Nominal**	**Backup**
Remote site with media gateway (500v2)	Local site survivability and trunk redundancy via PSTN only	N/A	N/A
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	cSBC @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	cSBC1 @IP cSBC2 @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 geo-redundancy	cSBC VIP @IP

3 CERTIFIED SOFTWARE and HARDWARE versions

3.1 Release 10.1 SP1

SIP User Agent		IP Office 10.1.0.1.0 build 3
IP-PBX Components	IP Office Server Edition	10.1.100.3
	IP Office Core Switch	10.1.100.3
Proprietary phones	1603, 1608, 1616	1.350B
	9608, 9611G, 9621G, 9641G, 9641GS	6.6.5.06
SIP phones	B179 SIP conference phones	2.4.1.5
	E129 SIP IP Phones	1.25.2.26
	J129 SIP IP Phones	1.1.0.0.15
Softphone	Communicator for Windows	2.1.4.0.274
	Communicator for iPad	2.0.6
DECT	3720, 3725,3740,3745	4.3.32
	DECT R4 - IPBS 1-IPBS2	10.0.5
Attendant	IP office Softconsole	10.0.0.3.0.Build 1
Voice mail	VoiceMail Pro	10.1.100.6
Unified Communications	One-X Mobile Preferred Edition for Android	10.0.0.5.220
	One-X Portal	10.1.120.30
	One-X Mobile Preferred Edition for iOS	4.1.8 - 763
Media GW	IP500v2 (IP Office Core Switch)	10.1.100.3
VoIP	Cisco VoIP GW	IOS as of 15.5(3)S6
	OneAccess BLB VoIP GW	OneOS as of V5.2R2E4_HA2

Specific	FAX	Not Supported

4 SIP TRUNKING CONFIGURATION CHECKLIST

The checklist below presents all the steps of configuration required for interoperability between BTIP/BT and Avaya IP Office IPBX.

4.1.1 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[1]	System	-	System	Locale	France2 (French)
System configuration – DSCP configuration					
Every platform in the solution	System	-	LAN1 -> VoIP	DSCP (Hex) / DSCP	B8 / 46
				Video DSCP (Hex) / Video DSCP	88 / 34
				SIG DSCP (Hex) / SIG DSCP	B8 / 46
DHCP configuration offer					
Primary IPO	System	-	LAN1 -> DHCP Poll	Start address	Start IP address
				Subnet Mask	Subnet Mask
				Default Router	Router IP address
				Pool size	DHCP pool size
Codec configuration					
Every platform in the solution	System Line	-	Telephony -> Telephony	Companding Law	A-Law
				High Quality Conferencing	Checked
			VoIP	Ignore DTMF	Checked

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

				Mismatch For Phones	
				RFC2833 Default Payload	101
				Default Codec Selection -> Selected	G.722 64K G.711 ALAW 64K
Call Admition Control & Location configuration[2]					
Solution level	Location	Location	Location	Location Name	RS140
				Subnet Address	6.201.40.0
				Subnet Mask	255.255.255.0
				Parent Location for CAC	<None>
				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	HQ313
Fallback configuration[3]					
Primary IPO	Location	Location	Location	Fallback System	Local GW's IP address
SCN lines configuration					
Primary IPO[4]	Line	IP Office line [5]	Line	Outgoing Group ID	99998
				Transport Type	Proprietary
				Networking Level	SCN
				Gateway -> Address	Backup IPO's IP address
				Gateway -> Location	Location name
				SCN Resiliency Options -> Supports Resiliency	Checked

[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

				- 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
		IP Office Line [6]	Line	Outgoing Group ID	99901 - 99930
				Transport Type	Proprietary
				Networking Level	SCN
				Gateway -> Address	Local GW's IP address
				Gateway -> Location	Location name
				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
Secondary IPO (if used)[7]	Line	IP Office line [8]	Line	Outgoing Group ID	99999
				Transport Type	Proprietary
				Networking Level	SCN
				Gateway -> Address	Primary IPO's IP address
				Gateway -> Location	Location name
				SCN Resiliency Options -> Supports Resiliency	Checked
				- Backs up my IP Phones	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 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
		IP Office Line [9]	Line	Outgoing Group ID	99901 - 99930
				Transport Type	Proprietary
				Networking Level	SCN
				Gateway -> Address	Local GW's IP address
				Gateway -> Location	Location name
				SCN Resiliency Options -> Supports Resiliency	Unchecked
			VoIP settings	Allow Direct Media Path	Checked
Expansion Gateway	Line[10]	IP Office line [11]	Line	Outgoing Group ID	99999
				Transport Type	Proprietary
				Networking Level	SCN
				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

		IP Office Line [12]	Line	Outgoing Group ID	99998
				Transport Type	Proprietary
				Networking Level	SCN
				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 – local PSTN access					
Expansion Gateway	Line	PRI 30 (Universal)[13]	PRI line	Incoming Group ID	3
				Outgoing Group ID	3
SIP Trunks configuration – Global settings					
Primary IPO	System	-	LAN1 -> VoIP	SIP Trunks Enable	Checked
				SIP Registrar Enable	Checked
				Media Connection Preservation	Enabled
				Inhibit Off-Switch Forward/Transfer	Unchecked
Secondary IPO (if used)	System	-	LAN1 -> VoIP	SIP Trunks Enable	Checked
				SIP Registrar Enable	Checked
				Media Connection Preservation	Enabled
				Inhibit Off-Switch Forward/Transfer	Unchecked
SIP Trunks configuration – SIP line					
Primary IPO	Line	SIP Line	SIP Line	Line Number	10
				Local Domain Name	Primary IPO's IP address
				Location	Cloud

[12] SCN line to secondary server

[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	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
			SIP URI	Local URI	Use Internal Data
				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	10
				Outgoing Group	10
				Max Sessions	Default=10 Range 1 - 250
			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
			SIP Advanced	Use + for International	On/Off [14]
				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
				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_AVAIL=480
		SIP Line	SIP Line	Line Number	11

[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).

				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	backup SBC's IP address
				Layer 4 Protocol	UDP
				Network Configuration -> Use Network Topology Info	None
				Send Port	5060
				Listen Port	5060
			SIP URI	Local URI	Use Internal Data
				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	11
				Outgoing Group	11
				Max Sessions	Default=10 Range 1 - 250
			VoIP	Codec Selection	Custom

				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
			SIP Advanced	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
				Media -> P-Early-Media Support	All
				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 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 [16]

[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).

				Location limit	
				Call Control -> Suppress Q.850 Reason Header	Checked
			Engineering	Custom String	SLIC_NO_USER_AVAIL=480
Secondary IPO (if used)	Line	SIP Line	SIP Line	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
				Check OOS	Checked
				Session Timers -> Refresh Method	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
			SIP URI	Local URI	Use Internal Data
				Contact	Use Internal Data
				Display Name	Use Internal Data
				Identity->Identity	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.

				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
			VoIP	Codec Selection	Custom
				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
			SIP Advanced	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
				Media -> Media Connection Preservation	System
				Media -> Indicate HOLD	Checked
				Call Control -> Call Initiation Timeout (s)	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).

				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[18]
				Call Control -> Suppress Q.850 Reason Header	Checked
			Engineering	Custom String	SLIC_NO_USER_AVAIL=480[19]
		SIP Line	SIP Line	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
				Check OOS	Checked
				Session Timers -> Refresh Method	Reinvite
				Session Timers -> Timer (seconds)	14880
				Redirect and Transfer -> Incoming Supervised REFER	Never
				Redirect and Transfer -> Outgoing Supervised REFER	Never
			Transport	ITSP Proxy Address	backup SBC's IP address

[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.

				Layer 4 Protocol	UDP
				Network Configuration -> Use Network Topology Info	None
				Send Port	5060
				Listen Port	5060
			SIP URI	Local URI	Use Internal Data
				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	111
				Outgoing Group	111
				Max Sessions	Default=10 Range 1 - 250
			VoIP	Codec Selection	Custom
				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
			SIP Advanced	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

[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).

				Add UUI Header to redirected calls	Checked
				Media -> P-Early-Media Support	All
				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 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
			Engineering	Custom String	SLIC_NO_USER_AVAIL=480
DECT line configuration					
Primary IPO	Line	IP DECT Line	Gateway	Enable Provisioning	Checked
				SARI/PARK	PARK license key[22]
				Subscriptions	Auto-Create / Preconfigured
				Authentication Code	1234[23]
				Enable Resiliency	Checked
			VoIP	Gateway IP Address	DECT IPBS's IP address
				Allow Direct Media Path	Checked

[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.

[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

				Codec Selection	Custom
				Codec Selected	G.711 ALAW 64K
Security settings for IP DECT					
Primary IPO	Security	Services	HTTP -> Service details	Service Security Level	Unsecure + Secure
		Right Group	IPDECT Group -> HTTP	DECT R4 Provisioning	Checked
		Service Users	IPDECTService -> Service User Details	Name	IPDECTService
				Password	password
				Account status	Enabled
				Account Expiry	No Account Expiry
				Right Group Membership	IPDECT Group
Dial Plan configuration[24]					
Dial Plan – General dialing configuration					
Primary IPO	System	-	Telephony ->Telephony	Dial Delay Time (secs)	10
				Dial Delay Count	0
				Default No Answer Time	15
Secondary IPO (if used)	System	-	Telephony ->Telephony	Dial Delay Time (secs)	10
				Dial Delay Count	0
				Default No Answer Time	15
Dial Plan – Short Codes and ARS configuration when local PSTN access is not used					
Primary IPO	ARS	ARS1	ARS	Route Name	Main
			Add…	Code	N
				Feature	Dial
				Telephone Number	N
				Line Group ID	10
			Add…	Code	N
				Feature	Dial
				Telephone Number	N
				Line Group ID	11

[24] This is common configuration. It may be required to adjust dial plan configuration per particular system.

	Short Code	Short Code	-	Code	002XXXXXXXX[25]
				Feature	Dial
				Telephone Number	02N
				Line Group ID	50: Main
		Short Code	-	Code	000N;
				Feature	Dial
				Telephone Number	00N
				Line Group ID	50: Main
Secondary IPO (if used)	ARS	ARS1	ARS	Route Name	Main
			Add…	Code	N
				Feature	Dial
				Telephone Number	N
				Line Group ID	110
			Add…	Code	N
				Feature	Dial
				Telephone Number	N
				Line Group ID	111
	Short Code	Short Code	-	Code	002XXXXXXXX[26]
				Feature	Dial
				Telephone Number	02N
				Line Group ID	50: Main
		Short Code	-	Code	000N;
				Feature	Dial
				Telephone Number	00N
				Line Group ID	50: Main
Dial Plan – Short Codes and ARS configuration when local PSTN access is used[27]					
Primary IPO	ARS	ARS2[28]	ARS	Route Name	PSTN_for_HQ313
			Add…	Code	N

[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.

[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.

[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.

				Feature	Dial
				Telephone Number	9N
				Line Group ID	99901
		ARS1	ARS	Route Name	HQ313
				Alternate Route	PSTN_for_HQ313
			Add…	Code	N
				Feature	Dial
				Telephone Number	N
				Line Group ID	10
			Add…	Code	N
				Feature	Dial
				Telephone Number	N
				Line Group ID	11
	User Rights	User Rights	User	Name	RS140
			Short Codes	-	Apply User Rights value
				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 Code	Short Code	-	Code	002XXXXXXXX[29]
				Feature	Dial
				Telephone Number	02N
				Line Group ID	54: RS140
		Short Code	-	Code	000N;
				Feature	Dial
				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	1. Select first ARS table created in previous steps and click **Export as Template (Binary)** in top-right window menu. 2. Repeat this action for all other ARS tables created on primary IPO.			

[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.

Secondary IPO (if used)	ARS	1. Chose **New from Template (Binary)** and select from the list saved ARS table[30]. 2. Double-click on the Short Code entry within added ARS table and modify Line Group ID with the equivalent number configured on secondary IPO. 3. Repeat the steps above for each ARS table copied from primary IPO.			
Expansion Gateway	Short Code	Short Code	-	Code	9N
				Feature	Dial
				Telephone Number	NS225374380[31]
				Line Group ID	3
Dial Plan – Incoming Call Route configuration - Incoming call to phone user[32]					
-	Incoming Call Route	Incoming Call Route 10	Standard	Line Group ID	10
				Incoming Number	+33296084361
			Destinations	Destination -> Default Value	4701001 Extn4701001
		Incoming Call Route 11	Standard	Line Group ID	11
				Incoming Number	+33296084361
			Destinations	Destination -> Default Value	4701001 Extn4701001
		Incoming Call Route 3[33]	Standard	Line Group ID	3
				Incoming Number	225374381[34]
			Destinations	Destination -> Default Value	4701001 Extn4701001[35]
Dial Plan – Incoming Call Route configuration - Incoming call to destination other than phone user (i.e. voicemail, hunt group)					
Option1: Configure separate entry dedicated to particular service					
Primary IPO	Line	SIP Line 10	SIP URI	Local URI	+33296084362
				Contact	+33296084362
				Display Name	+33296084362
				Identity -> Identity	None
				Identity->Header	P Asserted ID

[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.

[33] Dedicated for local PSTN access (optional)

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

[35] 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
		SIP Line 11	SIP URI	Local URI	+33296084362
				Contact	+33296084362
				Display Name	+33296084362
				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
-	Incoming Call Route	Incoming Call Route 10	Standard	Line Group ID	10
				Incoming Number	+33296084362
			Destinations	Destination -> Default Value	Voicemail / Hunt group / etc.
		Incoming Call Route 11	Standard	Line Group ID	11
				Incoming Number	+33296084362
			Destinations	Destination -> Default Value	Voicemail / Hunt group / etc.
Option2: Configure common entry using Auto					
Primary IPO	Line	SIP Line 10	SIP URI	Local URI	Auto
				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	10
				Max session	Default=10 Range 1 - 250
		SIP Line 11	SIP URI	Local URI	Auto
				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
-	Incoming Call Route	Incoming Call Route 10	Standard	Line Group ID	10
				Incoming Number	+33296084362
			Destinations	Destination -> Default Value	Voicemail / Hunt group / etc.
		Incoming Call Route 11	Standard	Line Group ID	11
				Incoming Number	+33296084362
			Destinations	Destination -> Default Value	Voicemail / Hunt group / etc.
Dial Plan configuration for Emergency calls					
Dial Plan configuration for Emergency calls – Short Code: Dial Emergency[36]					
Primary IPO	Short Code	Short Code	-	Code	112
				Feature	Dial Emergency
				Telephone Number	112
				Line Group ID	Blank
	ARS	ARS	ARS	Route Name	HQ313-Emergency
				Alternate Route	PSTN_for_HQ313
			Add…	Code	N
				Feature	Dial
				Telephone Number	N
				Line Group ID	20[37]
			Add…	Code	N
				Feature	Dial
				Telephone Number	N
				Line Group ID	21[38]

[36] If the system uses prefixes for external dialing, the dialing of emergency numbers with and without the prefix should be allowed.

[37] This value must be different than the one used for standard calls.

[38] This value must be different than the one used for standard calls.

	Location	Location	Location	Emergency ARS	HQ313-Emergency
	Line	SIP Line 10	SIP URI	Local URI	Example: +33296083900
				Contact	Example: +33296083900
				Display Name	Example: +33296083900
				Identity->Identity	None
				Identity->Header	P Asserted ID
				Forwarding and Twinning -> Send Caller ID	Diversion Header
				Diversion Header	None
				Incoming Group	0
				Outgoing Group	20[39]
				Max session	Default=10 Range 1 - 250
		SIP Line 11	SIP URI	Local URI	Example: +33296083900
				Contact	Example: +33296083900
				Display Name	Example: +33296083900
				Identity->Identity	None
				Identity->Header	P Asserted ID
				Forwarding and Twinning -> Send Caller ID	Diversion Header
				Diversion Header	None
				Incoming Group	0
				Outgoing Group	21[40]
				Max Session	Default=10 Range 1 - 250
Secondary IPO (if used)	Short Code	Short Code	-	Code	112
				Feature	Dial Emergency
				Telephone Number	112
				Line Group ID	Blank
	ARS	ARS	ARS	Route Name	HQ313-Emergency
				Alternate Route	PSTN_for_HQ313

[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!

			Add…	Code	N
				Feature	Dial
				Telephone Number	N
				Line Group ID	120[41]
			Add…	Code	N
				Feature	Dial
				Telephone Number	N
				Line Group ID	121[42]
	Location	Location	Location	Emergency ARS	HQ313-Emergency
	Line	SIP Line 10	SIP URI	Local URI	Example: +33296083900
				Contact	Example: +33296083900
				Display Name	Example: +33296083900
				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[43]
				Max session	Default=10 Range 1 - 250
		SIP Line 11	SIP URI	Local URI	Example: +33296083900
				Contact	Example: +33296083900
				Display Name	Example: +33296083900
				Identity->Identity	None
				Identity->Header	P Asserted ID
				Forwarding and Twinning -> Send Caller ID	Diversion Header
				Diversion Header	None
				Incoming Group	0

[41] This value must be different than the one used for standard calls.

[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!

				Outgoing Group	121[44]
				Max Session	Default=10 Range 1 - 250
User / Extension creation – manual for IP endpoints[45]					
Primary IPO	User	User	User	Name	Extn3130001
				Password	password[46]
				Audio Conference PIN	PIN
				Extension	3130001
				Profile	Basic User / Power User[47]
			Telephony -> Supervisor Settings	Login Code	login code[48]
	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 [49]
User / Extension creation - Public numbers assignment: NDI number declaration for non-DID users					
Primary IPO	User	User	SIP	SIP Name	Example: +33296084360
				SIP Display Name (Alias)	Example: +33296084360
				Contact	Example: +33296084360
User / Extension creation - Public numbers assignment: NDI number declaration for DID users[50]					
Primary IPO	User	User	SIP	SIP Name	Example: +33296084361
				SIP Display Name (Alias)	Example: +33296084361
				Contact	Example: +33296084361
User / Extension creation - The “NoUser” configuration					

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

[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.

[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.

Primary IPO	User	NoUser	Source Numbers	Source Number	MEDIA_DISABLE_RFC2833_ON_IPO[51]
Secondary IPO (if used)	User	NoUser	Source Numbers	Source Number	MEDIA_DISABLE_RFC2833_ON_IPO[52]
Expansion Gateway (if used)	User	NoUser	Source Numbers	Source Number	MEDIA_DISABLE_RFC2833_ON_IPO[53]

4.1.2 Ecosystem configuration

Avaya Communicator for Windows

Access type: application.

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

B179 Conference Station

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

Menu	Tab	Parameter
Codec configuration – G.722		
Settings	Media	Codec priorities: ▪ G722: 4 – High ▪ G711 Alaw: 3 ▪ G711 Ulaw: 0 – Disabled ▪ G729: 0 – Disabled
SIP settings		
Primary Account	Enable account	YES
	Account name	Extn3133102

[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

	User	3133102
	Registrar	Primary IPO IP address
	Realm	*
	Autentication name	3133102
	Password	Password
Fallback Account	Enable account	YES
	Account name	Extn3133102
	User	3133102
	Registrar	Secondary IPO IP address or Local GW IP address
	Realm	*
	Autentication name	3133102
	Password	Password

DECT IP Base Station

Access type: DECT IP Base Station Administration web page.

Menu	Tab	Parameter	Value
		LAN configuration	
LAN	DHCP	Mode	disabled
	IP	IP Address	IPBS static IP address
		Network Mask	255.255.255.0
		Default Gateway	default gateway's IP address
		DECT configuration	
DECT	Master	Mode	Active * restart required
	Radio	Name	IPBS
		Password	password
		Master IP Address	127.0.0.1
		Authentication Code	1234[54]
	Air Sync	Sync Mode	Master * restart required
	System	System Name	DECT
		Password	password[55]

54 Authentication code has to match the one configured on primary IPO for DECT line under Authentication Code

55 The same password has to be configured as in **Master** tab

		Confirm password	password
		Subscriptions	With User AC
	Master	PBX	IPO
		Protocol	H.323/XMobile
	Trunks	Name	Trunk1 (default)
		Local Port	1720 (default)
		CS IP Address	primary IPO's IP address
		CS Port	1720 (default)
	SARI	SARI	license number[56]
PROVISIONING configuration			
Services	Provisioning	Current view	Primary
		Enable	Checked
		PBX IP Address	IP address Primary IPO
		User Name	IPDECTService[57]
		Password	Password [58] • reset required
DECT configuration for AIWS			
UNITE	Device Management	Unite IP Address	AIWS' IP address
HTTP Client configuration			
Services	HTTP Client	Password	Password[59]
Switch Resilience configuration			
Services	Provisioning	Current view	Redundant
		Enable	Checked
		PBX IP Address	IP address Backup IPO
		User Name	IPDECTService[60]
		Password	Password[61] • reset required
DECT	Master	PBX Resiliency	Checked
	Trunks	Status Inquiry period	30[62]

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

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

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

59 Password the same as for Provisioning

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

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

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

		Supervision timeout	120[63]
		Redundant Trunks -> Name	Trunk2 (default)
		Local Port	1720 (default)
		CS IP Address	backup IPO's IP address
		CS Port	1720 (default)

One-X Portal

Access type: IP Office Manager application.

Menu	Submenu	Parameter	Value
Primary IPO	LAN1 -> VOIP	SIP Registrar FQDN	Primary FQDN
		SIP Domain Name	IPO's Domain Name
Secondary IPO	LAN1 -> VOIP	SIP Registrar FQDN	Secondary FQDN
		SIP Domain Name	IPO's Domain Name

Access type: One-X Portal Administration web page.

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

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

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

One-X Mobile

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

Menu	Submenu	Parameter	Value
Settings	Server ID and user account	Server ID	IPO Domain Name (example: ipo.labobs.com)
		Username	Extn3130001
		Password	password[64]
	Voice Over IP	Voice Over IP	Checked

E129 Phone Configuration

Access type: web page.

Menu	Submenu	Parameter	Value
Account 1	General Settings	Account active	YES
		Account name	Extn3131101

64 Password used to login.

		SIP server	Primary IPO IP address:5060
		Secondary SIP server	Secondary IPO IP address:5060
		SIP User ID	3131101
		Authentication ID	3131101
		Authentication Password	password

5 MAIN SIP features and FUNCTIONAL restrictions

- Codecs:

- G711 A 20 ms only (G729 A not certified)
- G722 may be used for internal calls

- SIP Transport:

- UDP

- Functional restrictions:

- In early media scenarios, Avaya H323 phone cannot send DTMF tones to Orange network before the call has been answered. This lightly affects Orange service like SVA (Service à Valeur Ajoutée) where it is common to play announcements to the caller before the call is answered. Typically the user could not stop the announcement while sending DTMF. This is a well-known restriction of Avaya IP office solution.