Business Talk & BTIP
For IPBX Avaya IP Office

Versions addressed in this guide: Avaya IP Office 11.1 and 11.0

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.

Latest edition: October 2023

Table of Contents

1. Goal of this document

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

2. Certified architectures

2.1 Introduction to architecture components and features

This document describes “only” the main supported architectures either strictly used by our customers or that are used as reference to add specific usages often required in enterprise context (specific ecosystems, redundancy, multi-codec and/or transcoding, recording…).

Please note that Fax communications via Business Talk (International) is not supported by the Orange support teams.

Concerning the fax support in France, due to an IP Office behavior not compliant with BTIP, the usage of analog fax machines, usually connected on vendor gateways (IP500v2) or specific gateways (ex: Mediatrix) is not supported at this time. Evolution request to Avaya was raised in consequence.

Please contact your Orange sales representative to see what possible fax solution can be considered (FaxServer, FaxPLug ...).

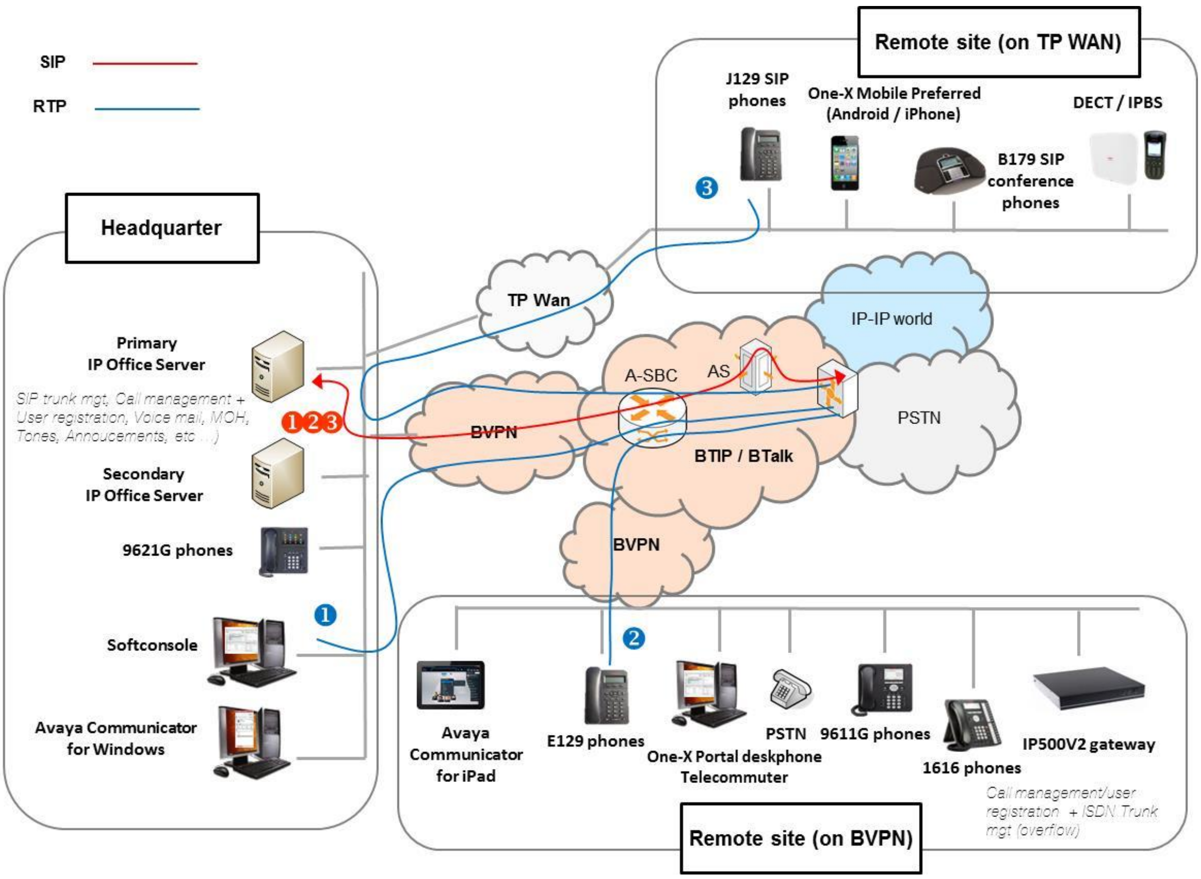
Concerning the Quality of Service, Business VPN and BTIP/Btalk networks trust the DSCP (Differenciated Services Code Point) values sent by customer voice equipment. That’s why Orange strongly recommends setting the IPBX, IP phones and other voice applications with a DiffServ/TOS value* = 46 (or PHB value = EF) at least for media.

“BTIP DROM” architectures are now supported. Dedicated aSBC pairs have been installed in Caribbean and Indian Ocean zones for local calls. For a trunking point of view, the mechanism is similar to “BTIP out of France”, the IPBX must support international dial plans and route local calls to the dedicated aSBC pair.

*cf QoS parameters in the Configuration Checklist → “System configuration – DSCP configuration”.

2.2 SIP trunk on Avaya IP Office over BVPN

2.2.1 Architecture



Notes:

- In the diagram above, the SIP and proprietary internal flows are hidden.

❶ call from/to Headquarter
❷ call from/to remote site (on Business VPN)
❸ call from/to remote site (on Third Party WAN)

- Call flows will be the similar with or without IPO Call Server redundancy.

In this architecture

- All 'SIP trunking' signaling 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 **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

2.2.2 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

2.2.3 Codecs consideration

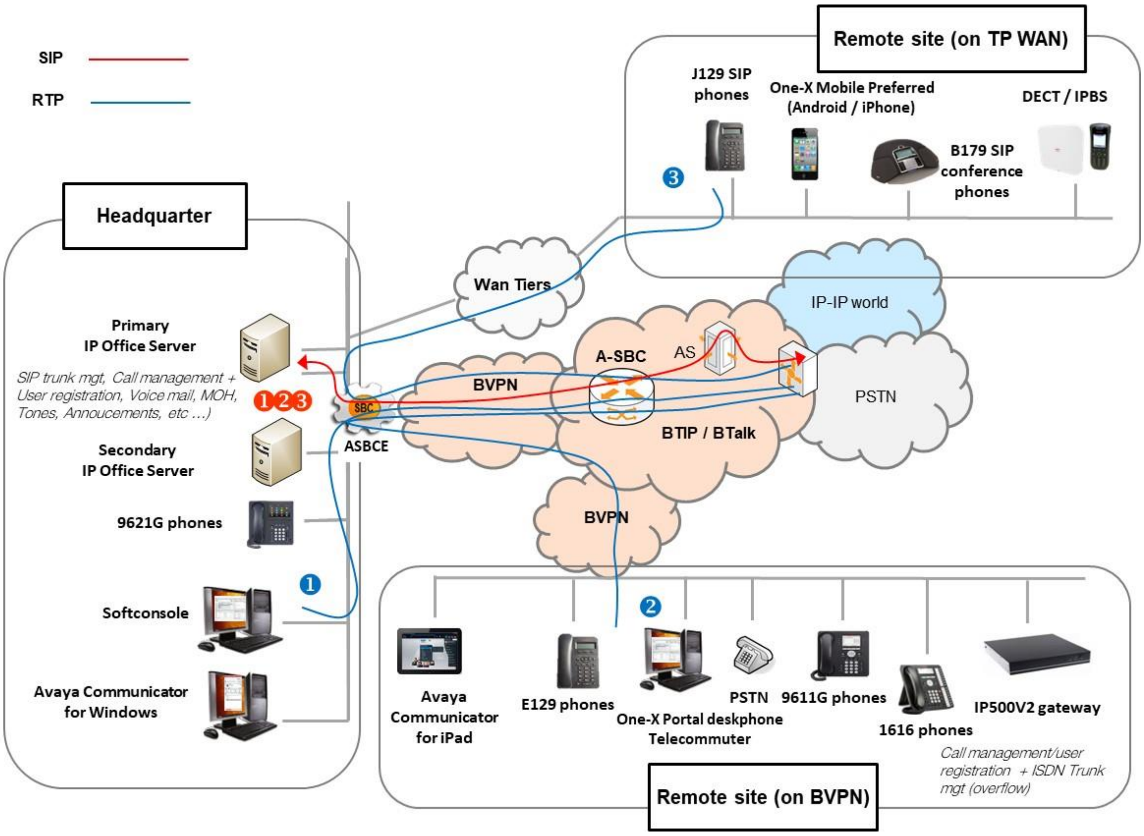
Only G711A and G722 codecs are supported.
G711U can be supported in option.
G729A codec is not certified.

2.2.4 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.

2.3 SIP trunk on customer SBC over BVPN

2.3.1 Architecture



Notes:

- In the diagram above, the SIP and proprietary internal flows are hidden.

❶ call from/to Headquarter
❷ call from/to remote site (on Business VPN)
❸ call from/to remote site (on Third Party WAN)

- Call flows will be the similar with or without IPO Call Server redundancy.

Avaya Session Border Controller for Enterprise (ASBCE) is standard, so doesn't need any specific implementation request.

If the Avaya IPO customer solution is complemented by a SBC equipment, which is not an Avaya SBCE, Orange will offer one of the following approaches:

- A "Certified Border" approach (or "Certified SBC equipment"), if the SBC used is already certified by Orange, regardless of the PBX solution used. Recommendations on this SBC are also available on the Orange Business Services website.
- A “Generic Offer” approach, if the SBC is not certified by Orange. Orange will not be able to give any recommendation on the choice of hardware, software or configuration, but offers a ‘Validation Assistance Service’ for the SBC+PBX architecture.

In this architecture, both 'SIP trunking' and RTP media flows between endpoints and the Business Talk/BTIP are anchored by the enterprise SBC:

- for the Headquarter site, media flows are routed through the enterprise SBC and the main BVPN connection
- for Remote Sites either on BVPN or Third Party WAN, media flows transit through the Headquarter enterprise SBC and use the central BVPN connection (= centralized architecture, cf sizing below).

Warning: 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 enterprise SBC

***if media release is not activated on the enterprise SBC

2.3.2 Resiliency consideration

Secondary ASBCE can be located on the same site as the primary ASBCE or on a remote site.

2.3.3 Codecs consideration

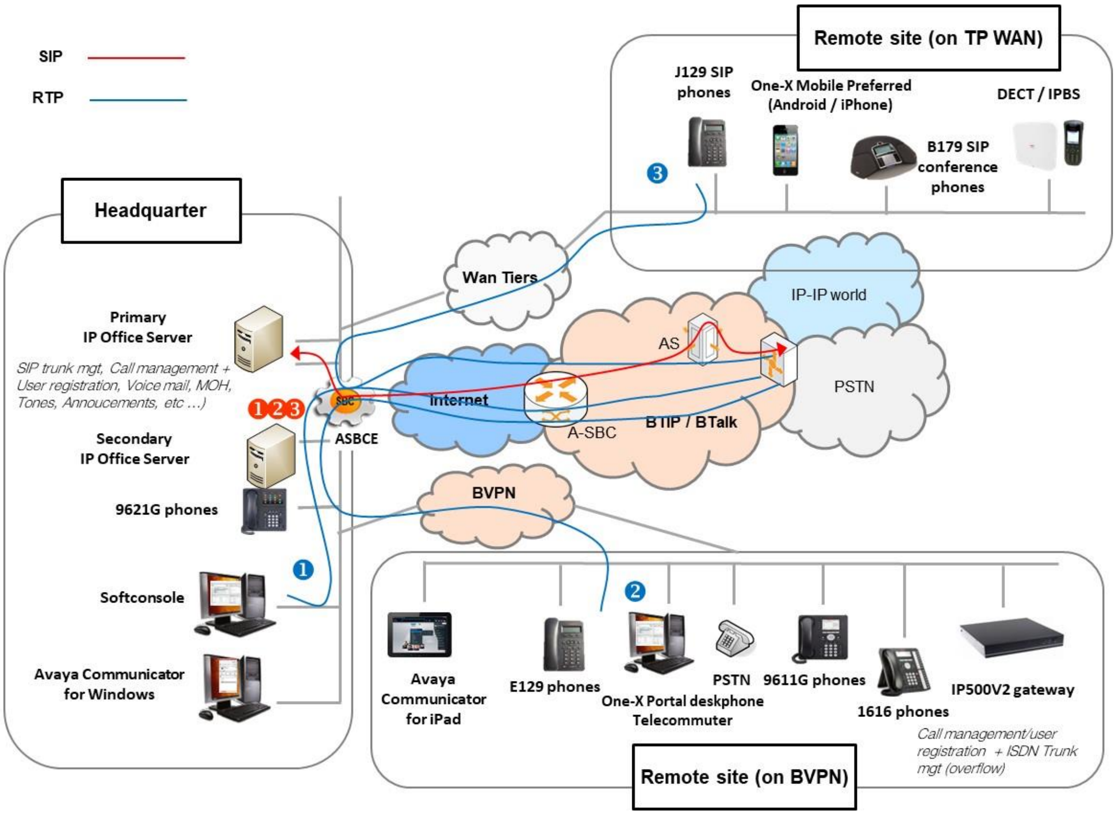
Only G711A and G722 codecs are supported.
G711U can be supported in option.
G729A codec is not certified.

2.3.4 Sizing approach

Specific sizing approach to be considered with ASBCE solution as the RTP flow is not direct between Avaya phones and Orange a-SBC but anchored by the enterprise SBC.

2.4 SIP trunk on customer SBC over Internet

2.4.1 Architecture



Notes:

- In the diagram above, the SIP and proprietary internal flows are hidden.

  ❶ call from/to Headquarter
  ❷ call from/to remote site (on Business VPN)
  ❸ call from/to remote site (on Third Party WAN)

- Call flows will be the similar with or without IPO Call Server redundancy.

  SIP TLS + Secured RTP: all SIP messages and media packets are encrypted on the public internet between Orange and the customer Internet SIP & Media endpoints. This is the level of encryption recommended by default by Orange to ensure security & privacy. Refer to the dedicated configuration section chapter 7 for more details.

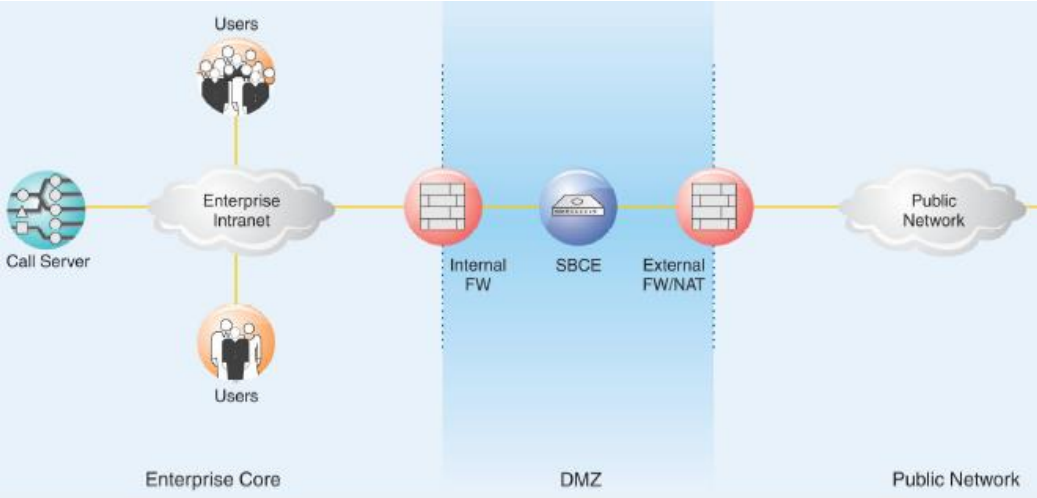
  In this architecture, both 'SIP trunking' and RTP media flows between endpoints and the Business Talk/BTIP are anchored by the enterprise SBC*:

  - For the Headquarter site, media flows are routed through the SBC and the Internet access
  - For Remote Sites, media flows transit through the Headquarter SBC and use the BTIP over Internet (BTIPoI) / Business Talk over Internet (BToI) connection (= centralized architecture).

  * Avaya Session Border Controller is standard, so doesn't need any other specific implementation request.

Note: To avoid any security risk the clients should always install on ASBCE the latest mandatory patch/hotfix released by the Avaya vendor.

Concerning the deployment of the ASBCE, the two-wire topology, also referred to as inline, is the simplest and most basic model.



Avaya SBCE is positioned at the edge of the network in the DMZ. Avaya SBCE is directly inline with the call servers, and protects the enterprise network against all inadvertent and malicious intrusions and attacks.
In this configuration, the Avaya SBCE performs border access control functionality such as internal and external Firewall or Network Address Translation (FW/NAT) traversal, access management and control. These functions are based on domain policies that the user can configure, and intrusion functionality to protect against DoS, spoofing, stealth attacks, and voice SPAM.

The two-wire Avaya SBCE deployment enables TLS encryption of the signaling traffic and SRTP encryption of the media traffic carried over public internet between ASBCE and Orange A-SBC.

An X.509 v3 public key certificate is used to identify the Avaya SBCE when performing a TLS handshake for incoming and outgoing connections.

Media must be anchored on ASBCE to perform media transcoding between internal RTP and external SRTP.

2.4.2 Prerequisites

In order to establish the connection with public interface of A-SBC, several preliminary configuration steps have to be performed. These involve the following:

- Public IP address assignment
- Public DNS record
- Firewall updates
- Certificate updates

- TLS v1.2 cypher suites compliance
- SRTP encryption
- Supported codecs on BTIPoI/BToI

2.4.3 Public IP address assignment

The certified solution is using a public IP address directly configured on ASBCE interface placed within DMZ.

2.4.4 Public DNS record

Orange A-SBC can be reached via Fully Qualified Domain Name (FQDN) type SRV or type A deployed on public DNS. Customer premise ASBCE requires a record on public DNS that enables to reach it using FQDN via public internet. BTIPoI can be reached using FQDN only, whereas BToI can be reached either via public IP address or FQDN.

- BTIPoI supports type SRV & type A for DNS resolution and do not support direct public IP connections.
- BToI supports both public IP and type A for DNS resolution and do not provide any type SRV record connections.

2.4.5 Firewall updates

Firewalls in the way of traffic between ASBCE and A-SBC have to be updated in order to open required ports. BToI and BTIPoI vary concerning the UDP port range.

The media UDP port ranges required by Orange BTIPoI SIP Trunk is **6000-38000** and for Orange BToI SIP Trunk is **6000-20000**.

<table>
<tr><th colspan="5">BTIPoI/BToI port matrix</th></tr>
<tr><th>Source device</th><th>Source ports</th><th>Destination device</th><th>Destination ports</th><th>Purpose</th></tr>
<tr><td>ASBCE public @IP</td><td>Defined Signaling port range on ASBCE: Network & Flows -> Advanced Options
e.g. TCP 51001-55000
Depending on customer context or needs.</td><td>A-SBC public @IP</td><td>TCP 5061</td><td rowspan="2">TLS
SIP
signaling</td></tr>
<tr><td>A-SBC public @IP</td><td>TCP Any</td><td>ASBCE public @IP</td><td>TCP 5061</td></tr>
<tr><td>ASBCE public @IP</td><td>BTIPoI: UDP 6000-38000
BToI: UDP 6000-20000</td><td>A-SBC public @IP</td><td>BTIPoI: UDP 6000-38000
BToI: UDP 6000-20000</td><td rowspan="2">SRTP
media</td></tr>
<tr><td>A-SBC public @IP</td><td>BTIPoI: UDP 6000-38000
BToI: UDP 6000-20000</td><td>ASBCE public @IP</td><td>BTIPoI: UDP 6000-38000
BToI: UDP 6000-20000</td></tr>
</table>

2.4.6 Certificate updates

In order to ensure the security of traffic, public root & intermediate certificates need to be exchanged between ASBCE and Orange A-SBC. ASBCE would require an identity certificate signed by a public root CA certificate (including any intermediate certificates in the path). The customer should send public Root & Intermediate certificates which signed ASBCE identity certificate to OBS to be uploaded on Orange A-SBC in case of using a different Public Certificate Authority on their side. This is described in details in following chapters of ASBCE secure configuration.

In case of different public Root & intermediate certificates used by Orange (Digicert) Customer should retrieve ours which signed Orange A-SBC's certificates and upload them to ASBCE. This is described in detail in following chapters of ASBCE secure configuration.

2.4.7 TLS v1.2 cipher suites compliance

The following cipher suites are supported by Orange SBC for TLS 1.2. Compliant cypher suites with Orange SBC are marked in bold.

- **TLS_ECDHE_RSA_WITH_AES_256_GCM_SHA384 (0xc030)**
- **TLS_ECDHE_RSA_WITH_AES_128_GCM_SHA256 (0xc02f)**
- **TLS_ECDHE_RSA_WITH_AES_256_CBC_SHA384 (0xc028)**
- **TLS_ECDHE_RSA_WITH_AES_128_CBC_SHA256 (0xc027)**
- TLS_DHE_RSA_WITH_AES_128_GCM_SHA256 (0x009e)
- TLS_DHE_RSA_WITH_AES_256_GCM_SHA384 (0x009f)
- TLS_DHE_RSA_WITH_AES_128_CBC_SHA256 (0x0067)
- TLS_DHE_RSA_WITH_AES_256_CBC_SHA256 (0x006b)

Cipher suites supported by ASBCE for TLS 1.2 are listed below. Compliant cipher suites with Orange SBC are marked in bold. At least one ASBCE cipher suite must be compliant with BToI/BTIPoI to work.

- **TLS_ECDHE_RSA_WITH_AES_256_GCM_SHA384 (0xc030)**
- TLS_ECDHE_ECDSA_WITH_AES_256_GCM_SHA384 (0xc02c)
- **TLS_ECDHE_RSA_WITH_AES_256_CBC_SHA384 (0xc028)**
- TLS_ECDHE_ECDSA_WITH_AES_256_CBC_SHA384 (0xc024)
- TLS_ECDHE_RSA_WITH_AES_256_CBC_SHA (0xc014)
- TLS_ECDHE_ECDSA_WITH_AES_256_CBC_SHA (0xc00a)
- TLS_ECDH_RSA_WITH_AES_256_GCM_SHA384 (0xc032)
- TLS_ECDH_ECDSA_WITH_AES_256_GCM_SHA384 (0xc02e)
- TLS_ECDH_RSA_WITH_AES_256_CBC_SHA384 (0xc02a)
- TLS_ECDH_ECDSA_WITH_AES_256_CBC_SHA384 (0xc026)
- TLS_ECDH_RSA_WITH_AES_256_CBC_SHA (0xc00f)
- TLS_ECDH_ECDSA_WITH_AES_256_CBC_SHA (0xc005)
- TLS_RSA_WITH_AES_256_GCM_SHA384 (0x009d)
- TLS_RSA_WITH_AES_256_CBC_SHA256 (0x003d)
- TLS_RSA_WITH_AES_256_CBC_SHA (0x0035)
- TLS_RSA_WITH_CAMELLIA_256_CBC_SHA (0x0084)
- **TLS_ECDHE_RSA_WITH_AES_128_GCM_SHA256 (0xc02f)**
- TLS_ECDHE_ECDSA_WITH_AES_128_GCM_SHA256 (0xc02b)
- **TLS_ECDHE_RSA_WITH_AES_128_CBC_SHA256 (0xc027)**
- TLS_ECDHE_ECDSA_WITH_AES_128_CBC_SHA256 (0xc023)
- TLS_ECDHE_RSA_WITH_AES_128_CBC_SHA (0xc013)

- TLS_ECDHE_ECDSA_WITH_AES_128_CBC_SHA (0xc009)
- TLS_ECDH_RSA_WITH_AES_128_GCM_SHA256 (0xc031)
- TLS_ECDH_ECDSA_WITH_AES_128_GCM_SHA256 (0xc02d)
- TLS_ECDH_RSA_WITH_AES_128_CBC_SHA256 (0xc029)
- TLS_ECDH_ECDSA_WITH_AES_128_CBC_SHA256 (0xc025)
- TLS_ECDH_RSA_WITH_AES_128_CBC_SHA (0xc00e)
- TLS_ECDH_ECDSA_WITH_AES_128_CBC_SHA (0xc004)
- TLS_RSA_WITH_AES_128_GCM_SHA256 (0x009c)
- TLS_RSA_WITH_AES_128_CBC_SHA256 (0x003c)
- TLS_RSA_WITH_AES_128_CBC_SHA (0x002f)
- TLS_RSA_WITH_CAMELLIA_128_CBC_SHA (0x0041)

ASBCE and A-SBC will negotiate the most secure matched cipher suite (TLS_ECDHE_RSA_WITH_AES_256_GCM_SHA384) to establish TLS connection.

2.4.8 SRTP encryption on BTIPoI/BToI

Media encryption preferred format: AES_CM_128_HMAC_SHA1_80

2.4.9 Supported codecs on BTIPoI/BToI

Supported codec is G.711A (20ms) for BTIPoI and BToI.
G.711u (20ms) can be requested on specific case for BToI.

3. 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.

3.1 Architecture without "Customer SBC" over BVPN

<table>
<tr><th rowspan="2">Head Quarter (HQ) architecture</th><th rowspan="2">Level of Service</th><th colspan="2">Customer IP addresses used by the service</th></tr>
<tr><th>Nominal</th><th>Backup</th></tr>
<tr><td colspan="4">ARCHITECTURE 1: NO REDUNDANCY</td></tr>
<tr><td>1 IPO Server (Call Server) or 1 IPO IP500V2 system</td><td>No redundancy
1 single call server or 1 IP500v2 system</td><td>IPO IP@</td><td>N/A</td></tr>
<tr><td colspan="4">ARCHITECTURE 2: REDUNDANCY - 2 IPO systems (active/active) - 1 NUMBERING PLAN</td></tr>
<tr><td>2 IPO systems (active/active), nominal/backup for a group of users (1 numbering plan).
The IPO systems can be hosted by the same site or by 2 different physical sites.
Each IPO system (IPO1 and IPO2) has its own SIP trunk but IPO2 is only used as a backup. Both IPO systems are independent but considered as being part of one HQ.

- Nominal mode: All users register with IPO1
- Backup mode: All users re-register with IPO2

Remark: 1 IPO system can be 1 IPO Server (Call Server) or 1 IPO IP500V2 system</td><td>User registration redundancy (IP phones only)
Rerouting at SBC level</td><td>IPO1 IP@</td><td>IPO2 IP@</td></tr>
<tr><td colspan="4">ARCHITECTURE 3: REDUNDANCY - 2 IPO systems (active/active) - 2 NUMBERING PLANS</td></tr>
<tr><td rowspan="2">2 IPO systems (active/active) hosted by 2 different physical sites.
Each IPO system manages a range of users (2 numbering plans).
Each IPO system (IPO1 and IPO2) has its own SIP trunk and each manages its own group of users in nominal mode.
- Nominal mode:
All HQ1 users register with IPO1 HQ1
All HQ2 users register with IPO2 HQ2

- Backup mode:
In case of IPO1 HQ1 crash, all HQ1 users re-register onto IPO2 HQ2
In case of IPO2 HQ2 crash, all HQ2 users re-register with IPO1 HQ1

Remark: 1 IPO system can be 1 IPO Server (Call Server) or 1 IPO IP500V2 system

Warnings: Both HQ accesses capacity to be sized adequately</td><td>For IPO1 HQ1
User registration redundancy (IP phones only)
Rerouting at AS level</td><td>IPO1 HQ1 IP@</td><td>N/A</td></tr>
<tr><td>For IPO2 HQ2
User registration redundancy (IP phones only)
Rerouting at AS level</td><td>IPO2 HQ2 IP@</td><td>N/A</td></tr>
</table>

Remote Site (RS) architecture Any Remote site architecture can be associated to any Head Quarter Architecture listed above	Level of Service	Customer IP addresses used by the service	
		Nominal	Backup
Remote site without Avaya media gateway (IP500v2) / ARCHITECTURES 1 or 2	No survivability, no trunk redundancy	N/A	N/A
Remote site without Avaya media gateway (IP500v2) / ARCHITECTURE 3		N/A	N/A
Remote site with Avaya media gateway (IP500v2) / ARCHITECTURES 1 or 2	Local site survivability and trunk redundancy via PSTN only	N/A	N/A
Remote site with Avaya media gateway (IP500v2) / ARCHITECTURE 3		N/A	N/A
Remote site with Avaya gateway (IP500v2) + SIP trunk as backup / ARCHITECTURES 1 or 2	Local survivability for the remote site hosting the gateway/SIP Trunk in case of non-access to HQ (HQ crash) Nominal outgoing and incoming traffic goes through HQ	GW IP@	N/A
Remote site with Avaya gateway (IP500v2) + SIP trunk as backup / ARCHITECTURE 3		GW IP@	N/A

3.2 Architecture with "Customer SBC" over BVPN

Architecture with Customer SBC over BVPN	Level of Service	Customer IP addresses used by the service	
		Nominal	Backup
ARCHITECTURE 4: Avaya Session Border Controller Enterprise (ASBCE)			
Single ASBCE	No redundancy	ASBCE IP@	NA
One ASBCE pair **in High Availability vendor mode** A pair consists in one SBCE server acting as primary (active) and another one server as secondary (standby). Both SBCE servers share the same IP@ (ASBCE VIP@).	Local vendor redundancy with nominal/backup behaviour. The 2 SBCE servers can be located on two different geographic sites but Layer 2 connection between servers 150 ms max round Trip is required. Loss of audio for all active calls on primary SBCE by only 1 second when it fails and its connection with the secondary ASBCE server is up. Loss of audio for all active calls on primary SBCE by 15 seconds when it fails and its connection with the secondary ASBCE server is down.	ASBCE VIP@	NA

Two ASBCE (ASBCE1 and ASBCE2) in Nominal/Backup mode on vendor side	Local vendor redundancy with nominal/backup behaviour. Both ASBCE are hosted on the same site. Nominal/Backup mode on Orange a-SBC side for incoming traffic to the customer ASBCE. Loss of active calls handled by the ASBCE that fails.	ASBCE1 IP@	ASBCE2 IP@
Two ASBCE pairs in High Availability and in Nominal/Backup mode on vendor side One ASBCE1 pair (2 ASBCE servers) with shared ASBCE1 VIP@ and one ASBCE2 pair (2 ASBCE servers) with shared ASBCE2 VIP@).	Local/geographical redundancy. The two ASBCE pairs are hosted on the same site or on 2 different geographic sites. Nominal/Backup mode on Orange a-SBC side for incoming traffic to the customer ASBCE pairs. If a full ASBCE pair fails, active calls are lost. Loss of audio for all active calls on primary SBCE of a pair by only 1 second when it fails and its connection with the secondary ASBCE server is up. Loss of audio for all active calls on primary SBCE of a pair by 15 seconds when it fails and its connection with the secondary ASBCE server is down.	ASBCE1 VIP@	ASBCE2 VIP@

Remote Site (RS) architecture Any Remote site architecture can be associated to any Customer SBC Architecture listed above	Level of Service	Customer IP addresses used by the service	
		Nominal	Backup
Remote site without Avaya media gateway (IP500v2) / ARCHITECTURE 4	No survivability, no trunk redundancy	N/A	N/A
Remote site with Avaya media gateway (IP500v2) / ARCHITECTURE 4	Local site survivability and trunk redundancy via PSTN only	N/A	N/A

3.3 Architecture with “Customer SBC” over Internet for BTIPoI

Architecture with Customer SBC over Internet	Level of Service	Customer IP addresses used by the service	
		Nominal	Backup
ARCHITECTURE 5: Avaya Session Border Controller Enterprise (ASBCE)			
Single ASBCE	No redundancy	ASBCE public FQDN DNS type A or type SRV	NA
One ASBCE pair in High Availability vendor mode A pair consists in one SBCE server acting as primary (active) and another one server as secondary (standby). Both SBCE servers share the same IP@ (ASBCE VIP@).	Local vendor redundancy with nominal/backup behaviour. The 2 SBCE servers can be located on two different geographic sites but Layer 2 connection between servers 150 ms max round Trip is required. Loss of audio for all active calls on primary SBCE by only 1 second when it fails and	ASBCE public FQDN DNS type A or type SRV	NA

	its connection with the secondary ASBCE server is up. Loss of audio for all active calls on primary SBCE by 15 seconds when it fails and its connection with the secondary ASBCE server is down.		
Two ASBCE (ASBCE1 and ASBCE2) in Nominal/Backup mode on vendor side	Local vendor redundancy with nominal/backup behaviour. Both ASBCE are hosted on the same site. Nominal/Backup mode on Orange a-SBC side for incoming traffic to the customer ASBCE. Loss of active calls handled by the ASBCE that fails.	ASBCE1 public FQDN DNS type A or type SRV	ASBCE2 public FQDN DNS type A or type SRV
Two ASBCE pairs in High Availability and in Nominal/Backup mode on vendor side One ASBCE1 pair (2 ASBCE servers) with shared ASBCE1 VIP@ and one ASBCE2 pair (2 ASBCE servers) with shared ASBCE2 VIP@).	Local/geographical redundancy. The two ASBCE pairs are hosted on the same site or on 2 different geographic sites. Nominal/Backup mode on Orange a-SBC side for incoming traffic to the customer ASBCE pairs. If a full ASBCE pair fails, active calls are lost. Loss of audio for all active calls on primary SBCE of a pair by only 1 second when it fails and its connection with the secondary ASBCE server is up. Loss of audio for all active calls on primary SBCE of a pair by 15 seconds when it fails and its connection with the secondary ASBCE server is down.	ASBCE1 public FQDN DNS type A or type SRV	ASBCE2 public FQDN DNS type A or type SRV

Remote Site (RS) architecture Any Remote site architecture can be associated to any Customer SBC Architecture listed above	Level of Service	Customer IP addresses used by the service	
		Nominal	Backup
Remote site without Avaya media gateway (IP500v2) / ARCHITECTURE 5	No survivability, no trunk redundancy	N/A	N/A
Remote site with Avaya media gateway (IP500v2) / ARCHITECTURE 5	Local site survivability and trunk redundancy via PSTN only	N/A	N/A

3.4 Architecture with “Customer SBC” over Internet for BToI

Architecture with Customer SBC over Internet	Level of Service	Customer IP addresses used by the service	
		Nominal	Backup
ARCHITECTURE 6: Avaya Session Border Controller Enterprise (ASBCE)			
Single ASBCE	No redundancy	ASBCE public IP@ or public FQDN DNS type A	NA
One ASBCE pair **in High Availability vendor mode** A pair consists in one SBCE server acting	Local vendor redundancy with nominal/backup behaviour. The 2 SBCE servers can be located on two	ASBCE public IP@ or public	NA

as primary (active) and another one server as secondary (standby). Both SBCE servers share the same IP@ (ASBCE VIP@).	different geographic sites but Layer 2 connection between servers 150 ms max round Trip is required. Loss of audio for all active calls on primary SBCE by only 1 second when it fails and its connection with the secondary ASBCE server is up. Loss of audio for all active calls on primary SBCE by 15 seconds when it fails and its connection with the secondary ASBCE server is down.	FQDN DNS type A	
Two ASBCE (ASBCE1 and ASBCE2) in Nominal/Backup mode on vendor side	Local vendor redundancy with nominal/backup behaviour. Both ASBCE are hosted on the same site. Nominal/Backup mode on Orange a-SBC side for incoming traffic to the customer ASBCE. Loss of active calls handled by the ASBCE that fails.	ASBCE1 public IP@ or public FQDN DNS type A	ASBCE2 public IP@ or public FQDN DNS type A
Two ASBCE pairs in High Availability and in Nominal/Backup mode on vendor side One ASBCE1 pair (2 ASBCE servers) with shared ASBCE1 VIP@ and one ASBCE2 pair (2 ASBCE servers) with shared ASBCE2 VIP@).	Local/geographical redundancy. The two ASBCE pairs are hosted on the same site or on 2 different geographic sites. Nominal/Backup mode on Orange a-SBC side for incoming traffic to the customer ASBCE pairs. If a full ASBCE pair fails, active calls are lost. Loss of audio for all active calls on primary SBCE of a pair by only 1 second when it fails and its connection with the secondary ASBCE server is up. Loss of audio for all active calls on primary SBCE of a pair by 15 seconds when it fails and its connection with the secondary ASBCE server is down.	ASBCE1 public IP@ or public FQDN DNS type A	ASBCE2 public IP@ or public FQDN DNS type A

Remote Site (RS) architecture Any Remote site architecture can be associated to any Customer SBC Architecture listed above	Level of Service	Customer IP addresses used by the service	
		Nominal	Backup
Remote site without Avaya media gateway (IP500v2) / ARCHITECTURE 6	No survivability, no trunk redundancy	N/A	N/A
Remote site with Avaya media gateway (IP500v2) / ARCHITECTURE 6	Local site survivability and trunk redundancy via PSTN only	N/A	N/A

4. BTIP/BTalk/BTIPoI/BToI certified versions

Orange supports the last 2 major IPBX versions only if still supported by Avaya and will ensure Business Talk and BTIP infrastructure evolutions will rightly interwork with the related architectures. Orange will assist customers running supported IPBX versions and facing issues.

Avaya standard support policy is to provide support for the most current major releases via standard software service pack processes.

For more details about the versions supported by Avaya, please refer to Lifecycle Summary Matrix and PCN and PSN Reports available on the Avaya Support Web site https://support.avaya.com.

<table>
<tr><th colspan="7">AVAYA IP OFFICE IPBX – software versions</th></tr>
<tr><th colspan="2">Reference products & versions</th><th colspan="4">✓: Certified
NS : No supported</th><th rowspan="3">Comments/restrictions</th></tr>
<tr><th rowspan="2">AVAYA IP Office
Select edition</th><th rowspan="2">With Avaya
Session Border Controller for Enterprise</th><th colspan="4">Orange Services</th></tr>
<tr><th>BTIP</th><th>BTIPoI</th><th>BTalk</th><th>BToI</th></tr>
<tr><td>Avaya 11.1 FP3
(11.1.3.0.0 build 23)</td><td>10.1.2.0-64-23285</td><td>✓</td><td>✓</td><td>✓</td><td>✓</td><td rowspan="2">To avoid any security risk the clients should always install on ASBCE but also on IP Office platforms the latest mandatory patch/hotfix released by the Avaya vendor.</td></tr>
<tr><td>Avaya 11.1 FP2 SP4
(11.1.2.4.0 build 18)</td><td>8.1.3.2-38-22279 + Hotfix-3 sbce-8.1.3.2-38-23109-hotfix-03082023.tar.gz</td><td>✓</td><td>✓</td><td>✓</td><td>✓</td></tr>
<tr><td>Avaya 11.1 FP2 SP2
(11.1.2.2.0 build 20)</td><td>From 8.1.3.1-38-21632 + Hotfix-3 sbce-8.1.3.1-39-22407-hotfix-08232022.tar.gz</td><td colspan="4" rowspan="5">Versions not supported</td><td rowspan="5">These IP Office versions support a maximum length of the tag value in From and To SIP headers limited to 80 characters. However, with the transition to Full-IP and when third-party operators are involved the length of tag value sent to IP Office can be superior to 80 characters causing IP Office to cancel the call. Upgrade to corrective version (11.1 FP2 SP4 or higher) is therefore required.</td></tr>
<tr><td>Avaya 11.1 FP2 SP1
(11.1.2.1.0 build 3)</td><td>From 8.1.3.1-38-21632 + Hotfix-3 sbce-8.1.3.1-39-22407-hotfix-08232022.tar.gz</td></tr>
<tr><td>Avaya 11.1 FP1
(11.1.1.0 build 209)</td><td>From 8.1.2.0-31-19809 + Hotfix-8 sbce-8.1.2.0-37-21486-hotfix-01062022.tar.gz</td></tr>
<tr><td>Avaya 11.0 FP4 SP2
(11.0.4.2.0 build 58)</td><td>NA</td></tr>
<tr><td>Avaya 11.0 FP4
(11.0.4.0 build 74)</td><td>NA</td></tr>
</table>

4.1 Avaya IP Office endpoints and applications

<table>
<tr><th colspan="6">AVAYA IP OFFICE IPBX - Endpoints and applications</th></tr>
<tr><th colspan="2">Reference product</th><th>Software version
NA: not applicable</th><th>Certification
✓: Certified
NS : No supported</th><th>IP Office version</th><th>Comments</th></tr>
<tr><td rowspan="4">Avaya IPBX components</td><td rowspan="2">IP Office Server Edition</td><td>11.1.3.0 build 23</td><td>✓</td><td>11.1 FP3</td><td></td></tr>
<tr><td>11.1.2.4.0 build 18</td><td>✓</td><td>11.1 FP2 SP4</td><td></td></tr>
<tr><td rowspan="2">IP Office UC module</td><td>11.1.3.0 build 23</td><td>✓</td><td>11.1 FP3</td><td></td></tr>
<tr><td>11.1.2.4.0 build 18</td><td>✓</td><td>11.1 FP2 SP4</td><td></td></tr>
<tr><td>Avaya Gateway</td><td>IP500v2</td><td>11.1.3.0 build 23</td><td>✓</td><td>11.1 FP3</td><td></td></tr>
</table>

		11.1.2.4.0 build 18	✓	11.1 FP2 SP4	
Avaya Voice Mail	VoiceMail Pro	11.1.3.0 build 7	✓	11.1 FP3	
		11.1.2.4.0 build 2	✓	11.1 FP2 SP4	
Avaya Unified Communications and Mobility	One-X Portal	11.1.3.0 build 26	✓	11.1 FP3	
		11.1.2.4.0 build 3	✓	11.1 FP2 SP4	
	One-X Mobile Preferred Edition for Android	Refer to the Product Compatibility Matrix tool available on https://secureservices.avaya.com/compatibility-matrix/menus/product.xhtml to find for each Avaya product, the software releases compatible with Avaya IP Office release.	NS	11.1 FP3, 11.1 FP2 SP4	
Third-party endpoints & Applications	ISI-COM Interact	7.x/8.x	✓	11.1 FP3, 11.1 FP2 SP4	
Avaya endpoints	B179 SIP conference phones	Refer to the Product Compatibility Matrix tool available on https://secureservices.avaya.com/compatibility-matrix/menus/product.xhtml to find for each Avaya product, the software releases compatible with Avaya IP Office release.	✓	11.1 FP3, 11.1 FP2 SP4	
	J129 SIP phones				
	J129/J139/J169/J179 SIP phones				
	1603L, 1608L, 1616L IP phones				
	1603, 1608, 1616 IP phones				
	9608, 9611G, 9621G, 9641G, 9641GS IP phones				
Avaya Attendant	IP Office Softconsole	Refer to the Product Compatibility Matrix tool available on https://secureservices.avaya.com/compatibility-matrix/menus/product.xhtml to find for each Avaya product, the software releases compatible with Avaya IP Office release.	✓	11.1 FP3, 11.1 FP2 SP4	
Avaya Softphone	Workplace client (for Windows, Android, iOS)	Refer to the Product Compatibility Matrix tool available on https://secureservices.avaya.com/compatibility-matrix/menus/product.xhtml to find for each Avaya product, the software releases compatible with Avaya IP Office release.	✓	11.1 FP3, 11.1 FP2 SP4	
	Avaya Communicator for Windows	Refer to the Product Compatibility Matrix tool available on https://secureservices.avaya.com/compatibility-matrix/menus/product.xhtml to find for each Avaya product, the software releases compatible with Avaya IP Office release.	NS	11.1 FP3, 11.1 FP2 SP4	
	Avaya Communicator for iPad	Refer to the Product Compatibility Matrix tool available on https://secureservices.avaya.com/compatibility-matrix/menus/product.xhtml to find for each Avaya product, the software releases compatible with Avaya IP Office release.	NS	11.1 FP3, 11.1 FP2 SP4	
Avaya DECT	Avaya 3730,3735 DECT phones	Refer to the Product Compatibility Matrix tool available on https://secureservices.avaya.com/compatibility-matrix/menus/product.xhtml to find for each Avaya product, the software releases compatible with Avaya IP Office release.	✓	11.1 FP3, 11.1 FP2 SP4	
	Avaya 3720,3725 DECT phones				
	Avaya 3749 DECT phones				
	Avaya 3740,3745 DECT phones				
	DECT R4 – IPBS3	Refer to the Product Compatibility Matrix tool available on https://secureservices.avaya.com/compatibility-matrix/menus/product.xhtml to find for each Avaya product, the software releases compatible with Avaya IP Office release.	✓	11.1 FP3, 11.1 FP2 SP4	
	DECT R4 – IPBS1-IPBS2				
	DECT R4 – AIWS2				
	DECT R4 – AIWS1				

5. IP Office SIP trunking configuration checklist

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

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 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

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

<table>
<tr><th colspan="6">Codec configuration</th></tr>
<tr><td rowspan="6">Every platform in the solution</td><td rowspan="6">System</td><td rowspan="6">-</td><td rowspan="2">Telephony -> Telephony</td><td>Companding Law</td><td>A-Law</td></tr>
<tr><td>High Quality Conferencing</td><td>Checked</td></tr>
<tr><td rowspan="4">VoIP</td><td>Ignore DTMF Mismatch For Phones</td><td>Checked</td></tr>
<tr><td>RFC2833 Default Payload</td><td>101</td></tr>
<tr><td rowspan="2">Default Codec Selection -> Selected</td><td>G.722 64K
G.711 ALAW 64K*
(*or G.711 ULAW 64K in option)</td></tr>
<tr></tr>
<tr><th colspan="6">Call Admission Control & Location configuration[2]</th></tr>
<tr><td rowspan="7">Solution level</td><td rowspan="7">Location</td><td rowspan="7">Location</td><td rowspan="7">Location</td><td>Location Name</td><td>Ex:RS140</td></tr>
<tr><td>Subnet Address</td><td>6.201.40.0</td></tr>
<tr><td>Subnet Mask</td><td>255.255.255.0</td></tr>
<tr><td>Parent Location for CAC</td><td><None></td></tr>
<tr><td>Call Admission Control -> Total Maximum Calls</td><td>99</td></tr>
<tr><td>Call Admission Control -> External Maximum Calls</td><td>99</td></tr>
<tr><td>Call Admission Control -> Internal Maximum Calls</td><td>99</td></tr>
<tr><td>Every platform in the solution</td><td>System</td><td>-</td><td>System</td><td>Location</td><td>Ex:HQ313</td></tr>
<tr><th colspan="6">Fallback configuration[3]</th></tr>
<tr><td>Primary IPO</td><td>Location</td><td>Location</td><td>Location</td><td>Fallback System</td><td>Local GW's IP address</td></tr>
<tr><th colspan="6">SCN lines configuration</th></tr>
<tr><td rowspan="5">Primary IPO[4]</td><td rowspan="5">Line</td><td rowspan="5">IP Office line [5]</td><td rowspan="5">Line</td><td>Outgoing Group ID</td><td>99998</td></tr>
<tr><td>Transport Type</td><td>Proprietary</td></tr>
<tr><td>Networking Level</td><td>SCN</td></tr>
<tr><td>Gateway -> Address</td><td>Backup IPO's IP address</td></tr>
<tr><td>Gateway -> Location</td><td>Location name</td></tr>
</table>

[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

				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
		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
				- Backs up my Hunt Groups	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

				- 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
		IP Office Line [12]	Line	Outgoing Group ID	99998
				Transport Type	Proprietary

[9] SCN line to expansion gateway

[10] Redundant architecture only

[11] SCN line to Primary server

[12] SCN line to secondary server

				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
				Prefix	0
				National Prefix	00
				Country Code	33
				International Prefix	000

[13] Line type depends on line type attached to Expansion Gateway

				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
			Call Details	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 -> FIeld meaning -> Forwarding/Outgoing calls	Caller
				Local URI -> FIeld meaning -> Forwarding/Twinning	Original Caller
				Local URI -> FIeld meaning -> Forwarding/Incoming calls	Called
				Contact-> Display	Use Internal Data
				Contact-> Content	Use Internal Data
				Contact -> FIeld meaning -> Outgoing calls	Caller
				Contact -> FIeld meaning -> Forwarding/Twinning	Original Caller
				Contact -> FIeld 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/Twinning	Caller
				Diversion Header -> Fleld meaning -> Incoming Calls	None
			VoIP	Codec Selection	Custom
				DTMF Support	RFC2833/RFC4733
				Local HOLD Music	Checked
				RE-invite 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

[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).

				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
				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

			Call Details	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/Outgoing calls	Caller
				Local URI -> Fleld meaning -> Forwarding/Twinning	Original Caller
				Local URI -> Fleld meaning -> Forwarding/Incoming calls	Called
				Contact-> Display	Use Internal Data
				Contact-> Content	Use Internal Data
				Contact -> Fleld meaning -> Outgoing calls	Caller
				Contact -> Fleld meaning -> Forwarding/Twinning	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/Twinning	Caller
				Diversion Header -> Fleld meaning -> Incoming Calls	None
			VoIP	Codec Selection	Custom
				Codec Selected	G.722 64K G.711 ALAW 64K* (*or G.711 ULAW 64K in option)
				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 Location limit	Allow Voicemail / Reject Call[16]
				Call Control -> Suppress Q.850 Reason Header	Checked
			Engineering	Custom String	SLIC_NO_USER_AVAIL=480

[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).

[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.

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
			Call Details	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 -> FIeld meaning -> Forwarding/Outgoing calls	Caller
				Local URI -> FIeld meaning -> Forwarding/Twinning	Original Caller
				Local URI -> FIeld meaning -> Forwarding/Incoming calls	Called

				Contact-> Display	Use Internal Data
				Contact-> Content	Use Internal Data
				Contact -> Fleld meaning -> Outgoing calls	Caller
				Contact -> Fleld meaning -> Forwarding/Twinning	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/Twinning	Caller
				Diversion Header -> Fleld meaning -> Incoming Calls	None
			VoIP	Codec Selection	Custom
				Codec Selected	G.722 64K G.711 ALAW 64K* (*or G.711 ULAW 64K in option)
				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

[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).

				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 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

[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.

				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
				Layer 4 Protocol	UDP
				Network Configuration -> Use Network Topology Info	None
				Send Port	5060
				Listen Port	5060
			Call Details	Incoming Group	111
				Outgoing Group	111
				Max Sessions	Default=10 Range 1 - 250
				Local URI -> Display	Use Internal Data
				Local URI -> Content	Use Internal Data
				Local URI -> FIeld meaning -> Forwarding/Outgoing calls	Caller
				Local URI -> FIeld meaning -> Forwarding/Twinning	Original Caller
				Local URI -> FIeld meaning -> Forwarding/Incoming calls	Called
				Contact-> Display	Use Internal Data
				Contact-> Content	Use Internal Data
				Contact -> FIeld meaning -> Outgoing calls	Caller
				Contact -> FIeld meaning -> Forwarding/Twinning	Original Caller
				Contact -> FIeld 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/Twinning	Caller
				Diversion Header -> Fleld meaning -> Incoming Calls	None
			VoIP	Codec Selection	Custom
				Codec Selected	G.722 64K G.711 ALAW 64K* (*or G.711 ULAW 64K in option)
				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
				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 ->	18

[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).

				Call Initiation Timeout (s)	
				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
				Codec Selection	Custom
				Codec Selected	G.722 64K G.711 ALAW 64K* (*or G.711 ULAW 64K in option)
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

[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

		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
	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	ARS	ARS1	ARS	Route Name	Main

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

[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.

(if used)			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
				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

[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.

<table>
<tr><td rowspan="13"></td><td></td><td></td><td></td><td>Line Group ID</td><td>11</td></tr>
<tr><td rowspan="4">User Rights</td><td rowspan="4">User Rights</td><td>User</td><td>Name</td><td>RS140</td></tr>
<tr><td rowspan="2">Short Codes</td><td>-</td><td>Apply User Rights value</td></tr>
<tr><td>Short Code table (Code, Telephone Number, Feature, Line Group ID)</td><td>Please refer to next section</td></tr>
<tr><td>User Rights Membership</td><td>Member of this User Rights</td><td>All RS140 users</td></tr>
<tr><td rowspan="8">Short Code</td><td rowspan="4">Short Code</td><td rowspan="4">-</td><td>Code</td><td>002XXXXXXXX[29]</td></tr>
<tr><td>Feature</td><td>Dial</td></tr>
<tr><td>Telephone Number</td><td>02N</td></tr>
<tr><td>Line Group ID</td><td>54: RS140</td></tr>
<tr><td rowspan="4">Short Code</td><td rowspan="4">-</td><td>Code</td><td>000N;</td></tr>
<tr><td>Feature</td><td>Dial</td></tr>
<tr><td>Telephone Number</td><td>00N</td></tr>
<tr><td>Line Group ID</td><td>54: RS140</td></tr>
<tr><td colspan="6">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.</td></tr>
<tr><td>Primary IPO</td><td>ARS</td><td colspan="4">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.</td></tr>
<tr><td>Secondary IPO (if used)</td><td>ARS</td><td colspan="4">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.</td></tr>
<tr><td rowspan="4">Expansion Gateway</td><td rowspan="4">Short Code</td><td rowspan="4">Short Code</td><td rowspan="4">-</td><td>Code</td><td>9N</td></tr>
<tr><td>Feature</td><td>Dial</td></tr>
<tr><td>Telephone Number</td><td>NS225374380[31]</td></tr>
<tr><td>Line Group ID</td><td>3</td></tr>
</table>

[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.

[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

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)					
Primary IPO	Line	SIP Line 10	Call Details	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/Outgoing calls	Caller
				Local URI -> Fleld meaning -> Forwarding/Twinning	Original Caller
				Local URI -> Fleld meaning -> Forwarding/Incoming calls	Called
				Contact-> Display	Auto
				Contact-> Content	Auto
				Contact -> Fleld meaning -> Outgoing calls	Caller

[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.

				Contact -> FIeld meaning -> Forwarding/Twinning	Original Caller
				Contact -> FIeld meaning -> Incoming calls	Called
		SIP Line 11	Call Details	Incoming Group	11
				Outgoing Group	11
				Max Sessions	Default=10 Range 1 - 250
				Local URI -> Display	Auto
				Local URI -> Content	Auto
				Local URI -> FIeld meaning -> Forwarding/Outgoing calls	Caller
				Local URI -> FIeld meaning -> Forwarding/Twinning	Original Caller
				Local URI -> FIeld meaning -> Forwarding/Incoming calls	Called
				Contact-> Display	Auto
				Contact-> Content	Auto
				Contact -> FIeld meaning -> Outgoing calls	Caller
				Contact -> FIeld meaning -> Forwarding/Twinning	Original Caller
				Contact -> FIeld meaning -> Incoming calls	Called
-	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

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

				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]
	Location	Location	Location	Emergency ARS	HQ313-Emergency
	Line	SIP Line 10	Call Details	Incoming Group	0
				Outgoing Group	20[39]
				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/Twinning	Original Caller
				Local URI -> Fleld meaning -> Forwarding/Incoming calls	Called
				Contact-> Display	Example: +33296083900
				Contact-> Content	Example: +33296083900
				Contact -> Fleld meaning -> Outgoing Call	Explicit

[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.

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

				Contact -> FIeld meaning -> Forwarding/Twinning	Original Caller
				Contact -> FIeld meaning -> Incoming calls	Called
		SIP Line 11	Call Details	Incoming Group	0
				Outgoing Group	21[40]
				Max session	Default=10 Range 1 - 250
				Local URI -> Display	Example: +33296083900
				Local URI -> Content	Example: +33296083900
				Contact -> FIeld meaning -> Outgoing Call	Explicit
				Local URI -> FIeld meaning -> Forwarding/Twinning	Original Caller
				Local URI -> FIeld meaning -> Forwarding/Incoming calls	Called
				Contact-> Display	Example: +33296083900
				Contact-> Content	Example: +33296083900
				Contact -> FIeld meaning -> Outgoing Call	Explicit
				Contact -> FIeld meaning -> Forwarding/Twinning	Original Caller
				Contact -> FIeld meaning -> Incoming calls	Called
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
			Add…	Code	N
				Feature	Dial
				Telephone Number	N

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

				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 110	Call Details	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/Twinning	Original Caller
				Local URI -> Fleld meaning -> Forwarding/Incoming calls	Called
				Contact-> Display	Example: +33296083900
				Contact-> Content	Example: +33296083900
				Contact -> Fleld meaning -> Outgoing Call	Explicit
				Contact -> Fleld meaning -> Forwarding/Twinning	Original Caller
				Contact -> Fleld meaning -> Incoming calls	Called
		SIP Line 111	Call Details	Incoming Group	0
				Outgoing Group	121[44]
				Max Session	Default=10 Range 1 - 250

[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!

[44] 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/Twinning	Original Caller
				Local URI -> Fleld meaning -> Forwarding/Incoming calls	Called
				Contact-> Display	Example: +33296083900
				Contact-> Content	Example: +33296083900
				Contact -> Fleld meaning -> Outgoing Call	Explicit
				Contact -> Fleld meaning -> Forwarding/Twinning	Original Caller
				Contact -> Fleld meaning -> Incoming calls	Called
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.		

[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 / 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					
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]

[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

6. IP Office + ASBCE SIP trunking configuration over BVPN checklist

The aim of this chapter is to provide steps to configure an Avaya Session Border Controller for Enterprise for interworking between the IP Office and BTIP/Business Talk service.
This guide shows only the settings to be checked or changed. The other settings can remain at their default values.

Device Management -> **Licensing**	
External WebLM Server URL	https://<SMGR_server_IP>:52233/WebLM/LicenseServer or https://<SMGR_server_domain_name>:52233/WebLM/LicenseServer e.g. **https://6.5.27.232:52233/WebLM/LicenseServer** or **https://smgr80.warsaw.lab:52233/WebLM/LicenseServer**
Device Management -> Devices -> **Add**	
Host Name	e.g. **asbceipo**
Management IP	e.g. **6.3.12.91**
Device Management -> Devices -> **Install**	
Device Configuration Appliance Name	This name will be referenced in other configuration e.g. **asbce**
DNS Configuration Primary	e.g. **6.3.14.10**
Network Configuration Name	Interface name toward IP Office e.g. **Int-ASBCE-IPO**
Network Configuration Default Gateway	e.g. **6.5.53.254**
Network Configuration Subnet Mask or Prefix Length	e.g. **255.255.255.0**
Network Configuration Interface	e.g. **A2** Note: Interface must be enabled on SBCE virtual machine on ESXi host after installation is complete.
IP Address 1#	IP address of the internal SBCE interface e.g. **6.5.52.62**

Network & Flows -> Network Management - > Networks -> Add	
Name	Interface name toward Orange SBC e.g. Ext-SBCE-BTIP
Default Gateway	e.g. 172.22.235.30
Network Prefix or Subnet Mask	e.g. 255.255.255.240
Interface	e.g. B1 Note: Interface must be enabled on SBCE virtual machine on ESXi host after configuration is complete.
IP Address	IP address of the external SBCE interface e.g. 172.22.235.19 Note: Reboot of the SBCE is required after configuration of the IP addresses.
Gateway Override	e.g. 172.22.235.30
Network & Flows -> Network Management - > Interfaces	
Interface name A2	Enabled Note: Previously configured interface must be enabled
Interface name B1	Enabled Note: Previously configured interface must be enabled
Network & Flows -> Signaling Interface -> Add	
Name	Create a signaling interface for the internal side of the SBCE e.g. Sign_Int_SBCE-IPO
IP Address	Select ASBCE internal interface and associated IP address defined in previous step. Int_ASBCE-IPO (A2, VLAN 0) 6.5.53.62
UDP port	This is the port on which SBCE will listen to SIP messages from IP Office. 5060 Note: UDP protocol is used for communication between ASBCE & IP Office.
Network & Flows -> Signaling Interface -> Add	
Name	Create a signaling interface for the external side of the SBCE e.g. Sign_Ext_SBCE-BTIP
IP Address	Select ASBCE external interface and associated IP address defined in previous step. Ext_SBCE-BTIP (B1, VLAN 0) 172.22.235.19

UDP port	This is the port on which SBCE will listen to SIP messages from Orange SBC. 5060 Note: UDP protocol is used for communication between ASBCE & Orange SBC.
Network & Flows -> Advanced Options -> Port Ranges	
Signaling Port Range	Default range: 12000-21000
Config Proxy Internal Signaling Port Range	Default range: 22000 – 31000
Listen Port Range	Default range: 9000 – 9999
HTTP Port Range	Default range: 40001 – 50000
Network & Flows -> Media Interface -> Add	
Name	Create a media interface for the internal side of the SBCE e.g. Media_Int_SBCE-IPO
IP Address	Select ASBCE internal interface and corresponding ip address configured in previous step. Int_ASBCE-IPO (A2, VLAN 0) 6.5.53.62
Port Range	Default range: 35000 – 40000
Network & Flows -> Media Interface -> Add	
Name	Create a media interface for the external side of the SBCE e.g. Media_Ext_SBCE-BTIP
IP Address	Selec ASBCE external interface and corresponding ip address configured in previous step. Ext_SBCE-BTIP (B1, VLAN 0) 172.22.235.19
Port Range	Default range: 35000 – 40000
Configuration Profiles -> Server Interworking -> Interworking Profiles -> Add	
Profile Name	e.g. SBCE-IPO
General tab Leave default parameters and ensure following parameters are selected:	
Hold Support	None

180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	Unchecked
3xx Handling	Unchecked
Delayed SDP Handling	Unchecked
Re-Invite Handling	Unchecked
Prack Handling	Unchecked
Allow 18X SDP	Unchecked
T.38 Support	For fax transmission over VISIT SIP trunk enable T.38 support for future usage. Checked
URI Scheme	SIP
Via Header Format	RFC3261
SIP Timers tab Leave default parameters (blank fields).	
Privacy Leave default parameters (blank fields).	
Interworking Profile Advanced parameters:	
Record Routes	Both Sides
Include End Point IP for Context Lookup	Unchecked
Extensions	Avaya
Diversion Manipulation	Unchecked
Has Remote SBC	Checked

Route Response on Via Port	Unchecked
Relay INVITE Replace for SIPREC	Unchecked
MOBX Re-INVITE Handling	Unchecked
DTMF	
DTMF Support	None Note: Avaya sip phones sends DMFs over RTP according to RFC4733.
Configuration Profiles -> Server Interworking -> Interworking Profiles -> Add	
Profile Name	e.g. SBCE-BTIP
General Leave default parameters and ensure following parameters are selected:	
Hold Support	None
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	Unchecked
3xx Handling	Unchecked
Delayed SDP Handling	Unchecked
Re-Invite Handling	Unchecked
Prack Handling	Unchecked
Allow 18X SDP	Unchecked
T.38 Support	For fax transmission over VISIT SIP trunk enable T.38 support for future usage. Checked
URI Scheme	SIP

Via Header Format	RFC3261
SIP Timers Leave default parameters (blank fields).	
Privacy Leave default parameters (blank fields).	
Interworking Profile Advanced parameters	
Record Routes	Both Sides
Include End Point IP for Context Lookup	Unchecked
Extensions	None
Diversion Manipulation	Unchecked
Has Remote SBC	Checked
Route Response on Via Port	Unchecked
Relay INVITE Replace for SIPREC	Unchecked
MOBX Re-INVITE Handling	Unchecked
DTMF	
DTMF Support	None Note: Avaya sip phones sends DMFs over RTP according to RFC4733.
Services -> SIP Servers -> Server profiles -> Add	
Profile Name	Define profile for far away server: Avaya IP Office. Prof_SBCE-IPO
General	
Server Type	Call Server
SIP Domain	Leave empty
DNS Query Type	NONE/A
TLS Client Profile	none
IP Address / FQDN	Add primary and backup IPO if exists. e.g. 6.3.85.1 e.g. 6.3.85.2

Port	This is the port on which IP Office will listen to SIP messages from Avaya SBCE. 5060
Transport	Protocol used for SIP signaling between IP Office and the Avaya SBCE. UDP
Authentication Leave all fields blank.	
Heartbeat Configure Heartbeat to send Options to monitor status of a trunk toward IPO server (Primary and if exists) defined in previous step.	
Enable Heartbeat	Checked
Method	OPTIONS
Frequency	90
From URI	e.g. ping@6.3.85.1
To URI	e.g. ping@warsaw.lab
Registration Leave all fields blank.	
Ping Leave all fields blank.	
Advanced Leave default fields except following:	
Enable DoS Protection	Unchecked
Enable Grooming	With Grooming enabled the system can reuse the same connections for the same subscriber or port. Checked
Interworking Profile	Select the Interworking Profile for IP Office defined previously. SBCE-IPO
Signaling Manipulation Script	None
Securable	Unchecked
Enable FGDN	Unchecked
Tolerant	Unchecked
URI Group	None

Services -> SIP Servers -> Server profiles -> Add	
Profile Name	Define profile for far away server: Orange SBC. Prof_SBCE-BTIP
Server Type	Trunk Server
SIP Domain	Leave empty
DNS Query Type	NONE/A
TLS Client Profile	none
IP Address / FQDN	Add all Orange SBC servers (primary and backup if exists). e.g. 172.22.246.33 e.g. 172.22.246.73
Port	This is the port on which Orange SBC will listen to SIP messages from Avaya SBCE. 5060
Transport	Protocol used for SIP signaling between Orange BTIP SIP trunk service (i.e. Orange SBC primary and backup) UDP
Authentication Leave all fields blank.	
Heartbeat Configure Heartbeat to send Options to monitor status of a trunk toward the Orange SBC (Primary and Backup if exists) defined in previous step.	
Enable Heartbeat	Checked
Method	OPTIONS
Frequency	90
From URI	e.g. ping@172.22.235.19
To URI	e.g. ping@orange.sbc
Registration Leave all fields blank.	
Ping Leave all fields blank.	
Advanced	Leave default fields except following:
Enable DoS Protection	Unchecked

Enable Grooming	Unchecked
Interworking Profile	Select the Interworking Profile for Orange BTIP SIP trunk service defined previously. SBCE-BTIP
Signaling Manipulation Script	None
Securable	Unchecked
Enable FGDN	Unchecked
Tolerant	Unchecked
URI Group	None
Domain Policies -> Application Rules -> default -> **Application Rule**	
Audio	Regulate the number of audio sessions that are allowed for each trunk server, or a call server. In – checked Out - checked
Domain Policies -> Media Rules -> default-low-med -> **Encryption**	
Audio Encryption	
Preferred Formats	RTP
Interworking	Checked
Domain Policies -> Media Rules -> default-low-med -> **Advanced**	
Leave all checkboxes - **Unchecked**	
Domain Policies -> Media Rules -> default-low-med -> QoS -> **Edit**	
Media QoS Marking	
Enabled	Checked
DSCP	Selected
DSCP Audio	EF
DSCP Video	EF

Domain Policies -> Signaling Rules -> **Add**	
Rule Name	e.g. SigR_SBCE-IPO
Inbound Leave default parameters (Allow)	
Outbound Leave default parameters (Allow)	
Content-Type Policy	
Enable Content-Type Checks	Checked
Action	Allow
Multipart Action	Allow
Domain Policies -> Signaling Rules -> SigR_SBCE-IPO -> **Signaling QoS**	
Enabled	Checked
DSCP	Selected
Value	EF
Domain Policies -> Signaling Rules -> SigR_SBCE-IPO -> **UCID**	
Enabled	Unchecked
Domain Policies -> Signaling Rules -> SigR_SBCE-IPO -> Requests -> **Add in Request Control**	
Proprietary Request	Unchecked
Method Name	Options
In Dialog Action	Allow
Out of Dialog Action	Select **Block with** and type in first field **200** then in next field **OK**
Domain Policies -> Signaling Rules -> **Add**	
Rule Name	e.g. SigR_SBCE-BTIP

Inbound Leave default parameters (**Allow**).	
Outbound Leave default parameters (**Allow**).	
Content-Type Policy	
Enable Content-Type Checks	Checked
Action	Allow
Multipart Action	Allow
Domain Policies -> Signaling Rules -> SigR_SBCE -BTIP -> **Signaling QoS**	
Enabled	Checked
DSCP	Selected
Value	EF
Domain Policies -> Signaling Rules -> SigR_SBCE-BTIP -> **UCID**	
Enabled	Unchecked
Domain Policies -> Signaling Rules -> SigR_SBCE-BTIP -> Requests -> **Add in Request Control**	
Proprietary Request	Unchecked
Method Name	Options
In Dialog Action	Allow
Out of Dialog Action	Select **Block with** and type in first field **200** then in next field **OK**
Domain Policies -> End Point Policy Groups -> **Add**	
Group Name	e.g. **EPPG_SBCE-IPO**
Domain Policies -> End Point Policy Groups -> EPPG_SBCE-IPO -> **Edit Policy Set**	
Application Rule	default

Border rule	default
Media Rule	default-low-med
Security Rule	default-low
Signaling Rule	Select created previously: SigR_SBCE-IPO
Charging Rule	None
RTCP Monitoring Report Generation	Off
Domain Policies -> End Point Policy Groups -> **Add**	
Group Name	e.g. EPPG_SBCE-BTIP
Domain Policies -> End Point Policy Groups -> EPPG_SBCE-BTIP -> **Edit Policy Set**	
Application Rule	default
Border rule	default
Media Rule	default-low-med
Security Rule	default-low
Signaling Rule	select created previously: SigR_SBCE-BTIP
Charging Rule	None
RTCP Monitoring Report Generation	Off
Configuration Profiles -> Routing -> **Add**	
Profile name	e.g. Routing-to-IPO
Configuration Profiles -> Routing -> **Routing-to-IPO**	
Uri Group	*

Load Balancing	Priority
Transport	None
LDAP Server Profile	None
Matched Attribute Priority	Unchecked
Next Hop Priority	Checked
Ignore Route Header	Unchecked
ENUM	Unchecked
Time of Day	default
NAPTR	Unchecked
LDAP Routing	Unchecked
LDAP Base DN (Search)	None
Alternate Routing	Unchecked
Next Hop In-Dialog	Unchecked
ENUM Suffix	Leave this field blank.
Priority / Weight	1
SIP Server Profile	Select previously created: Prof_SBCE-IPO
Next Hop Address	Select IP address of the IPO Primary e.g. 6.3.85.1: 5060 (UDP)
Priority / Weight	2
SIP Server Profile	Select previously created: Prof_SBCE-IPO
Next Hop Address	Select IP address of the IPO Backup if exists e.g. 6.3.85.2: 5060 (UDP)

Configuration Profiles -> Routing -> Add	
Profile	e.g. Routing-to-BTIP
Configuration Profiles -> Routing -> Routing-to-BTIP	
Uri Group	*
Load Balancing	Priority
Transport	None
LDAP Server Profile	None
Matched Attribute Priority	Unchecked
Next Hop Priority	Checked
Ignore Route Header	Unchecked
ENUM	Unchecked
Time of Day	default
NAPTR	Unchecked
LDAP Routing	Unchecked
LDAP Base DN (Search)	None
Alternate Routing	Unchecked
Next Hop In-Dialog	Unchecked
ENUM Suffix	Leave this field blank.
Priority / Weight	1
SIP Server Profile	Select previously created: Prof_SBCE-BTIP

Next Hop Address	Select IP address of the Orange SBC Primary e.g. 172.22.246.33: 5060 (UDP)
Priority / Weight	2
SIP Server Profile	Select previously created: Prof_SBCE-BTIP
Next Hop Address	Select IP address of the Orange SBC Backup if exists e.g. 172.22.246.73: 5060 (UDP)
Configuration Profiles -> Topology Hiding -> Add	
Profile Name	This profile will be applied for the traffic from the Avaya SBCE to IP Office. e.g. THP_SBCE-IPO
Configuration Profiles -> Topology Hiding -> Topology Hiding Profiles -> THP_SBCE-IPO -> Add Header	
Header	Add all following headers: Via Request-Line SDP Record-Route Refer-To To From Referred-By For all headers set the following parameters:
Criteria	IP/Domain
Replace Action	Auto
Configuration Profiles -> Topology Hiding -> Add	
Profile Name	This profile will be applied for the traffic from the Avaya SBCE to Orange Business. e.g. THP_SBCE-BTIP
Configuration Profiles -> Topology Hiding -> Topology Hiding Profile -> THP_SBCE-BTIP -> Add Header	
Header	Add all following headers: Via Request-Line SDP Record-Route Refer-To To From Referred-By For all headers set the following parameters:

Criteria	IP/Domain
Replace Action	Auto
Network & Flows -> End Point Flows -> Server Flows -> Add	
Flow Name	Traffic from Orange SBC through Avaya SBCE toward IP Office: e.g. EPF_SBCE-IPO
SIP Server Profile	Select previously configured profile: Prof_SBCE-IPO
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Select the external signaling interface Sign_Ext_SBCE-BTIP
Signaling Interface	Select the internal signaling interface Sign_Int_SBCE-IPO
Media Interface	Select the internal media interface Media_Int_SBCE-IPO
Secondary Media Interface	None
End Point Policy Group	Select the endpoint policy group defined previously EPPG_SBCE-IPO
Routing Profile	Select the routing profile to direct traffic to BTIP SIP trunk Routing-to-BTIP
Topology Hiding Profile	Select the topology hiding profile defined for IP Office THP_SBCE-IPO
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	Unchecked
Network & Flows -> End Point Flows -> Server Flows -> Add	
Flow Name	Traffic from IP Office through Avaya SBCE toward Orange SBC: e.g. EPF_SBCE-BTIP

SIP Server Profile	Select previously configured profile: Prof_SBCE-BTIP
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Select the internal signaling interface Sign_Int_SBCE-IPO
Signaling Interface	Select the external signaling interface Sign_Ext_SBCE-BTIP
Media Interface	Select the external media interface Media_Ext_SBCE-BTIP
Secondary Media Interface	None
End Point Policy Group	Select the endpoint policy group defined previously EPPG_SBCE-BTIP
Routing Profile	Select the routing profile to direct traffic to IP Office Routing-to-IPO
Topology Hiding Profile	Select the topology hiding profile defined for BTIP SIP trunk THP_SBCE-BTIP
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	Unchecked

7. IP Office + ASBCE SIP trunking configuration over Internet checklist

Below table focuses on BToI/BTIPoI SIP trunk configuration on ASBCE indicating the required update of configuration in addition to already implemented BT/BTIP configuration described in previous chapter.

5

6

7

TLS Management -> Certificates > **Create CSR**	
Country Name	e.g. FR
State/Province Name	e.g. Bretagne
Locality Name	e.g. Rennes
Organization Name	e.g. Orange
Organizational Unit	e.g. Orange Business
Common Name	FQDN assigned to ASBCE public ip address. CN domain name must be resolved on public DNS. Allowed characters in the CN are alphanumeric and hypen [-]. Special characters must not be used. e.g. external.domain.com
Algorithm	SHA256
Key Size (Modulus Length)	2048 bits
Key Usage Extension(s)	Checked Key encipherment Checked Non-Repudiation Checked Digital Signature
Extended Key Usage	Checked Server Authentication Checked Client Authentication
Subject Alt Name	FQDN for SAN is the same as for CN. e.g. DNS: external.domain.com
Passphrase Confirm Passphrase	Allowed characters are alphanumeric and special character but Avaya recommends not to use the dollar sign ($) in Key Passphrase Specify the passphrase to encrypt the private key.
Contact Name	e.g. Slawomir

Contact E-Mail	Email address
TLS Management -> Certificates -> Install	
Type	Select Certificate
Name	This field is optional. Can be left blank.
Overwrite Existing	Unchecked
Allow Weak Certificate/Key	Unchecked
Certificate File	Upload the Identity certificate file
Trust Chain File	Upload Trust Chain file. If the third party CA provided separate Root CA and Intermediate certificates for ASBCE, you must combine both files into a single certificate file (trust chain file). To combine the files, add the contents of each certificate file one after the other, with the root certificate at the end. (e.g. IntermediateAndRootCAchain.crt)
Key	Ensure that the Common Name used during generation of CSR matches with the file name of the identity certificate being installed. Select Use Existing Key
Key File	Select from a drop down list existing key file.
TLS Management -> Certificates -> Install	
Type	Select CA Certificate
Name	This field is optional. Can be left blank.
Overwrite Existing	Unchecked
Allow Weak Certificate/Key	Checked
Certificate File	Upload the public CA root & intermediate certificates file (trust chain file) of the remote entity (Orange A-SBC). e.g. OrangeIntermediateAndRootCAchain.pem
TLS Management -> Server Profile -> Add	
Profile Name	e.g. ThirdPartyServer

Certificate	Select installed **ASBCE Identity certificate.**
SNI Options	None
Peer Verification	**Required**
Peer Certificate Authorities	Select **public CA root & intermediate certificates** file (trust chain file) of the remote entity (**Orange A-SBC**). **e.g. OrangeIntermediateAndRootCAchain.pem**
Verification Depth	Depends of the number of bundled certificates. In case the third party CA provided separate Root CA and Intermediate certificates for the Orange A-SBC that were bundled into one file the value will be set to number **2**.
Renegotiation Time	0
Renegotiation Byte Count	0
Version	For encrypted BTIP/BTalk SIP Trunk architecture we need to configure TLS v1.2. Check **TLS 1.2**
Ciphers	Select: **Default** The cipher suite recommended by Avaya.
TLS Management -> Client Profile -> **Add**	
Profile Name	e.g. ThirdPartyClient
Certificate	Select installed **ASBCE Identity certificate.**
SNI Options	Unchecked Enabled
Peer Certificate Authorities	Select **public CA root & intermediate certificates** file (trust chain file) of the remote entity (**Orange A-SBC**). **e.g. OrangeIntermediateAndRootCAchain.pem**
Verification Depth	Depends of the number of bundled certificates. In case the third party CA provided separate Root CA and Intermediate certificates for the Orange A-SBC that were bundled into one file the value will be set to number **2**.
Extended Hostname Verification	Unchecked
Renegotiation Time	0
Renegotiation Byte Count	0

Version	For encrypted BTIP/BTalk SIP Trunk architecture we need to configure TLS v1.2. Check **TLS 1.2**
Ciphers	Select: **Default**
Network & Flows -> Network Management -> Networks → Ext-SBCE-BTIP -> **Edit**	
Name	Interface name toward Orange A-SBC e.g. **Ext-SBCE-BTIP**
Default Gateway	e.g. **195.205.163.25**
Network Prefix or Subnet Mask	Network prefix or subnet mask e.g.255.255.255.248
Interface	**B1**
IP Address	Public Ip address of the external ASBCE interface (e.g. 195.205.163.30)
Public IP	**Leave blank**
Gateway Override	**Leave blank**
Network & Flows -> Signaling Interface -> Sign_Ext_SBCE_BTIP -> **Edit**	
Name	Signaling interface of the external side of the ASBCE. e.g. **Sign_Ext_SBCE-BTIP**
Ip Address	ASBCE external interface and associated public ip address defined in previous step. **Ext_SBCE-BTIP (B1, VLAN 0)** **Public IP address e.g. 195.205.163.30**
TLS port	This is the port on which ASBCE will listen to SIP messages from Orange A-SBC. **5061** Remark: **TLS** protocol is used for communication between ASBCE & Orange A-SBC.
TLS Profile	Select: **ThirdPartyServer**
Services -> SIP Servers -> Prof_SBCE-BTIP-> **Edit**	
Profile Name	Edit/add profile for the far end server: Orange A-SBC. **Prof_SBCE-BTIP**
Server Type	**Trunk Server**

SIP Domain	**Leave blank**
DNS Query Type	DNS type Service Record (SRV) allows to query DNS server to receive hostname, priority, port of the target servers. Alternatively you can configure ip address or DNS Query Type A. **SRV** **NONE/A** BTIPoI supports type SRV & type A for DNS resolution and do not support direct public IP connections. BToI supports both public IP and type A for DNS resolution and do not provide any type SRV record connections.
TLS Client Profile	Select **ThirdPartyClient**
FQDN IP Address / FQDN	**FQDN** of the Orange A-SBC if DNS Query Type SRV was configured e.g. **BTIPOI.iptel.one.equant.net.** **IP Address** or **FQDN** of the Orange A-SBC if DNS Query Type None/A was configured.
Port	This is the port on which Orange A-SBC will listen to SIP messages from Avaya SBCE. This value will be received from DNS server in SRV response. If DNS query type A was configured then insert port 5061. **Leave blank** if DNS Query Type SRV was configured. **5061** if DNS Query Type None/A was configured.
Transport	Protocol used for SIP signaling between ASBCE and Orange A-SBC. It will also result in the ASBCE will add by default SRV type query prefix "_sips._tcp." while querying DNS if DNS Query Type SRV was configured. **TLS**
Configuration Profiles -> Routing -> Routing-to-BTIP-> **Edit**	
Uri Group	*
Load Balancing	**DNS/SRV** if DNS Query Type SRV was configured in previous step. **Priority** if DNS Query Type None/A was configured in previous step.
Transport	**None**
Next Hop In-Dialog	**Unchecked**
Time of Day	**default**
Next Hop Priority	**Unchecked** if Load Balancing DNS/SRV was configured. **Checked** if Load Balancing Priority was configured.
Ignore Route Header	**Unchecked**

ENUM	Unchecked
NAPTR	Unchecked
ENUM Suffix	Leave this field blank.
Priority / Weight	N/A if Load Balancing DNS/SRV was configured. 1 if Load Balancing DNS/A was configured.
SIP Server Profile	Select previously created: Prof_SBCE-BTIP
Next Hop Address	Select FQDN of the Orange A-SBC if Load Balancing DNS/SRV was configured. e.g. FQDN (TLS) Select IP address or FQDN of the Orange SBC Primary if Load Balancing DNS/A was configured. e.g. 172.22.246.33: 5061 (TLS) or FQDN: 5061 (TLS)
Priority / Weight	2 if Load Balancing Priority was configured.
SIP Server Profile	Select previously created: Prof_SBCE-BTIP
Next Hop Address	Select IP address or FQDN of the Orange SBC Backup if exists. e.g. 172.22.246.33: 5061 (TLS) or FQDN: 5061 (TLS)
Domain Policies -> Media Rules -> Add	
Rule Name	Orange-med-enc
Audio Encryption & Video Encryption	
Preferred Format #1	AES_CM_128_HMAC_SHA1_80
Preferred Format #2	NONE
Preferred Format #3	NONE
Encrypted RTCP	Checked
MKI	Unchecked
Lifetime Leave blank to match any value	Leave blank
Interworking	Checked

Miscellaneous	
Capability Negotiation	Unchecked
Audio Codec & Video Codec	
Codec Prioritization	Unchecked
Transcode	Unchecked
Allow Preferred Codecs Only	Unchecked
Transrating	Unchecked
P-Time	20
Silencing	
Silencing Enabled	Unchecked
Binary Flow Control Protocol	
BFCP Enabled	Unchecked
Far End Camera Control	
FECC Enabled	Unchecked
ANAT	
ANAT Enabled	Unchecked
Local Preference	IP4
Use Remote Preference	Unchecked
Media Line Compliance	
Media Line Compliance Enabled	Unchecked
Media QoS Marking	
Enabled	Checked
DSCP	selected
DSCP Audio	EF
DSCP Video	EF

Domain Policies -> End Point Policy Groups -> EPPG_SBCE-BTIP -> Edit Policy Set	
Application Rule	default
Border rule	default
Media Rule	select created previously: Orange-med-enc
Security Rule	default-low
Signaling Rule	SigR_SBCE-BTIP
Network & Flows -> Advanced Options -> Port Ranges	
Signaling Port Range	Depending on customer context or need. ASBCE TLS/TCP/UDP source ports for the SIP signaling. Allocate e.g. range: 51001-55000
Config Proxy Internal Signaling Port Range	50001-51000
Listen Port Range	55001-55999
HTTP Port Range	40001-50000
Network & Flows -> Media Interface -> Media_Int_SBCE-IPO -> Edit	
Name	Edit/Add a media interface for the internal side of the ASBCE e.g. Media_Int_SBCE-IPO
IP Address	ASBCE internal interface and corresponding ip address: Int_SBCE-IPO (A2, VLAN 0) 6.5.53.62
Port Range	The Orange BTIPoI/BToI SIP Trunk service specifies media ports that customers use on the internal SIP trunk. ASBCE UDP ports for the RTP media: 6000-38000 for BTIPoI 6000-20000 for BToI
Network & Flows -> Media Interface -> Media_Ext_SBCE_BTIP -> Edit	
Name	Edit/Add media interface for the external side of the ASBCE e.g. Media_Ext_SBCE-BTIP
IP Address	ASBCE external interface and corresponding ip address: Ext_SBCE-BTIP (B1, VLAN 0) Public IP Address e.g.195.205.163.30

Port Range	The Orange BTIPoI/BToI SIP Trunk service specifies media ports that customers use on the external SIP trunk. ASBCE UDP ports for the SRTP media: **6000-38000** for BTIPoI **6000-20000** for BToI
Domain Policies -> Application Rules -> **default**	
Maximum Concurrent Session	Change the value to **2000**
Maximum Sessions Per Endpoint	Change the value to **2000**
Configuration Profiles -> Server Interworking -> SBCE-IPO -> **Edit**	
Profile Name	SBCE-IPO
General	
SIPS Required	No
Configuration Profiles -> Server Interworking -> SBCE-BTIP -> **Edit**	
Profile Name	SBCE-BTIP
General	
SIPS Required	No
Domain Policies -> Session Policies -> default -> **Media**	
Media Anchoring	**Checked** for media anchoring
Media Forking Profile	None
Converged Conferencing	Unchecked
Recording Server	Unchecked
Media Server	Unchecked
Network & Flows -> **Session Flows**	
Media must be anchored on ASBCE. Session Flows must be default. Remove any session flow if exists.	

8. Ecosystems and endpoints configuration

8

8.1 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

8.2 Avaya 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 (Or if G711 Ulaw is in option: ▪ G722: 4 – High ▪ G711 Alaw: 0 – Disabled ▪ G711 Ulaw: 3 ▪ G729: 0 – Disabled)
SIP settings		
Primary Account	Enable account	YES
	Account name	Extn3133102
	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

8.3 Avaya 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]
		Confirm password	password
		Subscriptions	With User AC

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

	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

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

	Trunks	Status Inquiry period	30[62]
		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)

8.4 Avaya 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

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

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

		HOST Domain Name -> Primary HOST Domain Name	Primary FQDN
		HOST Domain Name -> Secondary HOST Domain Name	Secondary FQDN
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

8.5 Avaya 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

64 Password used to login.