Business Talk & BTIP for Unify OpenScape Business IPBX

Versions addressed in this guide: V2_R4, V2_R5, V2_R6 and V2_R7.

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

Document Version

Version of 12/12/2019

Table of Contents

Goal of this document

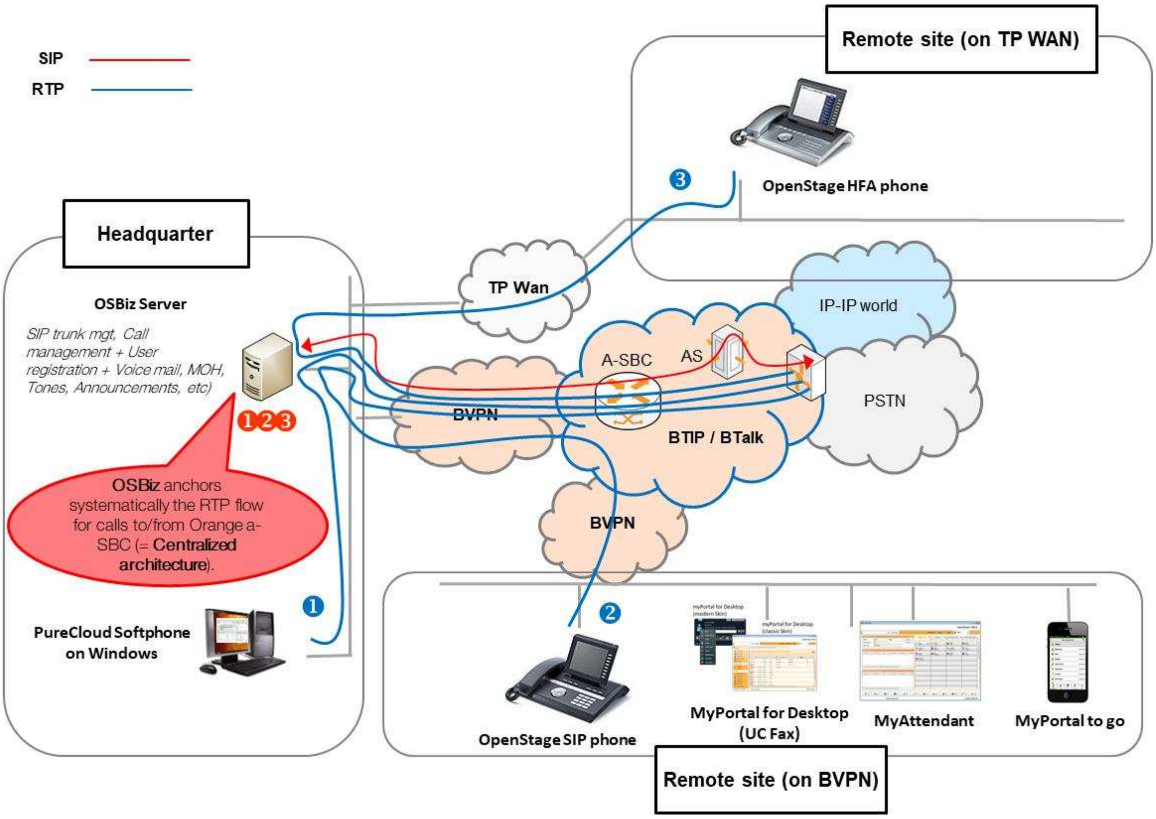
The aim of this document is to list technical requirements to ensure the interoperability between Unify OpenScape Business IPBX with Orange Business Services Business Talk / Business Talk IP SIP services, hereafter so-called “service”.

1 ARCHITECTURE OVERVIEW

Access to BT/BTIP is performed through 2 Orange a-SBC (nominal and backup).

Customer shall pay attention to get proper IPBX licencing.

2.1 Centralized architecture – without Customer SBC



In this centralized architecture

- All 'SIP trunking' signalling flows are carried by the OSBiz server and routed on the main BVPN connection.
- OSBiz server anchors systematically the RTP flow for calls to/from Orange a-SBC. Therefore, the RTP flow is not direct between SIP phones/softphones and Orange a-SBC. But IP routing differs from one site to another:
  - For the Head Quarter site, media flows are just routed on the main BVPN connection.
  - For Remote sites on BVPN or on Third Party WAN, media flows transit through the OSBIZ server and use the central BVPN connection (= centralized architecture, cf sizing below).

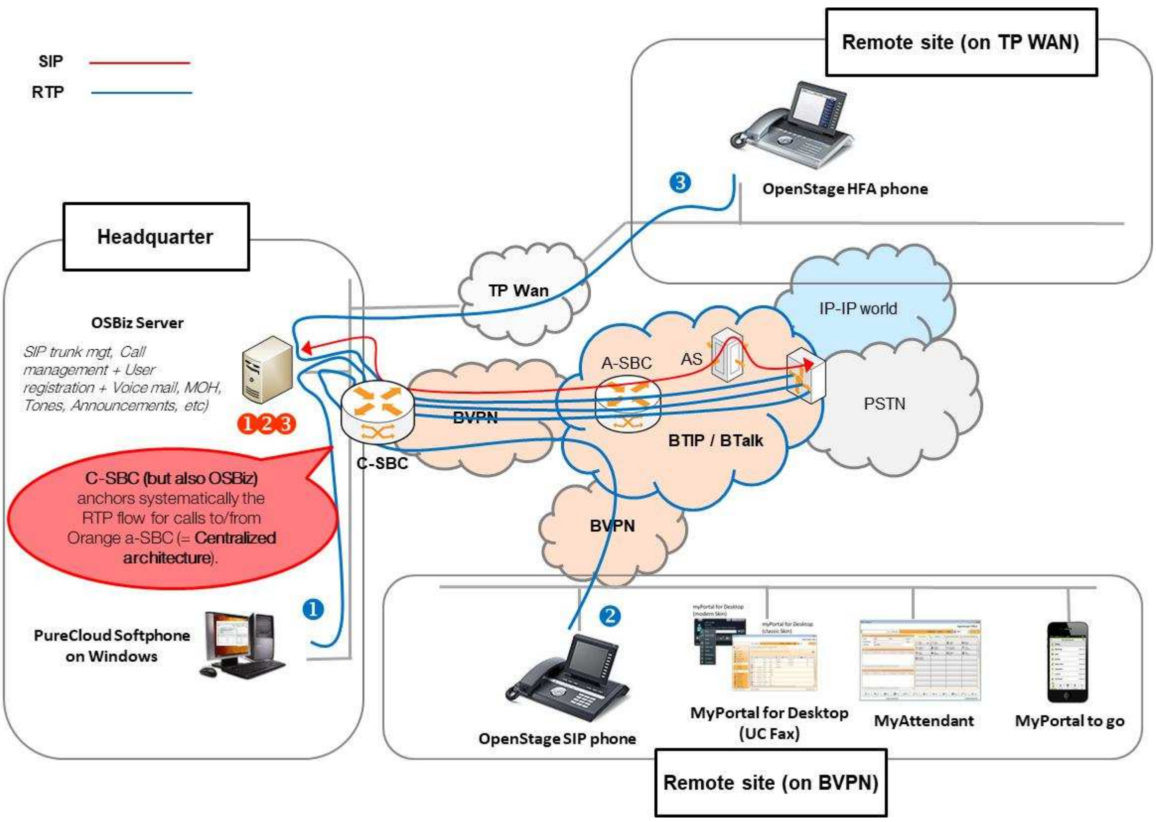
Notes: In the diagram above, the SIP and proprietary internal flows are hidden.

Here below a table with a few sizing elements

- Call scenario	Nb of voice channels/media resources used		
	OSBIZ	**WAN router***	**BTIP/BT**
1 offnet call from/to the headquarter (HQ)	**1**	**1** in HQ	**1** in HQ
1 offnet call from/to a remote site (RS) on BVPN	**2**	**2** in HQ **1** in RS	0 in HQ **1** in RS
1 offnet call from/to a remote site (RS) on TP Wan	**2**	**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 which **put the caller on hold**	**2**	**3** in HQ (ou 2) **1** in RS	0 in HQ **1** in RS
1 offnet call from/to a remote site **after transfer/forward to BTIP**	**2**	**2** 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**	**3** in HQ **1** in RS	**0** in HQ **0** in RS

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

2.2 Centralized architecture – with Customer SBC



Notes: in the diagram above, the SIP, proprietary internal flows are hidden:

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

In this centralized architecture:

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

Warning: With “customer SBC” architecture, site access capacity has to be sized adequately on the Head Quarter. Here below a table with a few sizing elements:

Call scenario	nb of voice channels/media resources used		
	OSBIZ	**WAN router***	**BTIP**
1 offnet call from/to the head quarter (HQ)	**1**	**1** in HQ	**1** in HQ
1 offnet call from/to a remote site (RS) on BVPN	**2**	**2** in HQ **1** in RS	0 in HQ **1** in RS
1 offnet call from/to a remote site (RS) on TP Wan	**2**	**1** in HQ BVPN **1** in HQ TPWan **1** in RS TPWan	0 in HQ **1** in RS
1 offnet call from/to a remote site **with put on hold**	**2**	**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**	**2**	0 in HQ*/3 in HQ** **0** in RS	0 in HQ **2** in RS
1 **forced onnet** call **from head quarter to a remote site** (= through Business Talk infrastructure)	**2**	**3** in HQ **1** in RS	**0** in HQ **0** in RS

*on the WAN router, 1 voice channel = 80Kb/s **if media release is activated on the cSBC ***if media release is not activated on the cSBC.

2.3 Survivability consideration

OpenScape Business proposes the two following survivability capacities:

- High Availability Server Hardware (redundant RAID disks, redundant power supply, …)

- VMWare High Availability

2.4 Codec consideration

Only G729 codec usage is not supported in the scope of Unify OpenScape Business SIP trunking interoperability with Orange Business Services. Only G711a codec is supported.

2.5 CAC consideration

OSBiz doesn’t support CAC. Nevertheless, OSBiz has a means of bandwidth control using number of maximum calls on ITSP connection only.

2.6 Sizing consideration

Specific sizing approach has to be considered with OpenScape Business solution due to the fact that:

- OSBiz anchors systematically the RTP flow for calls to/from Orange a-SBC (= centralized architecture). Therefore, the RTP flow is not direct between Unify phones and Orange a-SBC.
- But RTP flow is direct between Unify devices, for calls between customer sites (RS1, RS2, HQ).

3 PARAMETERS to be PROVIDED by CUSTOMER to ACCESS BT/BTIP service

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

Head Quarter (HQ) Centralized architecture	**Level of Service**	**Customer IP addresses used by service**	
		Nominal	**Backup**
Single OSBiz Server or OSBiz X3/X5/X8 system	No Call Server redundancy	OSBIZ Server/system IP@	N/A
Remote Site (RS) Centralized architecture	**Level of Service**	**Nominal**	**Backup**
Remote site with OpenScape X3/X5/X8 gateway	Local user survivability and trunk redundancy via PSTN only	N/A	N/A
Remote site without media gateway	No survivability, no trunk redundancy	N/A	N/A

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

4 CERTIFIED SOFTWARE and HARDWARE versions

4.1 Unify OpenScape Business IPBX

Unify OpenScape Business IPBX – software versions

Reference product	Software version	Certification ✓: Certified NS : No supported	Restrictions/Comments
	V2_R7.x	✓	
OpenScape Business software	V2_R6.x	✓	User-Agent header contains: OpenScape Business M5T SIP Stack/4.2.20.35
	V2_R5.x	✓	
	V2_R4.1.0	✓	

4.2 Unify OpenScape Business endpoints and applications

Unify OpenScape Business IBX - Endpoints and applications

Reference product		Software version NA: not applicable	Certification ✓: Certified NS : No supported	OpenScape Business version	Comments
SIP endpoints	OpenStage SIP 60, 80 phones	See Comments section	✓	V2_Rx	The software of these phones is shipped with the OpenScape Business software. Its exact version is indicated in the OpenScape Business release note.
Proprietary endpoints	OpenStage HFA 60, 80 phones	See Comments section	✓	V2_Rx	The software of these phones is shipped with the OpenScape Business software. Its exact version is indicated in the OpenScape Business release note.
TDM endpoints	OpenStage 20T	V2R1.x	✓	V2_Rx	
Analog endpoints	Euroset 5020	NA	✓	V2_Rx	
Attendant endpoints	MyAttendant	See Comments section	✓	V2_Rx	This application is shipped with the OpenScape Business software. Its exact version is indicated in the OpenScape Business release note.

Unified Communications and Mobility	MyPortal for Desktop	See Comments section	✓	V2_Rx	This application is shipped with the OpenScape Business software. Its exact version is indicated in the OpenScape Business release note.
	MyPortal to Go	See Store	✓	V2_Rx	

5 SIP TRUNKING CONFIGURATION CHECKLIST

Configuration guide required for interoperability between BTIP/BT and Unify OpenScape Business IPBX is available in Unify Wiki site: http://wiki.unify.com/wiki/Collaboration_with_VoIP_Providers#France.

6 MAIN SIP features and FUNCTIONAL restrictions

- Codecs:
  - G711 A 20 ms only (G729 A not certified)

- SIP Transport:
  - UDP

- FAX transmission:
  - T.38 supported

- Functional restrictions:
  - Only G729 codec is not supported by OSBiz
  - The use of IP phones, DECT phones, TDM phones & softphones into mobility conditions within the enterprise is not supported
  - User-To-User SIP header is not supported by OSBiz
  - OSBiz doesn't support CAC. Nevertheless, OSBiz has a means of bandwidth control using number of maximum calls on ITSP connection only.
  - In the case of a call forwarding (Unconditional or Busy) the initial caller will not hear the early media phase as long as the destination does not pick up the call. This concerns only destination which does not send 183SDP.
  - When an OSBiz station is unplugged (power / network) or unregistered, and receives an incoming call from PSTN, through Orange ITSP, OSBiz replies with a SIP 486 Busy. In consequence DTO option cannot trigger.
  - In call forwarding no answer scenarios, where an OSBiz user forwards an OSBiz user, who belongs to another (Networking) OSBiz node to PSTN, the caller has wrong phone display when the call is established; the caller displays the name of the original (OSBiz) called party and the number of the connected (PSTN) subscriber.
  - In call forwarding scenarios, where an OSBiz user forwards a call with a PSTN subscriber, the forwarded PSTN subscriber will display the original (dialed) OSBiz user and not the connected (forwarded to) party. On the other hand, the forwarded to party will have the correct display (i.e. calling party).
  - In an OSBiz conference with internal and external (PSTN) subscribers, when the internal subscribers leave the conference, depending on the configuration of the system the PSTN subscribers may continue to talk. While the OSBiz system didn't have configuration to drop the connection, the latter was dropped after 5 mins (configurable) and there was no tone to notify the PSTN subscribers that the connection was about to drop
  - In call transfer scenarios, where an OSBiz user transfers a call with a PSTN subscriber or transfers a call to a PSTN subscriber, the PSTN subscribers don't display the connected party, but continue to display the OSBiz user.
  - In blind transfer scenarios from an OSBiz user, during the transfer the transferred party hears MOH instead of ringback tone.