

Business Talk & BTIP for Unify OpenScape Business IPBX

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

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

Document Version

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Goal of this document

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

1 ARCHITECTURE OVERVIEW

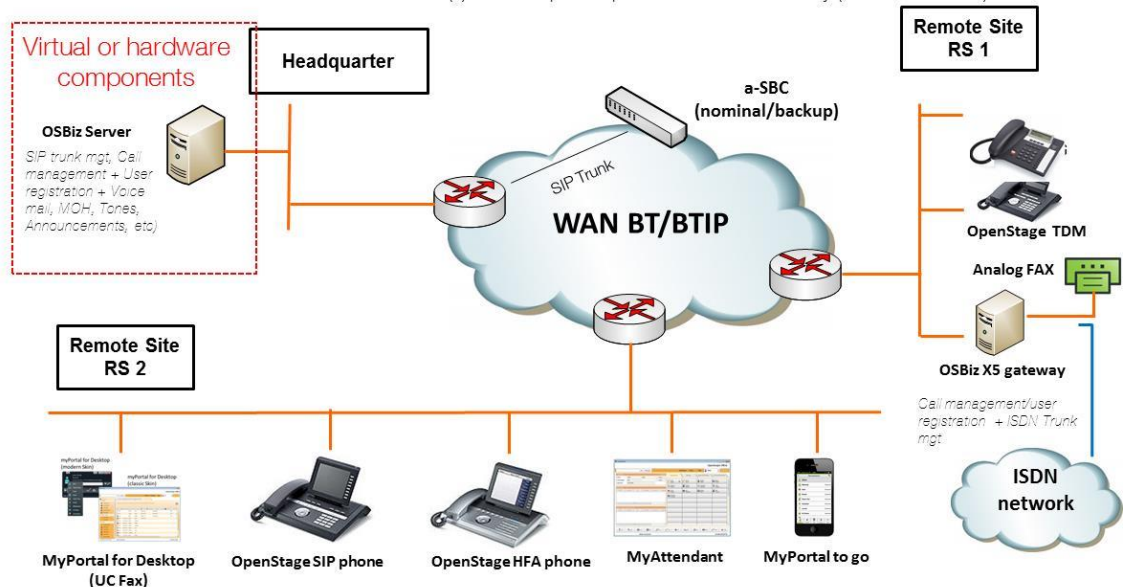
Access to BT/BTIP is performed through 2 Orange a-SBC (nominal and backup).

Customer shall pay attention to get proper IPBX licencing.

2.1 Distributed architecture (virtual + hardware) components

> Distributed Architecture

- 1 Headquarter based on the OpenScope Business solution (with SIP trunk)
- 1 or several Remote Site(s) with an OpenScope Business as a Gateway (without SIP trunk)
- 1 or several Remote Site(s) without OpenScope Business as a Gateway (without SIP trunk)



Survivability consideration

OpenScope Business proposes the two following survivability capacities:

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

Codec consideration

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

CAC consideration

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

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. 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) 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) 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

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

- SIP Transport:
 - o UDP

- FAX transmission:
 - o T.38 supported

- Functional restrictions:
 - o Only G729 codec is not supported by OSBiz
 - o The use of IP phones, DECT phones, TDM phones & softphones into mobility conditions within the enterprise is not supported
 - o User-To-User SIP header is not supported by OSBiz
 - o OSBiz doesn't support CAC. Nevertheless, OSBiz has a means of bandwidth control using number of maximum calls on ITSP connection only.
 - o 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.
 - o 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.
 - o 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.
 - o 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).
 - o 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
 - o 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.
 - o In blind transfer scenarios from an OSBiz user, during the transfer the transferred party hears MOH instead of ringback tone.