Architecture guide

Business Talk & Business Talk IP Genesys PureEngage on Premises

Versions addressed in this guide : 8.1.x

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

Latest edition : 09/04/2021

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Goal of this document

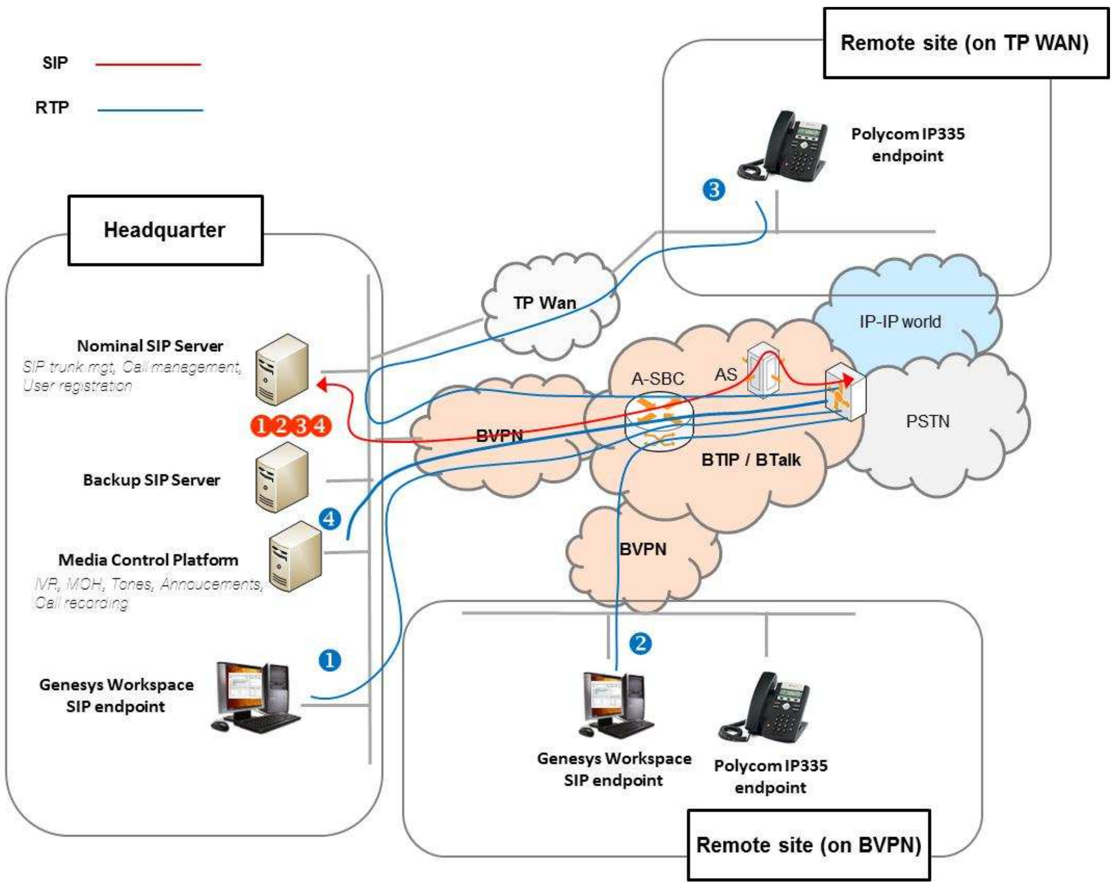
The aim of this document is to list technical requirements to ensure the interoperability between Genesys PureEngage solution on Customer Premises with OBS Business Talk & Business Talk IP SIP services, here after so-called “service”.

1 Architecture overview

Genesys PureEngage architecture involves a lot of various components such as SIP Server, Media Control Platform, Reporting Server, Resource Manager, Universal Routing Server …

In the below architecture only the components that interact with Orange SIP trunking infrastructure are laid out. So mainly the SIP server, the Resource Manager and the Media Control Platform components.

1.1 Architecture



Notes:

- in the diagram above, the SIP and proprietary internal flows are hidden.
- call flows will be the similar with or without SIP Server redundancy.
- Dedicated Media Control Platform, managed by the HQ Resource Manager can be deployed on remote sites to reduce inter-sites bandwidth.

In this architecture

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

Call scenario	**nb of voice channels/media resources used**		
	IPBX	**WAN router***	**BTIP**
1 offnet call from/to the headquarter (HQ)	**1** in HQ	**1** in HQ	**1** in HQ
1 offnet call from/to a remote site (RS) on BVPN	0 in HQ **1** in RS	0 in HQ **1** in RS	0 in HQ **1** in RS
1 offnet call from/to a remote site (RS) on TP Wan	0 in HQ **1** in RS	**1** in HQ BVPN **1** in HQ TP Wan **1** in RS TP Wan	0 in HQ **1** in RS
1 offnet call from/to a remote site **with put on hold** (on HQ)	**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

Redundancy consideration

Two SIP servers can be deployed in a primary/backup configuration in order to provide local redundancy. This requires that the 2 SIP servers are located on the same geographical site and in the same subnet. Both servers share a Virtual IP address. Consequently Orange a-SBC are configured to communicate with this single Virtual IP address.

Sizing consideration

There is no specific sizing approach to be considered with PureEngage solution. The RTP flow is direct between Genesys Media Control Platform or softphones/phones and Orange a-SBC.

QoS consideration

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 to set the IPBX, IP phones and other voice applications with a DiffServ/TOS value* = 46 (or PHB value = EF) at least for media.

2 Parameters to be provided by customer to access BTalk/BTIP service

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

Redundancy architecture - HQ Site architecture	Level of Service	Customer IP addresses used by the service	
		Nominal	Backup
Two SIP servers deployed in a primary/backup configuration in order to provide local redundancy. This requires that the 2 SIP servers are located on the same geographical site.	- Local redundancy	VIRTUAL IP@	NA

Remote Site (RS) architecture	Level of Service	Customer IP addresses used by the service	
		Nominal	Backup
Remote site with SIP phones/softphones	No local survivability, no SIP trunk redundancy.	NA	NA

3 Certified software and hardware versions

3.1 Genesys PureEngage

GENESYS PUREENGAGE – software versions				
Reference product	Software version	Certification ✓: Certified NS : No supported	Certified "Loads"	Restrictions
Genesys SIP Server	8.1	✓	8.1.104.06 8.1.103.37	
Genesys Media Control platform	8.5	✓	9.0.019.68 8.5.150.84	
Genesys Resource Manager	8.5	✓	8.5.178.80	

3.2 Endpoints and applications

Endpoints and applications					
Reference product		Software version NA: not applicable	Certification ✓: Certified NS : No supported	Genesys SIP Server version	Comments
Genesys endpoints	Genesys Workspace SIP Endpoint	8.1 & 8.5	✓	8.1	
Genesys desktop application	Interaction Workspace	8.1	✓	8.1	
	Workspace Desktop Edition	8.5	✓	8.1	
Audiocodes phones	4xxHD phones	-	Not tested	-	PureEngage supports a number of types of phones. Refer to the information provided by Genesys to find details about the phones that PureEngage supports.
Polycom phones	SoundPoint IP335	4.0.7.2514	✓	8.1	
	SoundPoint IPphones	-	Not tested	-	
	VVX IP phones	-	Not tested	-	
Yealink phones	SIP-TXX	-	Not tested	-	

4 SIP Trunking configuration checklist

The checklist below presents all the steps of configuration required for interoperability between BTIP/BT and Genesys PureEngage solution.
Please leave other options not specifically mentioned in this section as default and do not change their values.

4.1 SIP Trunk

Steps	Parameter Name	Value (example)
	SIP trunk to Orange SBC : creation and configuration	
Step 1	➢ Switches-> « Switch Name »-> DN -> New ➢ General	

Parameter name	Parameter value
1 Number	1 *Trunk Name* (ex : *SBC_Prim)*
2 Type	2 Trunk
3 Tenant	3 *Tenant name* (ex : *Resources)*
4 Association	4
5 Register	5 True
6 State	6 Enabled

➢ Options

Parameter name	Parameter value
1 TServer/contact	1 *Orange SBC IP address and port (ex : sip:192.168.1.1:5060)*
2 TServer/refer-enabled	2 false
3 TServer/make-call-rfc3725-flow	3 1
4 TServer/sip-enable-100rel	4 false
5 TServer/sip-early-dialog-mode	5 0
6 TServer/sip-busy-type	6 2
7 TServer/capacity-limit-inbound	7 true
8 TServer/capacity	8 *Total value (ex :100)*
9 TServer/oos-check	9 5
10 TServer/oos-force	10 10 seconds
11 TServer/oos-options-max-forwards	11 0

<table>
<tr><th></th><th>SIP server application configuration</th></tr>
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➢ Applications-> « SIP Server Name »

➢ Options

	Parameter name		Parameter value
1	TServer/audio-codecs	1	*Codec list (ex PCMA,telephone-event)*
2	TServer/event-ringing-on-100trying	2	false
3	TServer/divert-on-ringing	3	false
4	TServer/sip-address	4	*SIP Server IP address (ex :172.100.1.1)*
5	TServer/sip-port	5	*5060*
6	TServer/after-routing-timeout	6	*Value (ex :15)*
7	TServer/sip-enable-sdp-codec-filter	7	true
8	TServer/blind-transfer-enabled	8	false
9	TServer/refer-enabled	9	false
10	TServer/ringing-on-route-point	10	false
11	TServer/sip-enable-100rel	11	false
12	TServer/sip-treatments-continuous	12	true

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4.2 Outbound

SIP message INVITE for outbound calls need to contain SDP and a connection address different from 0.0.0.0

Outbound calls make by the Tmakecall request or the Troute function must use the extensions for “dummy SPD” Genesys functionality in:

sdp-c-host : *SIP Server address (ex : 172.100.1.1)*

sdp-m-port-low : *Value (ex : 8000)*

sdp-m-port-high : *Value (ex : 16000)*

4.3 Call Admission Control

The CAC is managed by the capacity and capacity-limit-inbound options in the trunk configuration.