

Business Talk & Business Talk IP For Genesys PureCloud Edge on Premises

Version addressed in this guide : 1.0.07616

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

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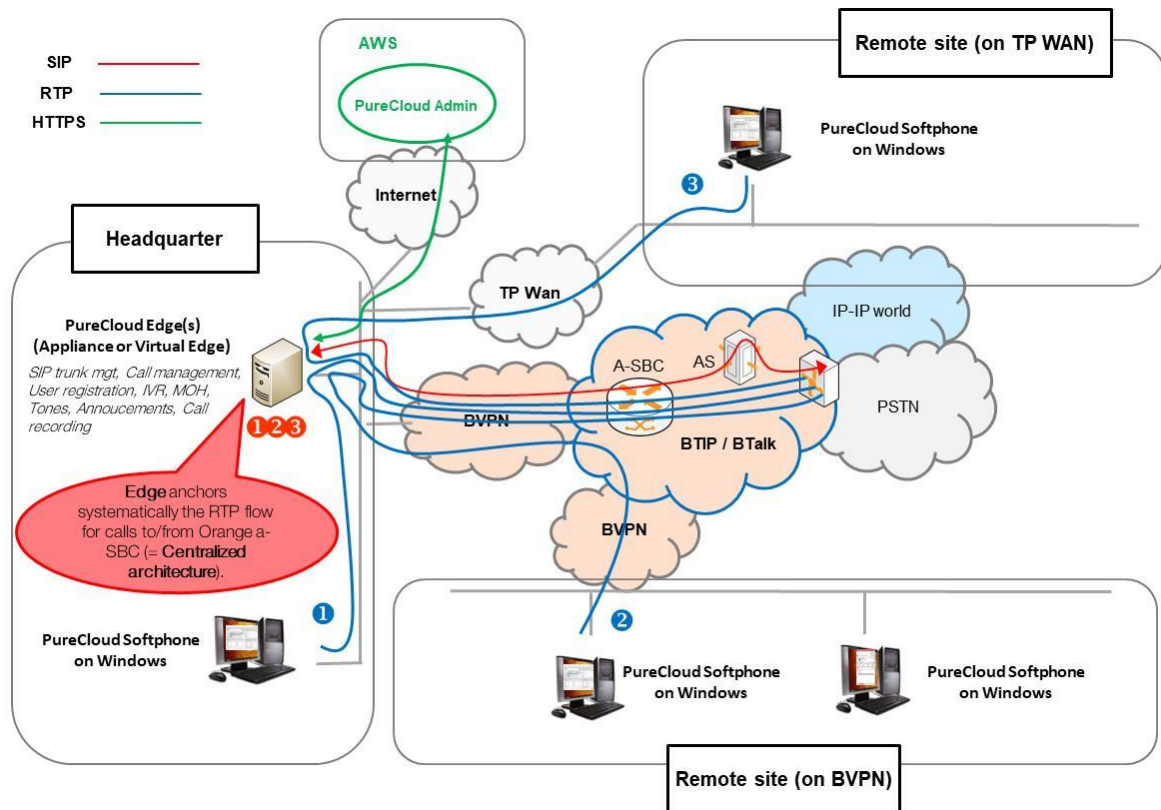
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Goal of this document

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

1 ARCHITECTURE OVERVIEW

1.1 Centralized Architecture – without Customer-SBC



In this centralized architecture

- All 'SIP trunking' signalling flows are carried by the PureCloud Edge and routed on the main BVPN connection.
- PureCloud Edge 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:
 - o For the Head Quarter site, media flows are just routed on the main BVPN connection.
 - o For Remote sites on BVPN or on Third Party WAN, media flows transit through the PureCloud Edge and use the central BVPN connection (= **centralized architecture**, cf sizing below).

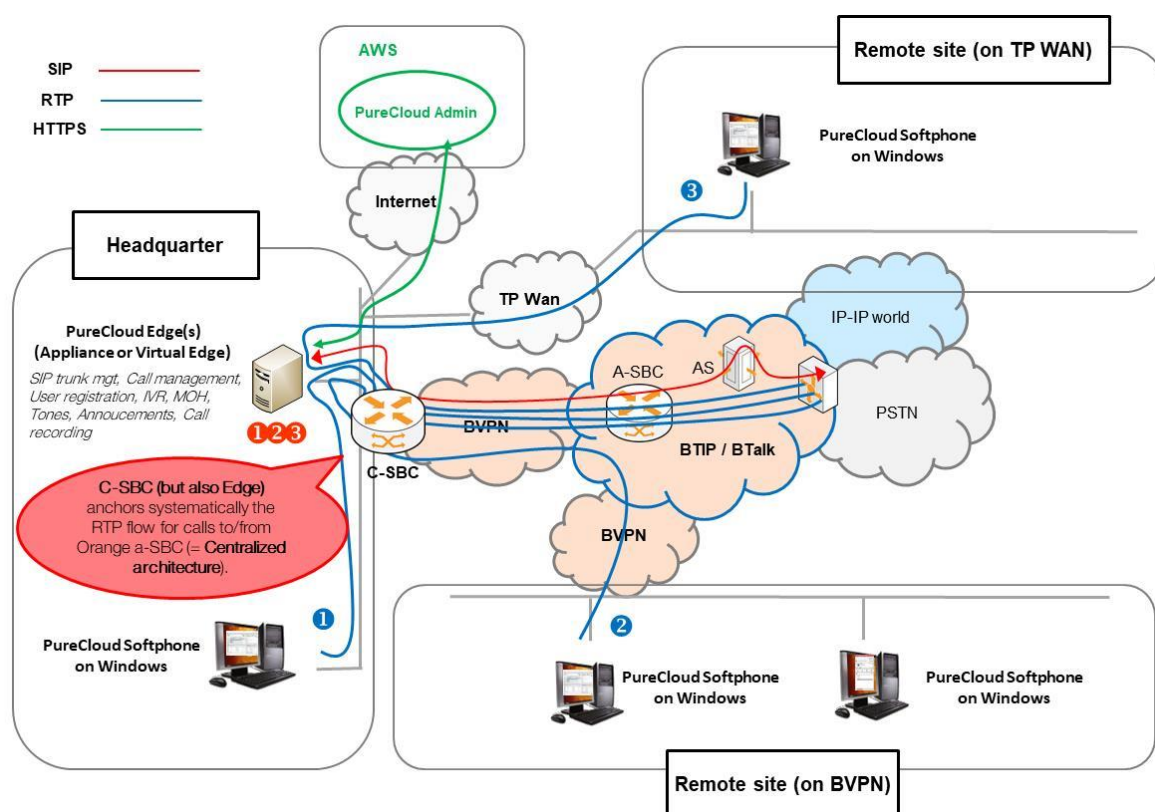
Notes: In the diagram above, the SIP and proprietary internal flows are hidden.

Here below a table with a few sizing elements

- Call scenario	Nb of voice channels/media resources used		
	EDGE	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

1.2 Centralized Architecture – with Customer-SBC



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

- ① call from/to Head Quarter
- ② call from/to remote site (on Business VPN)
- ③ 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		
	EDGE	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.

1.3 Redundancy consideration

Edge appliances can be deployed in an N+1 configuration in order to provide local redundancy.

This requires that:

- All the Edges are on the same geographical site or distributed in multi sites with very low latency between each site.
- All the Edges belong to the same PureCloud Edge Group.

The Orange a-SBC distributes the calls in round-robin on all the Edge IP addresses of the BTIP/BT Sites.

1.4 Sizing consideration

Specific sizing approach has to be considered with PureCloud Edge due to the fact that the Edge anchors systematically the RTP flow for calls to/from Orange a-SBC (= **Centralized architecture**). Therefore, the RTP flow is not direct between SIP phones/Softphones and Orange a-SBC.

2 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

N+1 Redundancy architecture - HQ Centralized architecture (Without Customer SBC)	Level of Service	Customer IP addresses used by the service	
		Nominal	Backup
N+1 redundancy: All edges are member of the same Edge group. The incoming load sharing is done using a-SBC round robin to all Edges. If an Edge passes out of Service, the secondary Edge becomes the new Primary for each softphone which registers on it, even when the previous Edge comes back In Service. If existing, a 3rd Edge becomes Secondary. Softphones or SIP phones are hosted on a physical site with or without Edge. A site from BT/BTIP point of view corresponds to a physical site with or without phones/softphones.	- Local Edge appliance redundancy - User registration redundancy: User is registered on 2 edges. One edge is seen as Primary for the user, the other one is seen as Secondary.	EDGE 1 IP@ EDGE 2 IP@ EDGE 3 IP@	NA

Remote Site (RS) Centralized architecture (Without Customer SBC)	Level of Service	Customer IP addresses used by the service	
		Nominal	Backup
Remote site	No survivability, no trunk redundancy.	NA	NA

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	

3 Business Talk & BTIP certified versions

Please refer to '<https://help.mypurecloud.com/edge-media-tier-release-notes/>' web page for more details about the versions supported by Genesys.

3.1 Genesys PureCloud Edge

GENESYS PURECLOUD EDGE – software versions				
Reference product	Software version	Certification ✓ : Certified NS : No supported	Certified "Loads"	Restrictions
Genesys PureCloud Edge	1.0.0	✓	1.0.07616	

3.2 Endpoints and applications

Endpoints and applications					
Reference product		Software version NA: not applicable	Certification ✓ : Certified NS : No supported	Genesys PureCloud Edge version	Comments
Genesys endpoints	PureCloud Softphone	2016 R2 micro programme : 16.2.0.1888	✓	1.0.07616	
Audiocodes phones	Audiocodes 400HD	-	Not tested	-	PureCloud supports a number of types of phones. Refer to the information provided by Genesys to find details about the phones that PureCloud supports.
Polycom phones	Polycom SoundPoint IP	-	Not tested	-	
	Polycom RealPresence Trio	-	Not tested	-	
	Polycom VVX	-	Not tested	-	
	Polycom SoundStation IP	-	Not tested	-	
Spectralink phones	Spectralink 8000 Portfolio Handsets	-	Not tested	-	

4 SIP TRUNKING CONFIGURATION CHECKLIST

The checklist below presents all the steps of configuration required for interoperability between BTIP/BT and Genesys PureCloud Edge.

Two external trunks must be created, one to each Orange a-SBC. Each “external trunk” instance applies to all edges.

Each edge uses these 2 trunks in Primary/Backup mode (see below). Each trunk is used for inbound and outbound calls.

One “Phone trunk” must be created for SIP Phone (Purecloud softphones), it applies to all edges.

4.1 External Trunk

Steps	Parameter Name	Value (example)																		
Creation et Configuration for an external trunk to a-SBC																				
Step 1	➤ Telephony→Trunks→External Trunks→ « Trunk Name »																			
	➤ General																			
	<table><thead><tr><th></th><th>Parameter Name</th><th>Value</th></tr></thead><tbody><tr><td>1</td><td>External Trunk Name</td><td>1 <i>Trunk Name (ex : SBC_113)</i></td></tr><tr><td>2</td><td>Type</td><td>2 External SIP</td></tr><tr><td>3</td><td>Trunk State</td><td>3 In-Service</td></tr><tr><td>4</td><td>Protocol</td><td>4 UDP</td></tr><tr><td>5</td><td>Listen Port</td><td>5 5060</td></tr></tbody></table>		Parameter Name	Value	1	External Trunk Name	1 <i>Trunk Name (ex : SBC_113)</i>	2	Type	2 External SIP	3	Trunk State	3 In-Service	4	Protocol	4 UDP	5	Listen Port	5 5060	
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➤ Sip Access Control																				
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➤ External Trunk Configuration/General

Parameter Name	Value
1 Call Draining	1 Enabled
2 Language	2 English
3 Max Calls	3 350
4 Max Call Rate	4 40/5s
5 Max Dial Timeout	5 120 sec
6 Asserted Identity	6 Disabled

➤ External Trunk Configuration/Transport

Parameter Name	Value
1 Transport DSCP Value	1 18 (24,011000)CS3
2 Retryable Reason Codes	2 500-599
3 Retryable Cause Codes	3 1-5,25,27,28,31,34,38,41,42,44,46,62,63,79,91,96,97,99,100,103
4 UDP T1 timeout	4 500ms
5 UDP T2 timeout	5 1s
6 UDP Max Packet retry	6 4
7 UDP Max Invite Retry	7 3

➤ External Trunk Configuration/Identity

Parameter Name	Value
Inbound	
1 Identity Type	1 From
Outbound	
2 Apply header privacy	2 Disabled or Enabled
3 Apply User privacy	3 Disabled
4 Calling/Address Transformation	4
5 Calling/Address Digits length	5 0
6 Calling/Address Omit+Prefix	6 Disabled
7 Called/Address Transformation	7
8 Called/Address Digits length	8 0
9 Called/Address Omit+Prefix	9 Disabled

➤ External Trunk Configuration/Media

Parameter Name	Value
1 DSCP Value	1 2E (46,101110) EF
2 Media Method	2 Normal
3 Preferred Codec List	3 Codec (Audio/PCMA, or Audio/G729)
4 SRTP Cipher Suite List	4 AES_CM_128_HMAC_SHA1_80
5 Ringback	5 Enabled
6 Disconnect on Idle RTP	6 Enabled
DTMF Setting	
7 DTMF Payload	7 101
8 DTMF Method	8 RTP Event
Recording	
9 Line Recording	9 Enabled
10 Dual Channel	10 Enabled
11 Continue on External Transfert	11 Enabled
12 Audio Format	12 audio/ PCMA
13 Automatic Level Control	13 Disabled
14 Recording/ Consent Required	14 Disabled

➤ External Trunk Configuration/Protocol	
Parameter Name	Value
Header/Invite	
1 Conversation Headers	1 Enabled
2 From Header Hostname	2 Automatically generate from Edge network Interface
3 Routing Address	3 Request-URI
4 Diversion Method	4 None
5 Asserted Identity Header	5 P-Asserted-Identity
6 Max Diversion Entries	6 4
7 Request Target Address	7
User to User Information (UUI)	
8 UUI Passthrough	8 Enabled
9 Header/Type	9 User-to-User
10 Encoding Format	10 Hex
11 Protocol Discriminator	11 04
12 Static User Data	12 Disable or Enabled
13 Header/name	13 User-to-User
14 Value	14 <i>Ex :</i> <i>0468656c6c6f20776f726c64;encoding=hex;purpose=isdn-interwork;content=isdn-uui</i>
15 Priority	15 Low
Take Back and Transfer	
16 Enable Take Back and transfer	16 Disabled
➤ External Trunk Configuration/Diagnostic	
Parameter Name	Value
1 Media Capture	17 Disabled
2 protocol Capture	18 Disabled
➤ External Trunk Configuration/Custom	
Parameter Name	Value

4.2 Phone Trunk

Steps	Parameter Name	Value (example)																	
	Creation and Configuration for Phone trunk																		
Step 1	➤ Telephony→Trunks→Phone Trunks→ « Trunk Name »																		
	➤ General																		
	<table><tr><th>Parameter Name</th><th>Value</th></tr><tr><td>1 Phone Trunk Name</td><td>1 <i>Phone Trunk Name</i></td></tr><tr><td>2 Type</td><td>(to_softphone)</td></tr><tr><td>3 Trunk State</td><td>2 SIP</td></tr><tr><td>4 Protocol</td><td>3 In-Service (check)</td></tr><tr><td>5 Listen Port</td><td>4 UDP</td></tr><tr><td></td><td>5 8060</td></tr></table>	Parameter Name	Value	1 Phone Trunk Name	1 <i>Phone Trunk Name</i>	2 Type	(to_softphone)	3 Trunk State	2 SIP	4 Protocol	3 In-Service (check)	5 Listen Port	4 UDP		5 8060				
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<table> <tr> <th>Parameter Name</th><th>Value</th></tr> <tr> <td>1 Require Authentications</td><td>1 Disabled</td></tr> <tr> <td>2 Prefer http Redirects</td><td>2 Disabled</td></tr> <tr> <td>3 Require HTTPS</td><td>3 Disabled</td></tr> <tr> <td>4 External Provision Server</td><td>4</td></tr> <tr> <td>5 HTTP Port</td><td>5 8098</td></tr> <tr> <td>6 HTTPS Port</td><td>6 8099</td></tr> </table>	Parameter Name	Value	1 Require Authentications	1 Disabled	2 Prefer http Redirects	2 Disabled	3 Require HTTPS	3 Disabled	4 External Provision Server	4	5 HTTP Port	5 8098	6 HTTPS Port	6 8099					
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4.3 Edges

Steps	Parameter Name	Value (example)	
Site Configuration			
Step 1	➤ Telephony→Edges→« Edge Name »		
	➤ General		
	Parameter Name	Value	
	1 Edge Name	1	Edge Name
	2 Site	2	Site
	3 Edge Group	3	Edge Group Name
	4 Network Interface for Internal Edge Communication	4	NIC name (ex: Port LAN 1)
➤ Network Interfaces→Interface Name →External Trunks			
Parameter Name			
1 Use the following trunks	1	Check	
2 select External Trunks	2	Trunk to Primary a-SBC Trunk to Secondary a-SBC	
➤ Network Interfaces→Interface Name →Phone Trunks			
Parameter Name			
1 Use the following trunks	1	Check	
2 select Phone Trunks	2	Phone Trunk Name	

4.4 Call routing

Call routing for Primary a-SBC and Backup a-SBC is set at the “site level”

Steps	Parameter Name	Value (example)	
Site Configuration			
Step 1	➤ Telephony→Sites→ Outbound Routes →« Outbound Route Name »		
	➤ General		
	Parameter Name		Value
	1 Outbound Route Name		1 Default Outbound Route
	2 Description		2 <i>Description</i>
	3 State		3 Enabled
4 Classifications		4 Emergency, National, International	
➤ distribution Pattern			
Parameter Name			
1 Sequential		1 Check	
2 Random		2 Uncheck	
➤ External Trunks			
Primary a-SBC Trunk Name (at the top)			
Secondary a-SBC Trunk Name (at the bottom)			

4.5 N+1 Redundancy

Steps	Parameter Name	Value (example)												
Site Configuration														
Step 1	➤ Telephony→Sites→ « Site Name »													
	➤ General													
	<table><tr><th></th><th>Parameter Name</th><th>Value</th></tr><tr><td>1</td><td>Site Name</td><td>1 Site Name</td></tr><tr><td>2</td><td>Description</td><td>2 Description</td></tr><tr><td>3</td><td>Location</td><td>3 Location</td></tr></table>		Parameter Name	Value	1	Site Name	1 Site Name	2	Description	2 Description	3	Location	3 Location	
		Parameter Name	Value											
	1	Site Name	1 Site Name											
2	Description	2 Description												
3	Location	3 Location												
➤ Phone Edge Assignements														
	<table><tr><th></th><th>Parameter Name</th><th>Value</th></tr><tr><td>1</td><td>Use these Sites</td><td>1 Check</td></tr><tr><td>2</td><td>Primary Sites</td><td>2 sites list (all sites containing Edges)</td></tr><tr><td>3</td><td>Secondary Sites</td><td>3 sites list (all sites containing Edges)</td></tr></table>		Parameter Name	Value	1	Use these Sites	1 Check	2	Primary Sites	2 sites list (all sites containing Edges)	3	Secondary Sites	3 sites list (all sites containing Edges)	
	Parameter Name	Value												
1	Use these Sites	1 Check												
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3	Secondary Sites	3 sites list (all sites containing Edges)												

4.6 Call Admission Control

The CAC is managed by the "Max Calls" options in the trunk configuration. The same Pure Cloud trunk is used from one a-SBC to all edges, so this value is the global CAC, in and out, for all the edges.

5 SIP SOFTPHONE CONFIGURATION WITH EDGE

This chapter describes the Purecloud configuration for Purecloud SIP phones. All options are listed.

Two configurations menus are used :

- “Based settings” for global configuration for a phone type.
- “Phone” to create an instance of a phone type and optionally to adjust the parameters set in the base settings.

5.1 Base Phone Settings

Steps	Parameter Name	Value (example)
	Base Phone settings	

Step 1	➤ Telephony→Phone Management→Base Settings → “Base Settings Name” ➤ General <table border="1"> <thead> <tr> <th data-bbox="363 371 836 405">Parameter Name</th><th data-bbox="836 371 1310 405">Value</th></tr> </thead> <tbody> <tr> <td data-bbox="363 405 836 439">1 Base Settings Name</td><td data-bbox="836 405 1310 439">1 <i>"BaseName"</i></td></tr> <tr> <td data-bbox="363 439 836 472">2 Phone Make and Model</td><td data-bbox="836 439 1310 472">2 PurecloudSoftphone</td></tr> </tbody> </table> ➤ Phone Configuration/General <table border="1"> <thead> <tr> <th data-bbox="363 551 836 584">Parameter Name</th><th data-bbox="836 551 1310 584">Value</th></tr> </thead> <tbody> <tr> <td data-bbox="363 584 836 618">1 Firmware Version</td><td data-bbox="836 584 1310 618">1 <i>"the last version"</i></td></tr> <tr> <td data-bbox="363 618 836 651">2 Dynamic Reload</td><td data-bbox="836 618 1310 651">2 Enable</td></tr> </tbody> </table> ➤ Phone Configuration/Media <table border="1"> <thead> <tr> <th data-bbox="363 730 836 763">Parameter Name</th><th data-bbox="836 730 1310 763">Value</th></tr> </thead> <tbody> <tr> <td data-bbox="363 763 836 797">1 DSCP Value</td><td data-bbox="836 763 1310 797">1 2E (46,101110) EF</td></tr> <tr> <td data-bbox="363 797 836 831">2 RTP Audio Port Start Range</td><td data-bbox="836 797 1310 831">2 16384</td></tr> <tr> <td data-bbox="363 831 836 864">3 Preferred Codec List</td><td data-bbox="836 831 1310 864">3 <i>Codec (Audio/PCMA or Audio G729)</i></td></tr> <tr> <td colspan="2" data-bbox="363 864 836 898">DTMF Setting</td></tr> <tr> <td data-bbox="363 898 836 931">1 DTMF Payload</td><td data-bbox="836 898 1310 931">1 101</td></tr> <tr> <td data-bbox="363 931 836 965">2 DTMF Method</td><td data-bbox="836 931 1310 965">2 RTP Event</td></tr> </tbody> </table> ➤ Phone Configuration/Network <table border="1"> <thead> <tr> <th data-bbox="363 1043 836 1077">Parameter Name</th><th data-bbox="836 1043 1310 1077">Value</th></tr> </thead> <tbody> <tr> <td data-bbox="363 1077 836 1111">1 Provision</td><td data-bbox="836 1077 1310 1111">1 From Edges within the Site</td></tr> <tr> <td data-bbox="363 1111 836 1144">2 TLS Authority ID</td><td data-bbox="836 1111 1310 1144">2</td></tr> <tr> <td colspan="2" data-bbox="363 1144 836 1178">Signaling</td></tr> <tr> <td data-bbox="363 1178 836 1211">1 DSCP</td><td data-bbox="836 1178 1310 1211">1 18 (24. 011000)CS3</td></tr> </tbody> </table> ➤ Phone Configuration/Custom <table border="1"> <thead> <tr> <th data-bbox="363 1267 836 1301">Parameter Name</th><th data-bbox="836 1267 1310 1301">Value</th></tr> </thead> <tbody> </tbody> </table>	Parameter Name	Value	1 Base Settings Name	1 <i>"BaseName"</i>	2 Phone Make and Model	2 PurecloudSoftphone	Parameter Name	Value	1 Firmware Version	1 <i>"the last version"</i>	2 Dynamic Reload	2 Enable	Parameter Name	Value	1 DSCP Value	1 2E (46,101110) EF	2 RTP Audio Port Start Range	2 16384	3 Preferred Codec List	3 <i>Codec (Audio/PCMA or Audio G729)</i>	DTMF Setting		1 DTMF Payload	1 101	2 DTMF Method	2 RTP Event	Parameter Name	Value	1 Provision	1 From Edges within the Site	2 TLS Authority ID	2	Signaling		1 DSCP	1 18 (24. 011000)CS3	Parameter Name	Value
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5.2 Phone

Steps	Parameter Name	Value (example)													
Phone création and configuration															
Step 1	➤ Telephony→Phone Management→Phones→"Phone Name"														
	➤ General														
	<table><tr><th>Parameter Name</th><th>Value</th></tr><tr><td>1 phone Name</td><td>1 "PhoneName"</td></tr><tr><td>2 Base settings</td><td>2 "Base_Setting_Name"</td></tr><tr><td>3 Site</td><td>3 "site name"</td></tr><tr><td>4 Hardware ID</td><td>4 "station FQDN" where the softphone is installed</td></tr></table>	Parameter Name	Value	1 phone Name	1 "PhoneName"	2 Base settings	2 "Base_Setting_Name"	3 Site	3 "site name"	4 Hardware ID	4 "station FQDN" where the softphone is installed				
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