



Basic configuration guide

Genesys Cloud / BYOC Cloud with Orange Business SIP trunk

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

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1. GOAL OF THIS DOCUMENT

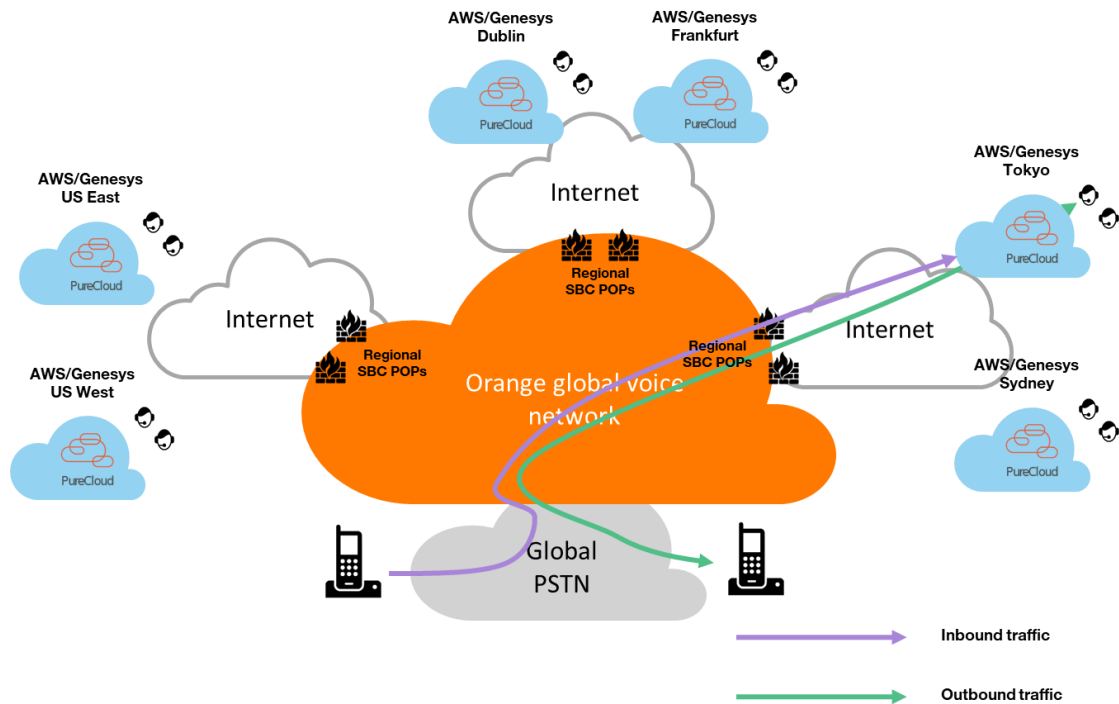
The aim of this document is to list basic technical requirements to connect France and International Orange Business SIP trunk to Genesys Cloud BYOC solution.

Please note that the whole settings required for this connection have been discussed with Genesys. Hence Orange Business invites the customer's Genesys integrator to directly contact Genesys for any further information.

Also, please be advised that this document assumes that a Genesys Cloud organization is already set up and the BYOC Cloud product added to the Genesys Cloud subscription and any needed pre-requisite configurations have been done.

Note: The settings for France Business Talk IP are the same, with the exception that it is limited to the Frankfurt and Dublin Genesys Cloud regions.

2. ARCHITECTURE OVERVIEW



3. EXTERNAL TRUNK SETTINGS FOR ORANGE

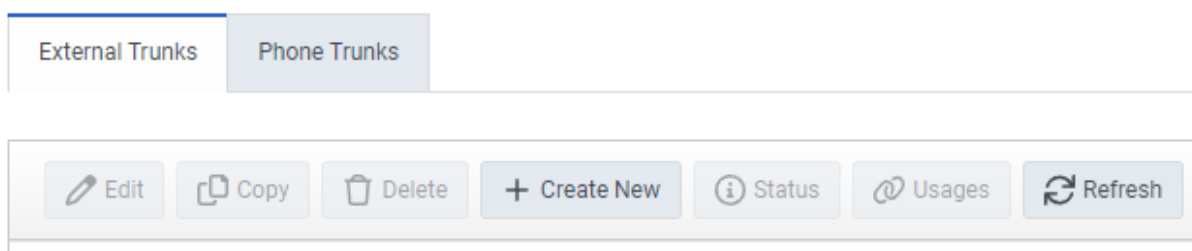
The below sections present all the steps of configuration essential for creating an external SIP trunk on Genesys Cloud for using Orange Business Talk service.

Only required settings are mentioned. Mandatory values for Orange are in bold and orange font colour. This document does not cover customer specific settings like number manipulation.

3.1 EXTERNAL TRUNK: CREATION

In Genesys Cloud Admin, under *Telephony*, click *Trunks*, then *External Trunks*.

And create a New External Trunk.



The screenshot shows the 'External Trunks' tab selected in the Genesys Cloud Admin interface. Below the tab, there is a toolbar with the following buttons: Edit, Copy, Delete, + Create New, Status, Usages, and Refresh.

For the different settings appeared select the appropriate value as indicated in the below table:

Setting	Value	Comment
External Trunk Name	Ex: Orange Business SIP Trunk	This filled is used to easily identify the trunk in the inventory, this value can and should be customized.
Type for the first list	BYOC Carrier	This value is mandatory to work with Orange Business SIP service.
Type for the second list	Generic BYOC Carrier	This value is mandatory to work with Orange Business SIP service.
Trunk State	In-Service	This parameter allows to activate/deactivate the trunk, by default it is "In-Service".
Protocol	TLS	TLS is mandatory

External Trunk Name

Orange Business SIP Trunk

Type

BYOC Carrier

Generic BYOC Carrier

Set the Trunk State to In-Service.

Select the appropriate trunk transport protocol from the **Protocol** list.

Trunk State ⓘ

In Service

Protocol ⓘ

TLS

3.2 EXTERNAL TRUNK: INBOUND SETTINGS

This section needs to be completed to configure Inbound/Termination.

Three modes are available to identify inbound traffic, TGRP, FQDN and DNIS.

Orange Business SIP service only supports the TGRP method as identifier to route the traffic.

The TGRP value is provided by Orange Business. The format of the TGRP value is an 8 alphanumeric capital characters started by "GE".

For the different settings select the appropriate value as indicated in the below table:

Setting	Value	Comment
Number Plan Site		Select a site to apply its number plan
Inbound SIP Termination Identifier	GEXXXXXX	The identifier value is provided by Orange Business.
DNIS Routing	Disabled	DNIS method must be Disabled.

Inbound

Number Plan Site

Select...

This site controls which number plan is used, both for transforms on inbound calls and for subsequent outbound transfers for calls which never flow through another site.

Inbound SIP Termination Identifier ⓘ

GEXXXXXX

Inbound SIP Termination Header ⓘ

DNIS Replacement Routing ⓘ

Disabled

Inbound Request-URI Reference

FQDN Method

INVITE sip:xxxxxxxxxxx@GEXXXXXX.byoc.mypurecloud.ie

TGRP Method ⓘ

INVITE sip:xxxxxxxxxxx;tgrp=GEXXXXXX;trunk-context=byoc.mypurecloud.ie@lb01.byoc.eu-west-1.mypurecloud.ie

3.3 EXTERNAL TRUNK: OUTBOUND SETTINGS

This section needs to be completed to configure Outbound.

Three modes are available to identify outbound traffic, TGRP, FQDN and DNIS. Orange Business Service SIP service support only the TGRP method as identifier to route the traffic. The TGRP value is provided by Orange Business. The format of the TGRP value is an 8 alphanumeric capital characters started by "GE". Other methods have to be disabled.

Setting	Value	Comment
Outbound SIP Termination FQDN		Leave this field free as Orange Business Service doesn't support FQDN method for BYOC.
Outbound SIP TGRP Attribute	GEXXXXXX	The identifier value is provided by Orange Business. It is the same value as the one used for Inbound
TGRP Context-ID	This field must contain one of the BYOC URLs of the Genesys Cloud BYOC region used.	
	byoc.euc1.pure.cloud	For eu-central-1 / Frankfurt
	byoc.euw1.pure.cloud	For eu-west-1 / Dublin
	byoc.use1.pure.cloud	For us-east-1 / US East
	byoc.usw2.pure.cloud	For us-west-2 / US West
	byoc.apne1.pure.cloud	For ap-northeast-1 / Tokyo
	byoc.apse2.pure.cloud	For ap-southeast-2 / Sydney
TGRP Context-ID	byoc.euw2.pure.cloud	For eu-west-2 / London
Outbound SIP DNIS		Leave this field free as Orange Business Service doesn't support DNIS method for BYOC

Outbound SIP Termination FQDN ⓘ

Outbound SIP TGRP Attribute ⓘ



TGRP Context-ID ⓘ



byoc.euw1.pure.cloud

Outbound SIP DNIS ⓘ



The below table specifies the settings to apply for the SIP Servers or Proxies.

Setting	Value	
SIP Servers or Proxies	This field must contain the FQDNs of Orange Business Service SBC used in a given BYOC region. For a given region 2 FQDN must be completed based on the values below. Here, Orange Business only supports FQDNs. Port is always 5061	
	Region	FQDNs
	eu-central-1 / Frankfurt	if143.nbijlbg620.emea.sbc.btip.orange-business.com
	eu-west-1 / Dublin eu-west-2 / London	if143.nbijzrb620.emea.sbc.btip.orange-business.com
	us-east-1 / US East us-west-2 / US West	if124.nbijlad620.nam.sbc.btip.orange-business.com if124.nbijlax620.nam.sbc.btip.orange-business.com
	ap-northeast-1 / Tokyo ap-southeast-2 / Sydney	if121.nbijxsp620.apa.sbc.btip.orange-business.com if121.nbijoko620.apa.sbc.btip.orange-business.com
Digest Authentication	Orange Business SIP service doesn't offer digest authentication.	

SIP Servers or Proxies ?

if143.nbijzrb620.emea.sbc.btip.orange-business.com:5061	
if143.nbijlbg620.emea.sbc.btip.orange-business.com:5061	

3.4 EXTERNAL TRUNK: CALLER ID

The below table specifies the settings to apply for Outbound Calling.

Setting	Value	Comment
Caller Address	+CCNSN E164 number format	Keep the default Phone Number selection (E.164 number) and set a DID configured on the Business Talk Site. Warning : Outbound calls will fail if this option is not set.
Caller Name	Ex : My Company Name	This name will replace the caller ID for outbound calls.
Prioritized Caller Selection	At the customer's choice	This set of options allow Genesys Cloud to pick the outbound CLI depending of a prioritized list. Refer to the Genesys Cloud documentation to know more about those options.
Suppress User Name	At the customer's choice	Refer to the Genesys Cloud documentation to know more about this option.

Caller ID

Caller Address ⓘ

E.164 number ▼

This trunk's caller address to use as the outgoing origination address.

Caller Name ⓘ

This trunk's caller name to use as the outgoing origination name.

Prioritized Caller Selection ⓘ

Add location ▼

This Trunk

⋮

Call Source (Queue / Campaign / User DID)

⋮

The caller address and name will come from the following locations. The locations are checked in order and the first found will be used.

Suppress User Name ⓘ

Never ▼

When the call source is a user, the user name can be suppressed, and the next "Caller Selection Location" will be checked.

3.5 EXTERNAL TRUNK: SIP ACCESS CONTROL SETTINGS

In order to allow incoming traffic from Orange SBC to PureCloud instance(s), it is required to whitelist Orange Business SBC signaling and media IPs.

Under SIP Access Control, build CIDR addresses to which you want to permit SIP access by entering CIDR (Classless Inter-Domain Routing) addresses and clicking Plus.

Use the controls in this section to enter and build CIDR addresses to which you want to allow SIP access.



The below table provides the Orange SBC CIDR addresses to consider for each region.

IP addresses used for SIP & Media according to Region	
Region	CIDR addresses
eu-central-1 / Frankfurt	57.67.41.248/29
eu-west-1 / Dublin	57.66.206.88/29
eu-west-2 / London	
us-east-1 / US East	57.77.35.120/29
us-west-2 / US West	57.68.164.216/29
ap-northeast-1 / Tokyo	57.73.72.40/29
ap-southeast-2 / Sydney	57.72.148.216/29

For example :

SIP Access Control ?

Allow the Following Addresses ?

57.67.41.248/29	
57.66.206.88/29	
57.67.41.253	
57.66.206.93	



3.6 EXTERNAL TRUNK: EXTERNAL TRUNK CONFIGURATION

3.6.1 General section

Scroll down to the “External Trunk Configuration” section at the bottom of the page and expand “General” settings.

Under “Calls” apply the following settings with values indicated in the below table.

Setting	Value	Comment
Call Draining	Enabled	Keep it in default state
Language	Customer’s choice	Setting the trunk to a language will set the language used in Architect flows
Max Concurrent Calls	Select Limited to: and enter the number of max simultaneous calls to be sent on the trunk.	Orange recommends putting the same value as the number of voice channels configured on the Orange SIP trunk. If the Max Calls value is above the number of voice channels configured on Orange SIP trunk and if this number is reached, the other calls will be rejected.
Max Call Rate	40/5s (default value)	Keep the default value
Max Dial Timeout	120s (default value)	Keep the default value
Max Calls Reason Code	603 (default value)	Keep the default value

General

Call Draining ?

Enabled

Language ?

English - United States (en-US)

Calls

Max Concurrent Calls ?

☒ Unlimited
 ☐ Limited to:

0

Max Call Rate ?

40/5s

Max Dial Timeout ?

120

sec

Max Calls Reason Code ?

603

3.6.2 Transport section

Scroll down to the “External Trunk Configuration” section at the bottom of the page and expand “Transport” settings

Apply the following settings with values indicated in the below table.

Setting	Value	Comment
Retryable Reason Codes	500-599	Keep the default value



Retryable Cause Codes	1-5, 25, 27, 28, 31, 34, 38, 41, 42, 44, 46, 62, 63, 79, 91, 96, 97, 99, 100, 103	Keep the default value
-----------------------	---	------------------------

▼ Transport

Retryable Reason Codes ⓘ

500-599

Retryable Cause Codes ⓘ

1-5,25,27,28,31,34,38,41,42,44,46,62,63,79,91,96,97,99,100,103

3.6.3 Identity section

Under “Inbound” apply the following settings with values indicated in the below table.

Setting	Value	Comment
Identity Type	At the customer's choice	Use this list to select the type of address you want to use for inbound identity. Available choices are: From (default) First Diversion Entry Last Diversion Entry Remote-Party-ID P-Asserted-Identity

Under “Outbound” apply the following settings with values indicated in the below table.

Setting	Value	Comment
Apply Header Privacy	At the customer's choice	Use this switch to enable or disable Genesys Cloud's capability to apply header privacy information. When enabled, an agent can use *67 to request privacy. More specifically, this prevents Genesys Cloud from sending the actual header information (typically the contact address) along with the call. Instead, Genesys Cloud replaces the actual header information with the word <i>Anonymous</i> . When disabled, an agent cannot use *67. Disable this setting if you do not want agents to represent your organization as private on a public telephony network. The default setting is Enabled.
Apply User Privacy	Disabled	Use this switch to enable or disable Genesys Cloud's capability to apply user privacy information. When enabled, an agent can use *67 to request privacy. More specifically, this prevents Genesys Cloud from sending ANI information with the call. Instead, Genesys Cloud replaces the actual ANI with <i>sip:anonymous@anonymous.invalid</i> . When disabled, an agent cannot use *67. Disable this setting if you do not want agents to represent your organization as private on a public telephony network. The default setting is Enabled. <i>Orange Business recommends Disabled value for this setting.</i>
Calling Address Omit + Prefix	Disabled	Sending the + is mandatory for Orange Business
Called Address Omit + Prefix	Disabled	Sending the + is mandatory for Orange Business

Inbound

Identity Type ?

From

Outbound

Apply Header Privacy ?

Enabled

Apply User Privacy ?

Enabled

Calling

Address Transformation ?

Match Regular Expression

Format Regular Expression

No Transformations

Match Regular Expression

Format Regular Expression

+

Address Digits Length ?

0

Address Omit + Prefix ? ↺

Disabled

Called

Address Transformation ?

Match Regular Expression

Format Regular Expression

No Transformations

Match Regular Expression

Format Regular Expression

+

Address Digits Length ?

0

Address Omit + Prefix ? ↺

Disabled

3.6.4 Media section

Apply the following settings with values indicated in the below table.

Setting	Value	Comment
Media Site	Customer's choice	The site configured here should determine the region used for media termination. Use this setting to choose the Genesys region that will be used for media termination (aka Global Media Fabric).
DSCP Value	2E (46 101110) EF	Keep the default value
Media Method	Normal	Orange Business only supports "Normal" method.



Preferred Codec List	Ex : Audio/PCMA	This section allows to choose and build a preferred list of codecs. Supported choices for Orange Business are: <ul style="list-style-type: none">audio/g729audio/PCMA (g711 A-Law)audio/PCMU (g711 μ-Law)
SRTP Cipher Suite List	AES_256_CM_HMAC_SHA1_80 AES_256_CM_HMAC_SHA1_32 AES_CM_128_HMAC_SHA1_80 AES_CM_128_HMAC_SHA1_32	Use the controls in this section to choose and build a preferred list of SRTP cipher suites to offer or allow in response. Supported choices for Orange Business are: <ul style="list-style-type: none">AES_256_CM_HMAC_SHA1_80AES_256_CM_HMAC_SHA1_32AES_CM_128_HMAC_SHA1_32AES_CM_128_HMAC_SHA1_80 <p>Note: to configure those ciphers clicking on 'Show legacy srtp ciphers' is required.</p>
Ringback	Enabled	Use this switch to enable or disable the line ringback. When enabled, this setting controls if a ringback should be generated and sent to the incoming trunk when a 18x response message that does not include an SDP is received\relayed from the outbound call. The default setting is Enabled.
Disconnect on idle RTP	Enabled	Use this switch to enable or disable the ability to disconnect a call when RTP is not received for an extended period of time. The default setting is Enabled.

Differentiated Services Code Point (DSCP)

2E (46, 101110) EF

DSCP value of Quality of Service (QoS) that will be placed in the upper 6 bits of the TOS (Type Of Service) field in the IP header of every RTP and RTCP packet.

Media Method

Normal

The method used to offer SDP when making an outgoing SIP call. Normal media sends an SDP offer in the initial SIP INVITE request. Delayed media waits for an offer SDP in a response before sending our answer SDP.

Preferred Codec List

↑ ↓	audio/PCMA	🗑
↑ ↓	audio/PCMU	🗑

Select a Codec

The preferred list of media codecs in mime format.

Ringback

Enabled

SRTP Cipher Suite List

↑ ↓	AES_CM_128_HMAC_SHA1_32	🗑
↑ ↓	AES_CM_128_HMAC_SHA1_80	🗑

Select a Cipher Suite

The preferred list of SRTP cipher suites to offer or allow in answer.

☐ Show legacy SRTP ciphers

Disconnect on Idle RTP

Enabled

Disconnect call when RTP is not received for an extended period of time.

Under “DTMF Settings” apply the following settings with values indicated in the below table.

Setting	Value	Comment
DTMF Payload	101	Use this box to specify the payload type value to use when the DTMF Method type is RTP Events. Valid range is 96–127. The default value is 101. Valid only when DTMF Method value is set to RTP Events. <i>Orange Business only supports 101 value for this setting.</i>
DTMF Method	RTP Events	Use the list to select the method to use to transmit dual-tone multifrequency (DTMF) signaling. The default value is RTP Events. There are three choices for the DTMF Method: <ul style="list-style-type: none">RTP Events: Enables out-of-band processing of events from the RTP stream (RFC 2833 or 4733).In-band Audio: Enables the processing, detection, and synthesis of events from the audio codec stream.None: Don't use a DTMF method. <i>Orange Business only supports RTP Events value for this setting.</i>

DTMF Settings**DTMF Payload**

101

Indicate the type of payload to be used when the DTMF type is specified as RTP Events. The default value is 101. Valid range is 96-127.

DTMF Method

RTP Events

The method used to transmit dual-tone multifrequency (DTMF) signaling. Set to RTP Events for RFC 2833 or 4733 compliancy.

Under “Recording” settings apply the following settings with values indicated in the below table.

Setting	Value	Comment
Record calls on this trunk	At the customer's choice	Use this checkbox to enable or disable line recording.
Require user consent before recording	At the customer's choice	Use this checkbox to enable or disable user consent request to allow recording.
Suppress Recording During	At the customer's choice	Allow to configure when a recording is automatically suppressed.
Recording Beep Tone	At the customer's choice	Allow to configure how a beep tone can be enabled.
Recording Audio	At the customer's choice	Allow to define the recording format. OPUS codec is advised because of its low storage footprint.

Recording

- ☒ Record calls on this trunk [↗](#)
- ☐ Require user consent before recording [ⓘ](#)

Suppress Recording During

- ☐ Consults [↗](#)
- ☐ Holds [↗](#)
- ☐ External bridged transfers [↗](#)

A transfer that leaves a call bridged with no active participants within Genesys Cloud.

Recording Audio

- ☒ Level the volume on both sides of the conversation [↗](#)

Audio Format

For recording transcription the format must be PCMU, PCMA, L16, or Opus, and recorded with dual channels.

Codec [ⓘ](#)

Opus

- ☒ Dual channel [↗](#)

Saves both channels in a separate stream. Only available when using PCMU, PCMA, L16, or Opus.

Recording Beep Tone

- ☐ Play periodically while recording [ⓘ](#)

Number of Tones [ⓘ](#)

- ☒ Single
- ☐ Dual

[Customize Tones](#)

Play Beep Tone Every

[14 seconds](#) [ⓘ](#)

3.6.5 Protocol section

Under “Header / Invite” apply the following settings with values indicated in the below table.

Setting	Value	Comment
Conversation Headers	Disabled	Keep it disabled
From Header Hostname	Set to default	Value for Orange Business is the default one: “Automatically generate from Edge Network interface”.
Routing Address	At the customer’s choice	Use this list to choose which field in the inbound SIP INVITE request that you want to use for routing decisions. There are two choices: To Header Request-URI
Diversion Method	Diversion Header	Use this list to choose how you want diversion information delivered to the remote end in the outbound SIP INVITE request. There are two choices: None Diversion Header Note: set to Diversion Header is the recommended choice.
Asserted Identity Header	P-Asserted-identity	Use this list to choose how you want identity information delivered to the remote end in the outbound SIP INVITE request. There are three choices: P-Asserted-identity First Diversion Entry

		Remote-Party-ID Orange Business recommends P-Asserted-identity value.
Max Diversion Entries	4	Use this box to specify the maximum number of diversion entries to be included on an outbound call.

▼ Protocol

Header / Invite

From Header Hostname ?

- ☒ Automatically generate from Edge Network Interface
☐ Custom

Routing Address ?

Request-URI

Diversion Method ?

None

Asserted Identity Header ?

P-Asserted-Identity

Max Diversion Entries ?

4

Under “User to User Information (UUI)” apply the following settings with values indicated in the below table.

Setting	Value	Comment
UUI Passthrough	At the customer's choice	Use this switch to enable or disable the sending of UUI data for outgoing calls.
Type	User-To-User	<p>Use this list to choose the type of UUI header information you want to use. There are three choices:</p> <ul style="list-style-type: none"> x-UserToUser: This is the Audiocodes proprietary header, which only includes the data. It does not use the protocol discriminator nor any of the other standard parameters and depending on the encoding you select, it is in the format: <i>x-User-to-User: hexdata</i> <i>x-User-to-User: ascii</i> User-To-User: This is the general header, which requires the use of the protocol discriminator in the format: <i>User-to-User: XXhexdata;encoding=hex;purpose=isdh-uui;content=isdh-uui</i> where XX is the protocol discriminator. User-To-User PD Attribute: This is the type that some gateways use where the protocol discriminator is specified separately in the format: <i>User-to-User: hexdata;pd=XX;encoding=hex;purpose=isdh-uui;content=isdh-uui</i> where XX is the protocol discriminator. <p>Orange Business only supports User-To-User value for this setting.</p>
Encoding Format	Hex	<p>Use this list to select the encoding format for the header. There are two choices:</p> <ul style="list-style-type: none"> Hex Ascii <p>Orange Business only supports Hex value for this setting.</p>
Protocol Discriminator	04 or 00	Use this box to specify the two-digit Hexadecimal protocol discriminator. You can specify any integers, lowercase letters from a-f, and uppercase letters from A-F.



		Note: If you select the X-UserToUser header type, this field is not available. Orange Business supports 04 or 00 value for this setting.
--	--	---

User to User Information (UUI)

UUI Passthrough

☐ Disabled

Dynamic UUI settings apply to UUI added to Architect call flows or agent scripts. All settings in this section apply to outbound only. For more information, [UUI Overview on the Resource Center](#).

Type

User-to-User

Selects the type of User To User header for UUI dynamically set within an IVR Flow or Agent Script. Does not apply to Static UUI.

Encoding Format

Hex

Sets the encoding format for dynamically set UUI.

Protocol Discriminator

00

Describes the protocol or structure used within the dynamically set UUI Data. Must be a two-digit hexadecimal value. Required when using User-To-User header type. Does not apply to Static UUI.

Under “Release Link Transfer” apply the following settings with values indicated in the below table.

Setting	Value	Comment
Take Back and Transfer	Disabled	Orange Business does not support REFER method.
Enable Release Link Transfer	Disabled	Use this switch to enable or disable the Release Link Transfer feature. The default setting is Disabled. When you enable the Release Link Transfer feature, you enable the outbound REFER method. Orange Business does not support REFER method.

Take Back and Transfer

Enable Take Back and Transfer ?

☐ Disabled

Release Link Transfer

Enable Release Link Transfer ?

☐ Disabled

3.6.6 Global Media Fabric

Prerequisite: A site with the desired Media Region(s) has to be created to be associated with the Trunk. In order to use Global Media Fabric attach the Site to the trunk in the Media section:

Media site

The media regions set on this site are used for inbound and outbound calls. The number plans of this site are also used for calls that came in on this trunk and have to be sent back out.

OBS Europe (Branch Site / BYOC Cloud)'s media regions

EMEA (Dublin)

The trunk will now use the Media Region(s) configured on the chosen Site.

3.7 EXTERNAL TRUNK: SITE ASSIGNMENT

Once the external trunk is configured, it has to be assigned to the default site attached to the Edge.

In PureCloud Admin, under *Telephony*, click *Sites*, then locate and click your site on the *Site Name* list.

Telephony / Sites / Edit Site

General | Number Plans | Outbound Routes | Simulate Call

Site Name: Customer Site

Description: Customer Site

Location: Default PCV Location

Media: Geo-Lookup TURN Disabled

Automatic Updates: Daily

Time Zone: Europe/Paris (+02:00)

Default Site: This is your default Site

Type: Branch Site

Media Model: Cloud

Phones: 4

Edge Group: PureCloud Voice - AWS

Topology Diagram: Show Topology

Click the *Outbound Routes* tab, then click *Default Outbound Route*.

From the *Select an External Trunk* list, select the external trunk configured formerly.

☰

Telephony / Sites / Edit Site

Topology

Metrics

Trunks

Sites

Edge Groups

Edges

Phone Management

Certificate Authorities

DID Numbers

Extensions

General

Number Plans

Outbound Routes

Simulate Call

+ New Outbound Route

Default Outbound Route

Delete Outbound Route

Outbound Route Name

Default Outbound Route

Description

State

Enabled

Classifications

Emergency x National x International x Network x

Distribution Pattern

☒ Sequential ☐ Random

External Trunks ?

+

-

● Orange Business SIP Trunk

🗑

Select External Trunks

Save Outbound Routes

Cancel

Setting	Value	Comment
External Trunks	Ex: Orange Business SIP Trunk	Select the external Trunk configured formerly.