



Genesys Cloud Provider SIP Trunk Configuration Guide

1	<i>Introduction.....</i>	3
2	<i>Requirements & Information.....</i>	3
2.1	GENESYS CLOUD	3
3	<i>Portal Configuration.....</i>	3
3.1	PORTAL LOGIN	4
3.2	ADD SIP TRUNK.....	4
4	<i>Cloud BYOC Carrier Configuration</i>	7
4.1	TRUNK GROUP ID & UNIQUE GENESYS ID	7
4.2	BYOC FEATURE LICENSE.....	7
4.3	EDGE GROUPS Error! Bookmark not defined.	
4.4	LOCATIONS	8
4.5	SITES	9
4.6	TRUNKS.....	11
4.7	OUTBOUND ROUTES.....	15
4.8	DID NUMBERS	16

1 Introduction

The intended purpose of this document is to provide a general guideline to configure your Genesys Cloud Provider SIP Trunk to Portal. This document does not cover advanced SIP configurations that could be used.

To access the Portal, follow the link below

<https://portal.infobip.com/apps/voice-and-webrtc/sip-trunks>

2 Requirements & Information

2.1 GENESYS CLOUD

- Your organization will be configured in a Genesys zone which can be identified by the URL you use to login to Cloud portal. For instance, in the United States:
 - **US East 1 (us-east-1)** - <https://login.myCloud.com>
 - **US East 2 (us-east-2)** - <https://login.use2.us-gov-pure.cloud>
 - **US West 2 (us-west-2)** - <https://login.usw2.pure.cloud/>
- BYOC license is required to create a BYOC Carrier Trunk (Provider trunk)
- BYOC requires a SIP Termination URI which is unique to each organization
- Incoming calls to BYOC are authenticated by IP Access Control List and Request URI SIP Termination URI
- BYOC requires E.164 format phone numbers

Genesys Cloud Login URL	Infobip Region Name (Portal)	Infobip Region Name (API)
https://login.myCloud.jp	AP Northeast 1	AP_NORTHEAST_1
https://login.apne2.pure.cloud	AP Northeast 2	AP_NORTHEAST_2
https://login.apne3.pure.cloud	AP Northeast 3	AP_NORTHEAST_3
https://login.aps1.pure.cloud	AP South 1	AP_SOUTH_1
https://login.myCloud.com.au	AP Southeast 2	AP_SOUTHEAST_2
https://login.cac1.pure.cloud	CA Central 1	CA_CENTRAL_1
https://login.myCloud.ie	EU West 1	EU_WEST_1
https://login.euw2.pure.cloud	EU West 2	EU_WEST_2
https://login.myCloud.de	EU Central 1	EU_CENTRAL_1
https://login.euc2.pure.cloud	EU Central 2	EU_CENTRAL_2
https://login.mec1.pure.cloud	ME Central 1	ME_CENTRAL_1
https://login.sae1.pure.cloud	SA East 1	SA_EAST_1
https://login.myCloud.com	US East 1	NA_US_EAST_1
https://login.use2.us-gov.pure.cloud	US East 2	NA_US_EAST_2
https://login.usw2.pure.cloud	US West 2	NA_US_WEST_2

3 Portal Configuration

We assume that you have created an Infobip Portal account and will purchase at least one DID.

3.1 PORTAL LOGIN

Login to the Portal using your login credentials. <https://portal.infobip.com/apps/voice-and-webrtc/sip-trunks>

3.2 ADD SIP TRUNK

- Click the "Create SIP Trunk" button and select "Genesys Cloud" trunk type:

Voice and WebRTC
Set up voice numbers, SIP trunks and Call Routing, Recording, Call Link and WebRTC settings. [Learn more](#)

Numbers **SIP Trunking** Call Routing Number Masking Call Link WebRTC Recording

SIP trunks
Initiate calls from your existing voice platform using Infobip infrastructure. [Learn more](#)

Showing 10

SIP trunk name	Location	Provider	Type	Status	Action status	Active calls
Genesys Test 1	New York	Genesys Cloud	N/A	Enabled	Success	0
Genesys 10x	Dubai 1	Genesys Cloud	N/A	Enabled	Success	0
Test	New York	Genesys Cloud	N/A	Enabled	Success	0
Test12212	Sandbox	Genesys Cloud	N/A	Enabled	Success	0
eeee	Dubai 1	Genesys Cloud	N/A	Enabled	Success	0
Test	Sandbox	Infobip	Static	Enabled	Success	0
Test SIP Trunk	Sandbox	Infobip	Static	Enabled	Creating	0
Test FRESHWORKS	Sandbox	Freshworks	N/A	Enabled	Creating	0
Freshworks test	Sandbox	Freshworks	N/A	Enabled	Creating	0
test 2 freshworks	Sandbox	Freshworks	N/A	Disabled	Success	0

CREATE SIP TRUNK
Infobip
Set up a static or registered SIP trunk with your SIP infrastructure
Freshworks
Set up a SIP trunk to your Freshcaller environment
Genesys Cloud
Set up a SIP trunk to your Genesys PureCloud environment

- Select the appropriate SIP channel plan:

Channels and Numbers / Voice / SIP Trunking / **Create SIP Trunk** PRICING ADD FUNDS

Choose plan Set up channels Configure settings Review order

Choose plan

Metered

Pay for your call minutes only.
[View detailed pricing](#)

Starting from
€9.00 per channel / per month

CHOOSE PLAN

Competitive channel price

Pay only for what you use

Fully automated, real-time SIP Channel management

Unlimited

US only

Unlimited calls within the US.
[View detailed pricing](#)

Starting from
€15.00 per channel / per month

CHOOSE PLAN

Predictable billing, every month

No increased costs as your business grows

Recommended for small and medium enterprises

CANCEL

- Enter the appropriate number of channels for the chosen channel plan:

The screenshot shows a web interface for setting up a SIP trunk. At the top, a breadcrumb trail reads 'Channels and Numbers / Voice / SIP Trunking / Create SIP Trunk'. To the right are links for 'PRICING' and 'ADD FUNDS', along with a help icon. A progress bar indicates four steps: 'Choose plan' (completed), 'Set up channels' (current), 'Configure settings', and 'Review order'. The main heading is 'Set up channels'. Below it, a box titled 'Number of channels' contains the instruction: 'Enter the maximum number of channels that you will need. A channel represents a single concurrent inbound or outbound call.' There is a text input field with the placeholder 'Enter number of channels'. Below the input, it says 'Estimated €0.00 per month'. At the bottom right, there are 'BACK' and 'NEXT' buttons.

- Choose the following trunk settings and click the “Next” button
 - A SIP Trunk name
 - Service Address = the primary place of use for this trunk. For more information on service addresses and why these should be correctly defined, see <https://www.infobip.com/docs/voice-and-video/sip-trunking#service-address>
 - Genesys Region = select the Region your organization is configured
 - TLS/SRTP Encryption = Enabled or Disabled
 - Enabled if you plan to do encryption using TLS/SRTP
 - Disabled if you plan to do unencrypted using UDP/RTP
 - International Calling = Disabled (Default) or Enabled

Channels and Numbers / Voice and WebRTC / SIP Trunking / Create

ADD FUNDS Balance

Choose plan Set up channels **Configure settings** Review order

Configure settings

SIP trunk name
Gen-PROD-001

Region
US West 2

Service address ⓘ
Avaz Twist Tower

TLS encryption ☒

International calls ⓘ ☒

BACK NEXT

- Validate that the trunk name, settings and Charges are the expected values and click the "Pay Now" button.

Channels and Numbers / Voice and WebRTC / SIP Trunking / Create

ADD FUNDS Balance

Choose plan Set up channels Configure settings **Review order**

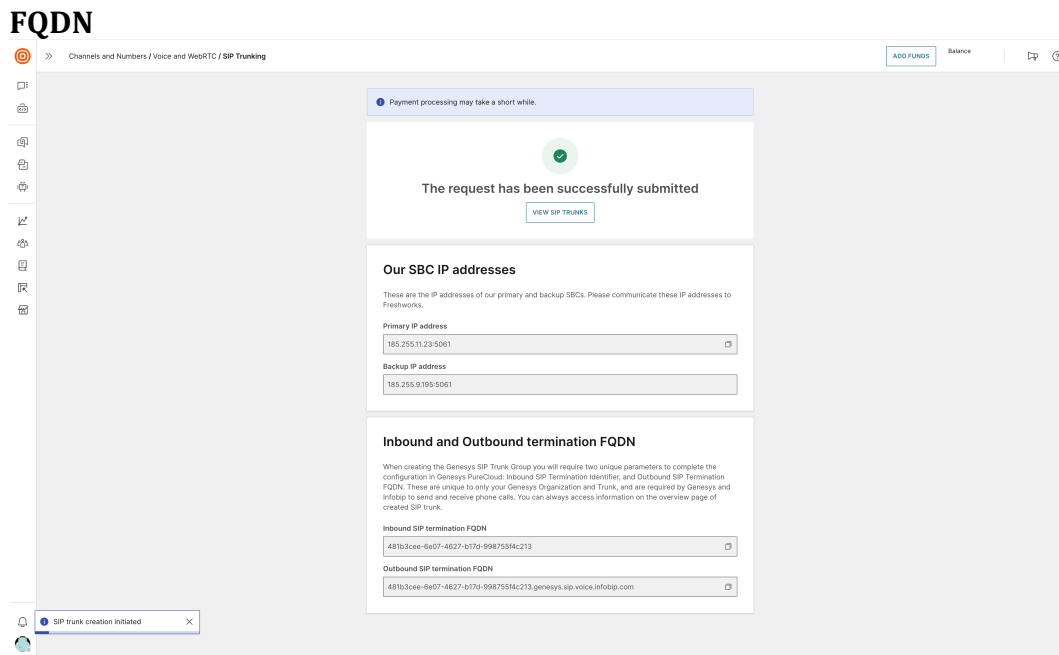
Review order

Summary	
Provider	Genesys Cloud
Plan	Metered
Number of channels	10
SIP trunk name	Gen-PROD-001
Service address	Avaz Twist Tower
Region	US West 2
TLS encryption	Enabled
International calls	Enabled

Pricing	
10x Metered SIP channels	€10.00 / month
Total	€100.00 / month
Plus taxes and regulatory fees. Billing of monthly recurring charges begins as soon as you create a SIP trunk.	

BACK PAY NOW

- On the order confirmation screen, take note of the **Infobip SBC IP address(es) and port**, **Inbound SIP termination identifier** and **Outbound SIP termination**



- Upon returning to the list of SIP trunk, monitor the status of your newly created Cisco Webex trunk until it is in status **Enabled** with an action status **Success**.
- Proceed to the Genesys Cloud Admin Portal.

4 Cloud BYOC Carrier Configuration

4.1 TRUNK GROUP ID & UNIQUE GENESYS ID

When creating the Genesys SIP Trunk Group we will require two unique parameters required to complete the configuration: **Inbound SIP Termination Identifier**, and **Outbound SIP Termination FQDN**. These are unique to only your Genesys Organization and Trunk and are required by Genesys and Portal carrier to send and receive phone calls. Both parameters are derived from the Trunk ID given at the end of section 3.2.

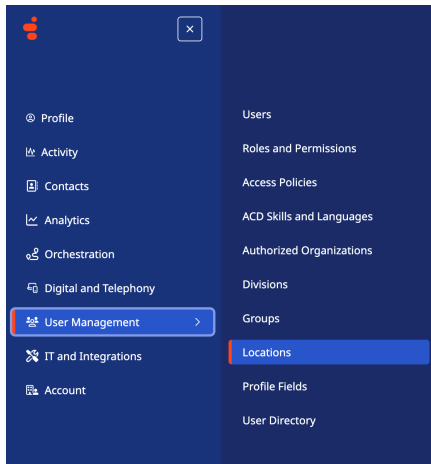
If you did not save this information while creating the trunk, go back to the list of SIP trunks on the portal (<https://portal.infobip.com/apps/voice-and-webrtc/sip-trunks>), click on your Genesys Cloud trunk and copy these identifiers from the Overview tab. The SBC IP addresses and associated ports are listed on the IP Addresses tab.

4.2 BYOC FEATURE LICENSE

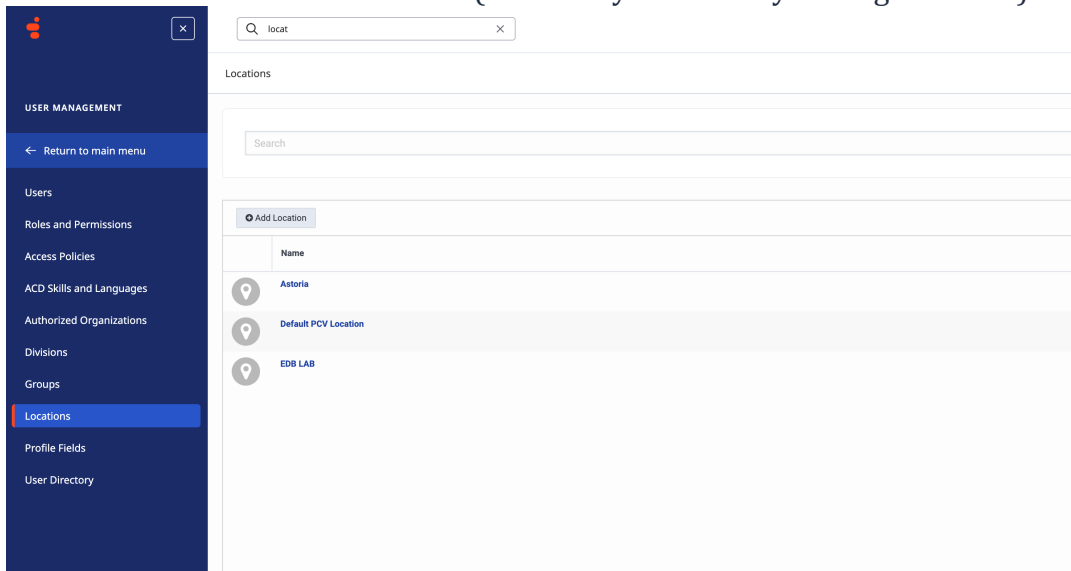
Make sure [BYOC Cloud](#) is enabled on your account.

4.3 LOCATIONS

- To validate and create Locations, navigate to User Management > Locations.



- You should see at least 1 Location (created by default in your organization).



- Click "Add Location".
 - Give the Location a Name.
 - Enter a valid Address (this address will be checked by an external web service).
 - Click "Save".
- Select "Configure the location for site emergency and availability" checkbox.
- Enter a valid Emergency Number.
- Assign a Site Contact.

- Click "Save" again.

Locations / Astoria-067863ica-c23-4744-889c-R5Pud761911

Location details

Name *
Astoria

Country
United States

Address *
868 38th St

Street Address 2

City *
Astoria

State / Province / Region *
OR

Zip / Postal Code *
97103-2443

This Address is stored with Emergency Services, in order to update the Address please create a new location with the new Address and re-associate it to the appropriate Sites.

☒ Configure the location for site emergency services and availability. Manage Site

The ESN number must be a valid phone number. Click for more information on how to Enable emergency services for a location or click on the question mark.

Emergency Number *
(112) 807-0309

☒ Use as the ARI only if the phone or user doesn't have a phone number
☐ Always use as the ARI

Notes

Site Contact
Search by name

Floors
Add Floor

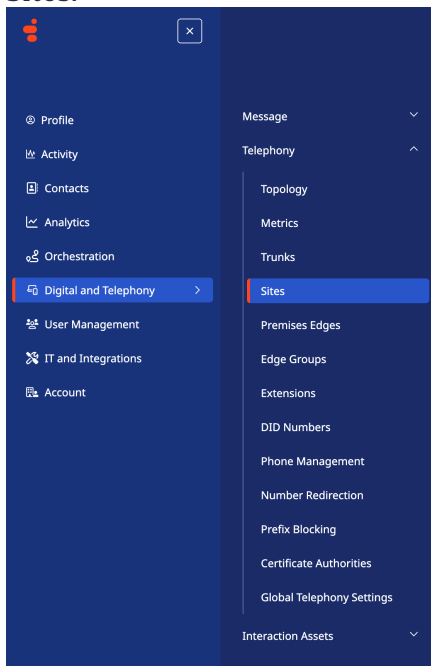
People
Change Membership

Save Cancel

Delete Location

4.4 SITES

- To validate and create Sites, navigate to Digital and Telephony > Telephony > Sites.



- You should see at least 1 Site (created by default in your organization).

The screenshot shows the Cisco Unified Communications Manager (CUCM) interface. On the left is a dark blue sidebar with the 'DIGITAL AND TELEPHONY' section expanded, showing various configuration options. 'Sites' is selected and highlighted in red. The main content area is titled 'Sites' and contains a search bar, a 'Filter' button, and a table of existing sites.

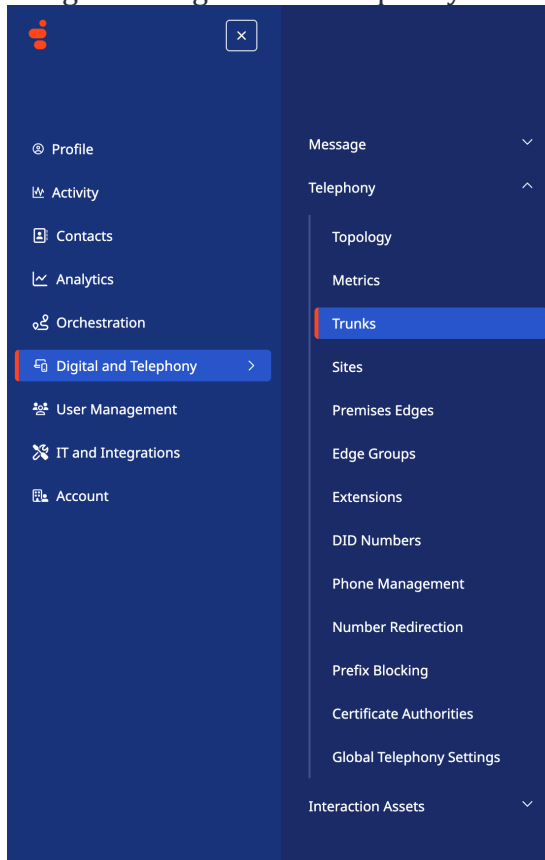
☐	Name	Status	Type	Media Model
☐	Catalaya Portal ☆ Default Site	Operational	Branch	Cloud
☐	PureCloud Voice - AWS	Operational	Core	Cloud
☐	Ribbon Wholesale	Operational	Branch	Cloud

At the bottom of the table, it shows '1 - 3 of 3' items and a '25 per page' selector.

- Click "Create New".
 - Give the Site a Name.
 - Assign the Location we just created.
 - Assign the appropriate Time Zone.
 - Click "Create Site".

4.5 TRUNKS

- Navigate to Digital and Telephony > Telephony > Trunks.



- From External Trunks tab, click "Create New".
- Give the External Trunk a Name.
- Select Generic BYOC Carrier from the Trunk Type dropdown.
- Set Trunk State to In Service.
- Select the appropriate Protocol from the Protocol dropdown:
 - Select TLS if you Enabled TLS/SRTP in section 3.2
 - Select UDP if you Disabled TLS/SRTP in section 3.2
- Input the unique identifier discussed in section 4.1 for the Inbound SIP Termination Identifier.

Create External Trunk

Basic information

Trunk name *
Infobip Trunk

Trunk type *
Generic BYOC Carrier

Trunk state
☒ In service

Status
New

Type
Generic BYOC Carrier

Transport protocol

Protocol
TLS

Inbound

SIP routing

For more information on inbound routing, see [Configure SIP routing for a BYOC Cloud trunk](#).

Termination identifier *
4cb98d55-ae4e-42f9-81c3-8920a68c2566

Your termination URI is unique within your Genesys Cloud organization's region. The termination URI will be used by the 3rd party PBX or carrier to direct SIP traffic to Genesys Cloud.

Termination header
The name of the SIP header the carrier will populate with the inbound SIP termination identifier value on inbound calls to Genesys Cloud. Only used when none of the standard identification methods are supported.

☐ DNS replacement routing
Genesys Cloud inspects the DNS portion of the inbound request-URI for the trunk's identifier. After the organization and trunk are identified, Genesys Cloud routes based on the address in the TO header.

Inbound Request-URI reference

[Carrier requirements for BYOC Cloud](#)

FQDN
INVITE sip:xxxxxxxxxxxx@4cb98d55-ae4e-42f9-81c3-8920a68c2566.byc.usw2.pure.cloud

TGRP
INVITE sip:xxxxxxxxxxxx:trg=4cb98d55-ae4e-42f9-81c3-8920a68c2566:trunk-context=byoc.usw2.pure.cloud@b01.voice.usw2.pure.cloud

- Scroll down the page to reveal more of the configuration.
- Input the Unique Outbound SIP Termination FQDN discussed in section 4.1 as the Outbound SIP Termination FQDN.
- Add the SIP Server IP and Ports one at a time, clicking the "+" button each time. These correspond to the SBC IP Addresses that were returned when creating the trunk under section 3.2. See section 4.1 if you did not copy and save the SBC IP Addresses at that time.

Outbound

SIP termination FQDN
4cb98d55-ae4e-42f9-81c3-8920a68c2566.genesys.sip.voice.infobip.com
The domain portion of the Request-URI.

SIP TGRP attribute
The value in the "trgp" parameter of the Request-URI.

SIP TGRP Context-ID
The value in the "trunk-context" parameter of the request-URI.

SIP DNIS
The DNIS in the outbound call attempt.

SIP servers or proxies

185.255.11.23:5061	
185.255.9.195:5061	

Hostname or IP Address Port

SIP servers or intermediate proxies where all outgoing requests should be sent. If no port is specified, the inbound listen port will be used.

☐ Digest authentication
Credentials to send when outbound requests are challenged with digest authentication.

Realm
The realm must match for the user name and password to be sent.

User name

Password
☐ Show password

Outbound Request-URI reference

INVITE sip:xxxxxxxxxxxx@4cb98d55-ae4e-42f9-81c3-8920a68c2566.genesys.sip.voice.infobip.com

- Scroll down the page to reveal more of the configuration.

- Validate the two SIP Server IP and were automatically added to SIP Access Control

The screenshot shows the Genesys Cloud Administration console. On the left is a dark blue sidebar with a menu under 'DIGITAL AND TELEPHONY'. The 'Trunks' menu item is highlighted. The main content area has a search bar at the top with 'locat' entered. Below the search bar, the 'Trunks' section is visible, showing a list of trunks. The 'SIP access control' section is highlighted with a red box. It contains a description of the access control list (ACL) and a list of allowed addresses. The allowed addresses are 185.255.11.23 and 185.255.9.195. Below the list is a text input field for adding more addresses. At the bottom of the console, there are tabs for 'General', 'Identity', and 'Media'.

Trunks

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.

Outbound IPs

High capacity outbound platform

Before enabling this feature, you must confirm the remote endpoint for this trunk can receive traffic from all Genesys Cloud outbound IPs. See the [resource center](#) for details.

SIP access control

Define an access control list (ACL) to limit what entities can contact this trunk. Applies to both inbound and outbound calls. HTTP provisioning is not affected by this configuration.

Allow the following addresses

185.255.11.23

185.255.9.195

Add an IP or CIDR address

Expand All Collapse All

General

Identity

Media

- Expand the Media section of External Trunk Configuration
 - Select the Site previously created in the Media Site dropdown
 - Set PMCU in the Preferred Codec List, and remove all other Codecs
 - If you Enabled TLS/SRTP in section 3.2, complete the below step for SRTP. If you Disabled TLS/SRTP in section 3.2, proceed to the next bullet point to set DTMF Payload.
 - Select "Show legacy SRTP ciphers" and add the following cipher suites below:
 - AES_CM_128_HMAC_SHA1_80

- AES_CM_128_HMAC_SHA1_32

Trunks

Media

Media site: Catalaya Portal

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.

Catalaya Portal's media regions

- Americas (US West)
- Home
- Americas (US East)
- Core

QoS DSCP value for RTP/RTCP packets: 2E (46, 101110) EF

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.

☒ Ringback

☒ Disconnect on idle RTP

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

Preferred codec list

- audio/PCMU

Select a codec

Prioritized SRTP cipher suites

- AES_CM_128_HMAC_SHA1_80
- AES_CM_128_HMAC_SHA1_32

Select a cipher suite

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

☒ Show legacy SRTP ciphers

- Set DTMF Payload to 101
- Select RTP Events from the DTMF Method dropdown

Trunks

AES_CM_128_HMAC_SHA1_32

Select a cipher suite

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

☒ Show legacy SRTP ciphers

DTMF settings

Dual-tone multifrequency (DTMF) signals generated when a caller presses a telephone key.

DTMF payload: 101

Indicate the type of payload to be used when the DTMF type is specified as RTP Events.

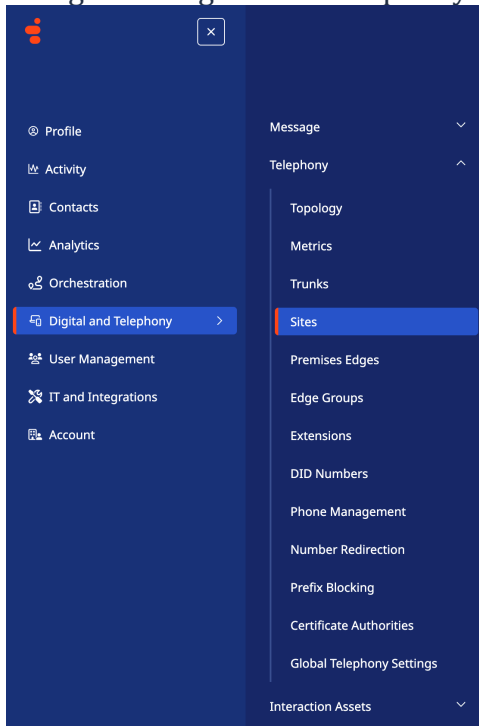
DTMF method: RTP events

The method used to transmit DTMF signaling. Set to RTP Events for RFC 2833 or 4733 compliancy.

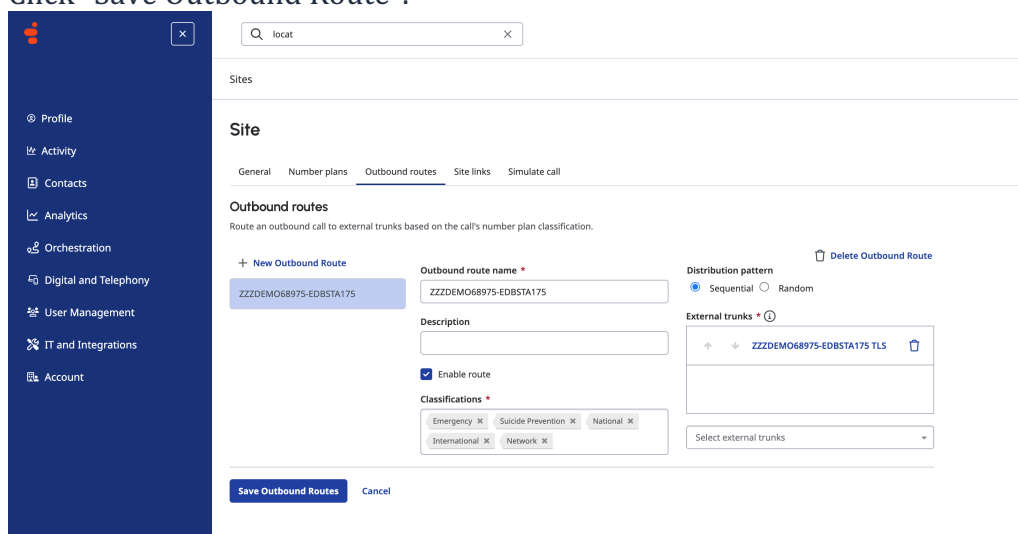
- Click "Save External Trunk" to finish and put the Trunk into service.

4.6 OUTBOUND ROUTES

- Navigate to Digital and Telephony > Telephony > Sites.



- Locate and select the Site created in previous steps.
- Select the Outbound Route tab.
 - Give the Outbound Route a name.
 - Set State to Enabled.
 - Select the previously created External Trunk from the Select External Trunk Dropdown.
 - Set the Classifications to International and National.
 - Click "Save Outbound Route".

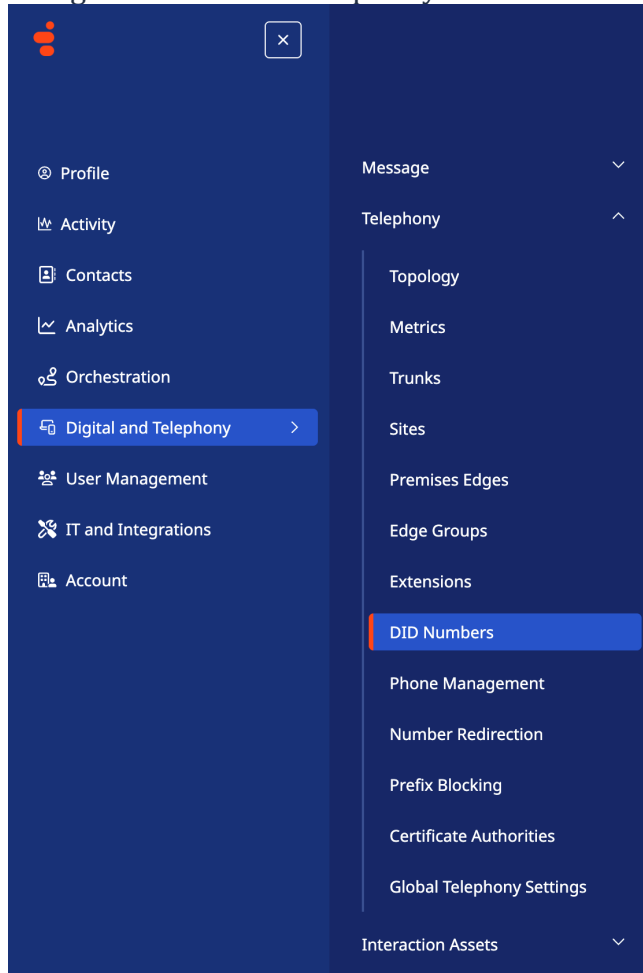


- Select the Simulate Call tab.
 - Enter a valid National or International number.

- Click "Simulate Call".
- Validate a successful response.

4.7 DID NUMBERS

- Navigate to Admin > Telephony > DID Numbers.



- Select the DID Ranges tab.

- Click "Create Range".

Search:

DID Numbers

DID Assignments DID Ranges

Create Range

☐	DID Range
☐	+1 218-607-0540 -> +1 218-607-0540
☐	+1 312-766-1518 -> +1 312-766-1518
☐	+1 646-558-2928 -> +1 646-558-2928

- Select the appropriate Country from the DID Start dropdown.
- Enter the DID that was purchased from the Infobip Portal in the DID Start.
- Select the appropriate Country from the DID End dropdown.
- Enter the DID that was purchased from the Infobip Portal in the DID End (enter the same as DID Start if only one DID was purchased).
- Enter the Service Provider.
- Click "Save".

Search:

DID Numbers

DID Assignments DID Ranges

Create Range

☐	DID Range	Service Provider	Comments
☐	+1 218-607-0540 -> +1 218-607-0540	Peerless	
☐	+1 312-766-1518 -> +1 312-766-1518	Peerless	
☐	+1 646-558-2928 -> +1 646-558-2928	Peerless	

Create Range

DID Start:

DID End:

Service Provider:

Comments:

1 - 3 of 3 25 per page

Save Cancel

- Assign this number to a Person from the DID Assignments tab.



DIGITAL AND TELEPHONY

← Return to main menu

Message

Telephony

Topology

Metrics

Trunks

Sites

Premises Edges

Edge Groups

Extensions

DID Numbers

Phone Management

Number Redirection

Q locat

DID Numbers

DID Assignments

DID Ranges

⊕ Assign

View:

DID Range: +12186070540 +12186070540

☐	DID Number	Assignee
☐	+1 218-607-0540	