

Provisioning on the Genesys Portal

Requirements

After the SIP Trunk has been provisioned on the Gamma Portal, you should have the following configuration information:

- Gamma endpoint reference
- Gamma outbound SIP termination FQDN
- Gamma signalling addresses
- Gamma SIP Trunk channel capacity
- Allocated numbers

Provisioning on the Genesys Portal

This article uses screenshots of the Genesys Portal to illustrate the provisioning process. Screenshots in this document may not be identical to the Genesys Portal as Genesys continues to evolve their product.

Please refer to the [Genesys Cloud Resource Centre](#) for detailed instructions.

To begin, select Genesys PureCloud Admin -

The screenshot displays the Genesys Cloud Admin dashboard. At the top, there is a navigation bar with tabs for Activity, Directory, Documents, Performance, Reports, and Admin. Below this, the main content area features a 'Current Task' section titled 'Welcome to Genesys Cloud!' with a 'Begin Task' button and a 'Mark this complete' link. To the right, it indicates a 'Time to complete' of 30-45 minutes and lists 'What you'll need' for setup, including location information, hardware, and DID numbers. A search bar is positioned below the task section. The bottom half of the dashboard is a grid of menu categories: Account Settings, People & Permissions, Directory, Integrations, Documents, Telephony, Genesys Cloud Voice, and Contact Center. The 'Telephony' category is highlighted, and its sub-items include Topology, Metrics, Trunks, Sites, Edge Groups, Edges, Phone Management, and Certificate Authorities.

Telephony - Sites

Under the Telephony menu, select Sites and create the sites required for the organisation. Please refer to the [Genesys Cloud Resource Centre](#) for detailed instructions.

Telephony - Trunks

Under the Telephony menu, select Trunks.

In the External Trunks tab, select Create New.

External Trunk Name	State	Listen Port	Protocol	Recording
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External Trunk Name

Type

Trunk State In Service

Status New
Type

Protocol

Configuration Setting	Value	Notes
External Trunk Name	Enter a descriptive name for the trunk	
Type	Select BYOC Carrier and Generic BYOC Carrier	
Trunk State	This toggle is used to enable or disable the trunk. The default is In-Service	
Protocol	Select TLS	TLS is mandatory

Inbound

Numbering Plan Site

Select the site which will use the trunk.

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

Genesys Purecloud provides three methods to identify traffic (TGRP, FQDN and DNIS).

Gamma SIP trunks only support the TGRP method. Each Gamma SIP trunk provisioned will have a unique identifier, referred to as the Gamma Endpoint Reference.

Configuration Setting	Value	Notes
Inbound SIP Termination Identifier	Enter the Gamma Endpoint Reference provided when the SIP trunk was provisioned	The Gamma Endpoint Reference will be a 14-character alphanumeric identifier with prefix of CG.
Inbound SIP Termination Header	Leave blank	
DNIS	Select Disabled (default)	

Outbound

Outbound

Outbound SIP Termination FQDN ?

Outbound SIP TGRP Attribute ?

TGRP Context-ID ?

Outbound SIP DNIS ?

SIP Servers or Proxies ?

Enter at least one address below

Rectangular Strip

Hostname or IP Address

Port

+

Configuration Setting	Value	Notes
Outbound SIP Termination FQDN	Enter the Gamma outbound SIP Termination FQDN provided when the Gamma SIP trunk was provisioned	
Outbound SIP TGRP Attribute	Enter the Gamma Endpoint Reference provided when the Gamma SIP trunk was provisioned	
TRGRP Context-ID	Enter the Gamma Outbound SIP Termination FQDN provided when the Gamma SIP trunk was provisioned	
Outbound SIP DNIS	Leave blank	
SIP Servers or Proxies	Enter the Gamma Outbound SIP Termination FQDN provided when the Gamma SIP	
Hostname or IP Address	Enter the Gamma Outbound SIP Termination FQDN with no port specified then click on the + symbol	
Port	Leave blank	

Digest Authentication

Digest Authentication ?

Disabled

Realm ?

User Name ?

Password ?

Show Password

Configuration Setting	Values	Notes
Digest Authentication	Select Disabled	Gamma SIP trunks do not support Digest Authentication
Realm	Leave blank	
Username	Leave blank	
Password	Leave blank	
Outbound SIP DNIS	Leave blank	

Caller ID

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.

Configuration Setting	Values	Notes
Caller Address	Select E.164 number	Gamma will only support numbers in E.164 format
Prioritized Caller Selection		Prioritised Called Selection allows configuration of Caller ID information for display to call recipients Gamma policy for Caller ID information may override this information

SIP Access Control

SIP Access Control ⓘ

Allow the Following Addresses ⓘ

100.255.1.10	
100.255.1.10	

+

Configuration Setting	Values	Notes
Allow the Following Addresses	Enter the Gamma Signalling Addresses provided when the Gamma SIP trunk was provisioned	IP addresses used by Gamma must be whitelisted on Genesys to allow incoming signalling and media to Genesys from Gamma

General

▼ General

Call Draining ⓘ

Enabled

Language ⓘ

English - United States (en-US) ▼

Calls

Max Concurrent Calls ⓘ

Unlimited

Limited to:

Max Call Rate ⓘ

Max Dial Timeout ⓘ

sec

Max Calls Reason Code ⓘ

Configuration Setting	Values	Notes
Max Concurrent Calls	Select Limited to: Enter the Gamma SIP Trunk Channel Capacity provisioned for the trunk on the Gamma Portal	The Gamma SIP trunk channel capacity is specified when the trunk is provisioned on the Gamma Portal. This capacity can be changed on the Gamma Portal.
Max Call Rate		Please refer to Gamma SIP Trunk for Genesys Cloud BYOC Service Description for default calls per second threshold for the trunk. This threshold can be increased, subject to Gamma approval.

Identity

▼ Identity

Inbound

Identity Type

From ▼

Which type of Address may be used for Inbound Identity.

Outbound

Apply Header Privacy

Enabled

If enabled, have Edge apply header privacy information.

Apply User Privacy

Enabled

If enabled, have Edge apply user privacy information.

Calling

Address Transformation

Match Regular Expression

Format Regular Expression

• No Transformations

Match Regular Expression

Format Regular Expression

+

The expressions will be applied in the order listed. If the address passes match expression, the format expression will be applied. The result format will be matched against the next expression. You can add up to three entries.

Address Digits Length

0

The number of trailing digits of the outgoing origination address that will be sent.

Address Omit + Prefix

Enabled

Exclude the + prefix of the outgoing origination address that will be sent.

Called

Address Transformation

Match Regular Expression

Format Regular Expression

No Transformations

Match Regular Expression

Format Regular Expression

+

The expressions will be applied in the order listed. If the address passes match expression, the format expression will be applied. The result format will be matched against the next expression. You can add up to three entries.

Address Digits Length

0

The number of trailing digits of the outgoing destination address that will be sent.

Address Omit + Prefix

Enabled

Exclude the + prefix of the outgoing destination address that will be sent.

Asserted Identity

Send asserted identity header ⓘ

Get asserted identity header from ⓘ

Determines what to use as the asserted identity header for calls on this trunk.

Caller address

Custom data

Name ⓘ

URI ⓘ

Configuration Setting	Values	Notes
Apply Header Privacy	Select Disabled	
Apply User Privacy	Select Disabled	
Calling - Address Omit + Prefix	Select Disabled	
Called - Address Omit + Prefix	Select Disabled	
Asserted Identity	Enter an allocated number provisioned on the Gamma SIP trunk	

Media

Media

Media site

Malthouse

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.

Malthouse's media regions

EMEA (London)

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

Select a Codec

The preferred list of media codecs in mime format.

SRTP Cipher Suite List

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

Configuration Setting	Values	Notes
Preferred Codec List	Select audio/PCMA as the only preferred Codec and remove any others in the list by clicking on the bin symbol	
SRTP Cipher Suite List	Select AES_CM_128_HMAC_SHA1_32	

Protocol

▼ Protocol

Header / Invite

Conversation Headers

 Disabled

Enables/disables inserting "x-inin-cnrv" custom conversation header with UUID value into SIP messages.

Routing Address

Which field in the inbound SIP INVITE request is to be used for routing decisions.

Asserted Identity Header

The way asserted identity information is delivered to the remote end in the outbound SIP INVITE request.

From Header Hostname

The name to be replaced as the host value of the From header on a SIP INVITE.

- Automatically generate from Edge Network Interface
 Custom

Diversion Method

The way diversion information is delivered to the remote end in the outbound SIP INVITE request.

Max Diversion Entries

Maximum number of diversion entries to be included in an outbound call.

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

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

Sets the encoding format for dynamically set UUI.

Protocol Discriminator

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.

Static User Data

 Disabled

Enables support for sending static user or billing information.

Name

Specifies the header name that will carry the static information.

Value

Specifies the value for the static UUI header. If the User-To-User Header Name is specified, this Value field must contain a protocol discriminator prefix.

Priority

Specifies the priority for the static UUI data against UUI data dynamically set within an IVR flow or Agent Script. If set to Low, dynamic data will be used if available. If set to high, static data will always be used.

Transfer

Take Back and Transfer Disabled

Allows for the REFER method and enables transferring of a local party when a transfer request is received.

Release Link Transfer (RLT) Disabled

Allows for the REFER method to be sent on the trunk to transfer the remote party to a new destination.

Outbound

Location conveyance [?](#)

Convey a geolocation when using Enhanced 911 HTTP-Enabled Location Delivery (HELD).

Custom SIP headers

SIP Headers and values that will be added to every outbound call sent on this trunk. To be used when the remote server requires extra or custom information in order to process calls.

Header	Value
No custom headers	

Header Value +

Diagnostics

For collection of information to debug issues.

Custom

Please refer to Genesys documentation.

Complete the process

Select **Save External Trunk** to complete the process.

Telephony - DID Numbers

Create Range

DID Start

Required

DID End

Service Provider

Comments

Numbers within a defined DID Range are assigned using the DID Assignments tab and can be assigned to a Person, Call Flow or Phone.

Once DID Ranges have been assigned, these numbers should be routed to the appropriate flows. Without any routing set-up, inbound calls will fail.