

Trunks

Navigation: Admin → Telephony → Trunks (or Admin → Telephony → BYOC Cloud → Trunks) **Last verified:** Genesys Cloud Resource Center — March 2026

What Are Trunks?

A trunk is the SIP communications link between Genesys Cloud and an external carrier or PBX. Trunks carry inbound and outbound SIP signaling and are consumed by Sites via Outbound Routes to send calls to the PSTN or connected telephony infrastructure.

Trunk Types

Trunk Type	Deployment Model	Used For
BYOC Carrier	BYOC Cloud	Third-party SIP carrier connectivity over the public internet
BYOC PBX	BYOC Cloud	SIP interconnect with an existing IP-PBX over the public internet
External SIP	BYOC Premises	On-premises SIP carrier or PBX connectivity through an Edge

“i BYOC Carrier and BYOC PBX are for BYOC Cloud. External SIP is for BYOC Premises. Do not mix deployment models.

Navigation

Task	Path
Open trunks	Admin → Telephony → Trunks

Task	Path
Open BYOC Cloud trunks	Admin → Telephony → BYOC Cloud → Trunks
Create a trunk	Trunks → Create Trunk
Edit a trunk	Trunks → select trunk → Edit
Use trunk in routing	Admin → Telephony → Sites → [Site] → Outbound Routes

Creating a BYOC Carrier Trunk — Field Reference

Field	Description	Notes
Name	Trunk name	Use a descriptive, consistent naming convention
Type	BYOC Carrier / BYOC PBX / External SIP	Determined by your deployment model
Subtype	Vendor/carrier profile where applicable	Optional
State	Operational state	Set to In Service when ready for production
Transport Protocol	SIP transport used to send calls	UDP / TCP / TLS — does not control inbound protocol
Inbound SIP Termination Identifier	Regionally unique ID for inbound SIP routing	Required for BYOC Carrier; confirm with carrier
Outbound Request URI	Controls SIP request routing for outbound calls	Carrier-specific
SIP Servers / Proxies	Remote SIP server or proxy addresses	Carrier-provided
Digest Authentication	SIP authentication	Enable if required by the carrier or PBX
Caller Address / Caller ID	Outbound caller number	E.164 format
Caller Name	Outbound caller name	Text
SIP Access Control	IP allowlist for inbound SIP signaling	Restrict to carrier signaling IPs only
PBX Passthrough	Enables PBX passthrough where supported	Optional
Custom SIP Headers	Additional SIP header configuration	Optional

Transport Protocol Behaviour

Protocol	Notes
UDP	Standard, connectionless — widely supported
TCP	Connection-oriented, more reliable for SIP
TLS	Encrypted SIP signaling; pairs with SRTP for full call security

“ ⚠ For BYOC Cloud, the transport protocol setting controls how Genesys **sends** calls on the trunk. It is **not enforced** on calls received on that trunk.

Step-by-Step: Create a BYOC Carrier Trunk

Step	Action
Step 1	Navigate to <code>Admin → Telephony → BYOC Cloud → Trunks</code>
Step 2	Click Create Trunk
Step 3	Select BYOC Carrier as the trunk type
Step 4	Enter the trunk Name
Step 5	Set State to In Service
Step 6	Select Transport Protocol
Step 7	Enter the Inbound SIP Termination Identifier
Step 8	Configure Outbound Request URI
Step 9	Enter SIP Servers / Proxies
Step 10	Enable Digest Authentication if required
Step 11	Under Calling, set Caller ID and Caller Name
Step 12	Configure SIP Access Control IP rules
Step 13	Save the trunk
Step 14	Add the trunk to a Site → Outbound Route

Step	Action
Step 15	Validate with test calls or Simulate Call

Dependencies

Component	Purpose
Sites	Outbound routes on sites reference external trunks
Number Plans	Classify dialed numbers before route/trunk selection
Outbound Routes	Select one or more trunks with Sequential or Random distribution
Carrier / PBX	Remote SIP endpoint the trunk connects to
Certificate Authorities	Required when using TLS trunks (BYOC Premises)

Troubleshooting

Issue	Cause	Resolution
Trunk not sending calls	Wrong transport protocol or routing config	Recheck protocol, URI, and remote endpoint requirements
Inbound calls fail	Incorrect inbound SIP identifier	Validate inbound SIP termination identifier with carrier
Secure calls fail	TLS/certificate mismatch	Validate TLS support and certificate/trust configuration
Unauthorized SIP traffic	SIP ACL not configured	Restrict signaling IPs using SIP Access Control
Wrong outbound identity	Caller ID/name misconfigured	Recheck Calling section values
Route not selecting trunk	Number plan or outbound route misconfiguration	Validate number plans, route classification, trunk selection

Quick Reference

Question	Answer
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What trunk types exist?	BYOC Carrier, BYOC PBX, External SIP
Which are for BYOC Cloud?	BYOC Carrier and BYOC PBX
Which is for BYOC Premises?	External SIP
What transport protocols are supported?	UDP, TCP, TLS
What does SIP Access Control do?	Permits signaling only from specific IP addresses
What is the Inbound SIP Termination Identifier?	A regionally unique ID used for inbound SIP routing on BYOC Carrier

Naming Convention

Resource	Example
Carrier trunk	CarrierA_BYOCCarrier_Prod
PBX trunk	CorpPBX_BYOCPBX_Test
Premises SIP trunk	HQ_ExternalSIP_Primary

Pattern: `<Provider>_<TrunkType>_<Environment>`

See Also

- **Sites** — outbound routes are configured here and reference trunks
- **Certificate Authorities** — required for TLS trunk trust (BYOC Premises)
- **Edges & Edge Groups** — BYOC Premises trunks attach to Edges
- **Architectural Build Order** — trunks are built in Phase 2

Screenshots

The screenshot shows a web interface for managing External Trunks. At the top, there are tabs for 'External Trunks' and 'Phone Trunks'. Below the tabs is a toolbar with buttons for 'Edit', 'Copy', 'Delete', '+ Create New', 'Status', 'Usages', and 'Refresh'. The main content is a table with the following columns: 'External Trunk Name', 'State', 'Listen Port', 'Protocol', and 'Recording'. A search filter is present above the table. One row is displayed: 'PureCloud Voice - AWS' with a state of 'In Service' and a recording status of 'Disabled'.

External Trunk Name

Test Trunk

Status

New

Type

Generic BYOC Carrier

Type

BYOC Carrier

Generic BYOC Carrier

Trunk State

In Service

Protocol

UDP

Inbound

Inbound SIP Termination Identifier

Inbound SIP Termination Header

DNIS Replacement Routing

Disabled

Outbound

Outbound SIP Termination FQDN

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

Hostname or IP Address

Port

+

Digest Authentication

Disabled

Realm

User Name

Password

Show Password

Calling

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.

SIP Access Control ⓘ

Allow the Following Addresses ⓘ

No addresses are currently allowed.

Add an IP or CIDR address +

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