WebRTC Phone Management

Navigation: Admin → Telephony → Phone Management **Last verified:** Genesys Cloud Resource Center — March 2026

What Is a WebRTC Phone?

A Genesys Cloud WebRTC phone is a browser or desktop app-based softphone that lets users place and receive calls directly in the Genesys Cloud client — no physical desk phone required. It is the most common phone type for contact center agents, remote users, and fast deployments.

Provisioning Model

WebRTC phones are provisioned in **two steps**:

```
Step 1: Create Base Settings
        ↓
Step 2: Create the Phone object
```

Base Settings is a shared configuration profile that defines how the WebRTC phone behaves. **The Phone** is the individual user-assigned record that uses those settings. Always build Base Settings first.

Navigation

Task	Path
Open Phone Management	`Admin → Telephony → Phone Management`
Create WebRTC Base Settings	Phone Management → **Base Settings** tab → **Add**
Create WebRTC Phone	Phone Management → **Phones** tab → **Create New**
Configure global WebRTC behavior	`Admin → Telephony → Global Telephony Settings`

Task	Path
Configure site media behavior	`Admin → Telephony → Sites → [Site] → General`
User selects WebRTC phone	Genesys Cloud client → Calls panel → phone selector

Required permission: `Telephony > Plugin > All`

Step 1: Create Base Settings

Step	Action
Step 1	Navigate to `Admin → Telephony → Phone Management`
Step 2	Open the **Base Settings** tab
Step 3	Click **Add**
Step 4	Enter a **Base Settings Name**
Step 5	In **Phone Make and Model**, select `Genesys Cloud WebRTC Phone`
Step 6	Enable **Persistent Connection** if needed
Step 7	Configure **Transport DSCP Value** and **Media DSCP Value**
Step 8	Click **Save Base Settings**

Base Settings Fields

Field	Description	Notes
Base Settings Name	Name for this configuration profile	Use a descriptive name e.g. `Support_WebRTC_Standard`
Phone Make and Model	Select the phone type	Choose `Genesys Cloud WebRTC Phone`
Persistent Connection	Keeps the WebRTC connection open after calls end	Improves subsequent call handling speed
Persistent Connection Timeout	How long the connection stays active	Configure based on call volume patterns
Transport DSCP Value	QoS marking for SIP signaling traffic	Align with enterprise voice network policy
Media DSCP Value	QoS marking for audio/media traffic	Align with enterprise voice network policy

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Step 2: Create the Phone

Step	Action
Step 1	In Phone Management, open the **Phones** tab
Step 2	Click **Create New**
Step 3	Enter the **Phone Name**
Step 4	Select the **Site**
Step 5	Select the **Base Settings** profile created in Step 1
Step 6	Assign the **User**
Step 7	Click **Save**
Step 8	Have the user select the WebRTC phone in the client and test calling

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Persistent Connection

Keeping the WebRTC connection open after a call ends allows subsequent calls to alert faster because the connection is already established.

Setting	Behaviour
Disabled	Connection closes after each call; next call requires fresh connection setup
Enabled	Connection stays open for the timeout period; subsequent calls alert immediately

> ⚠ **Important:** If you enable Persistent Connection after users are already logged in, they must **log out and back in** for the setting to apply. Genesys recommends making this change **outside business hours**.

QoS / DSCP

DSCP values mark WebRTC SIP and media traffic so your network can prioritize it. Set these values to match your organization's enterprise voice QoS policy.

DSCP Field	Applies To
Transport DSCP	SIP signaling traffic
Media DSCP	Audio/RTP media traffic

Site-Level WebRTC Media Settings

These are configured on the **Site**, not in Base Settings. Validate these for every site that will host WebRTC users.

Field	Options	Description
Media Regions	Available Home/Core/Satellite regions	Select and prioritize regions for WebRTC / Global Media Fabric
Relay/TURN Behavior	Any media region set on this site / Lowest latency via Geo-Lookup	Controls how Genesys selects TURN relay regions for WebRTC calls that need relay services

Relay/TURN Option	Best For
Any media region set on this site	Strict control — limits TURN relay to configured regions only
Lowest latency via Geo-Lookup	Best performance — Genesys dynamically selects lowest-latency TURN region

> ⚠ Forcing TURN relay can reduce resiliency and force RTP through relay services when not otherwise necessary.

User Experience

Users select and manage the WebRTC phone from the **Calls panel** in the Genesys Cloud client:

- Choose microphone and speaker device
- Adjust volume
- Run audio diagnostics

> ℹ Recommend using a quality headset rather than laptop built-in speakers and microphone to avoid echo and audio quality issues.

Troubleshooting

Issue	Cause	Resolution
User cannot answer calls reliably	Persistent connection disabled or not applied	Enable it; have user log out and back in
No audio	Wrong microphone/speaker selected	Verify audio devices in WebRTC phone client settings
Poor call quality	DSCP/network/headset issue	Check QoS policy, headset, and local network
WebRTC phone not visible to user	Phone not created or not assigned to correct user	Recheck phone assignment
Calls do not route	Site/routing configuration issue	Validate site, number plans, and trunks
Settings change not applied	User session retained old settings	Log out and back in
Unexpected TURN/media path	Site Media Regions or Relay/TURN Behavior misconfigured	Review assigned Site's General tab

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Quick Reference

Question	Answer
How do you add a WebRTC phone?	Create Base Settings first, then create the Phone
What model do you select?	`Genesys Cloud WebRTC Phone`
Why enable Persistent Connection?	Improves subsequent call handling speed
Where is Relay/TURN Behavior configured?	On the Site, not in Base Settings
What should users do after Persistent Connection is enabled later?	Log out and back in

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Naming Convention

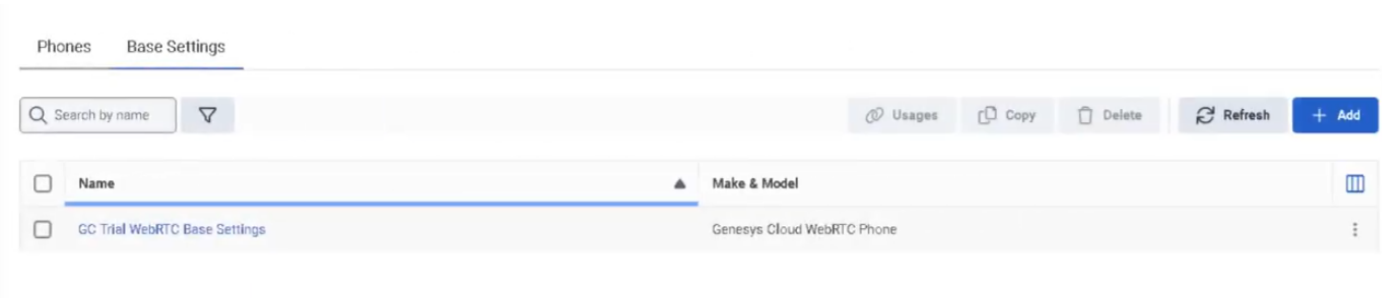
Resource	Example
Base Settings	`Support_WebRTC_Standard`
Base Settings	`Sales_WebRTC_Remote`
Phone	`AgentName_WebRTC`

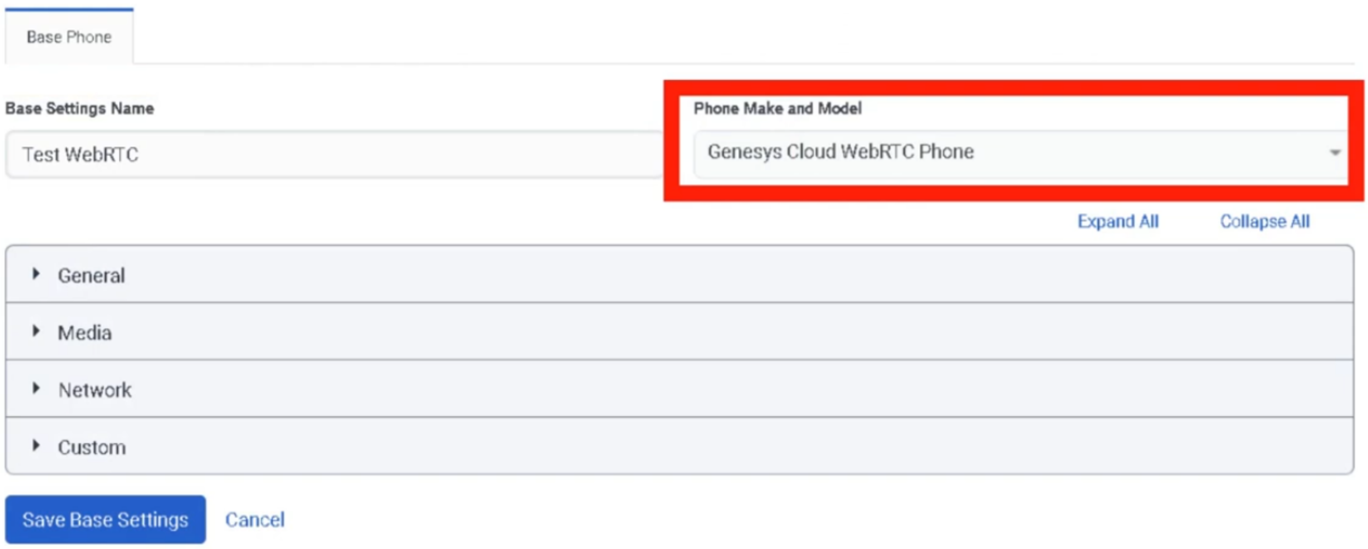
Pattern: `<BusinessArea>_WebRTC_<Purpose>`

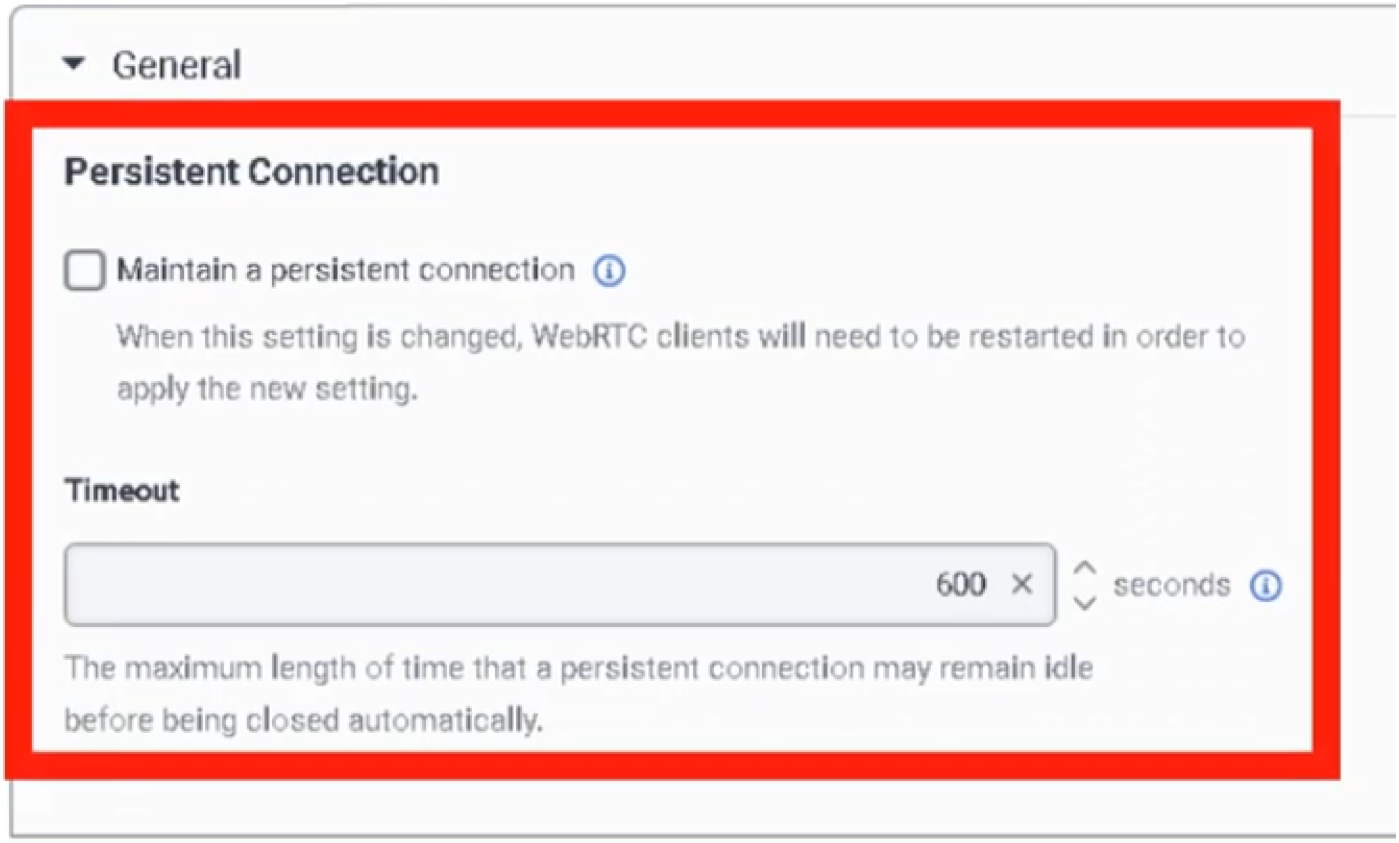
See Also

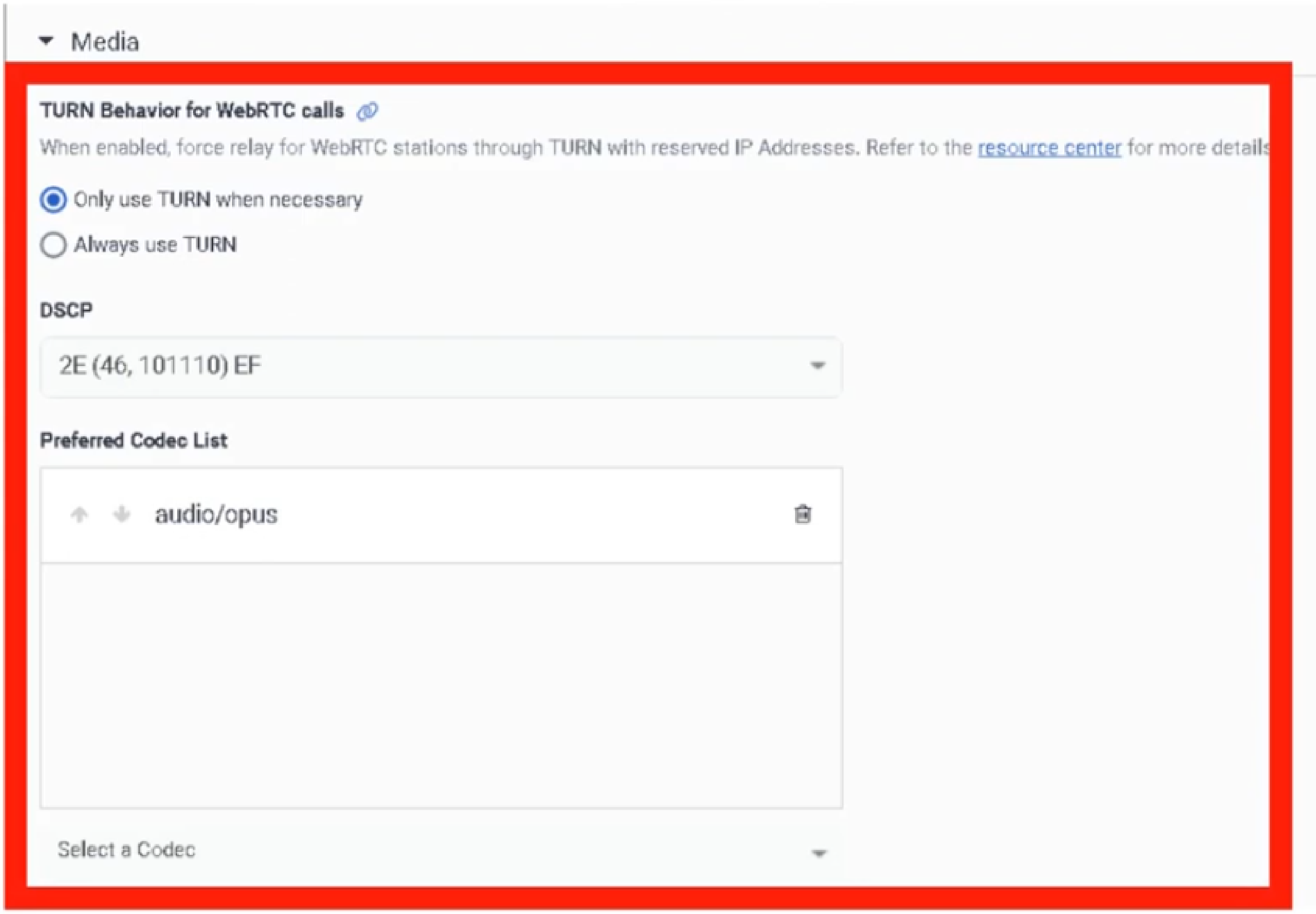
- **Sites** — Media Regions and Relay/TURN Behavior are configured here
- **Topology** — confirm phone-to-edge assignments and site relationships
- **Architectural Build Order** — phones are assigned in Phase 3 (People)

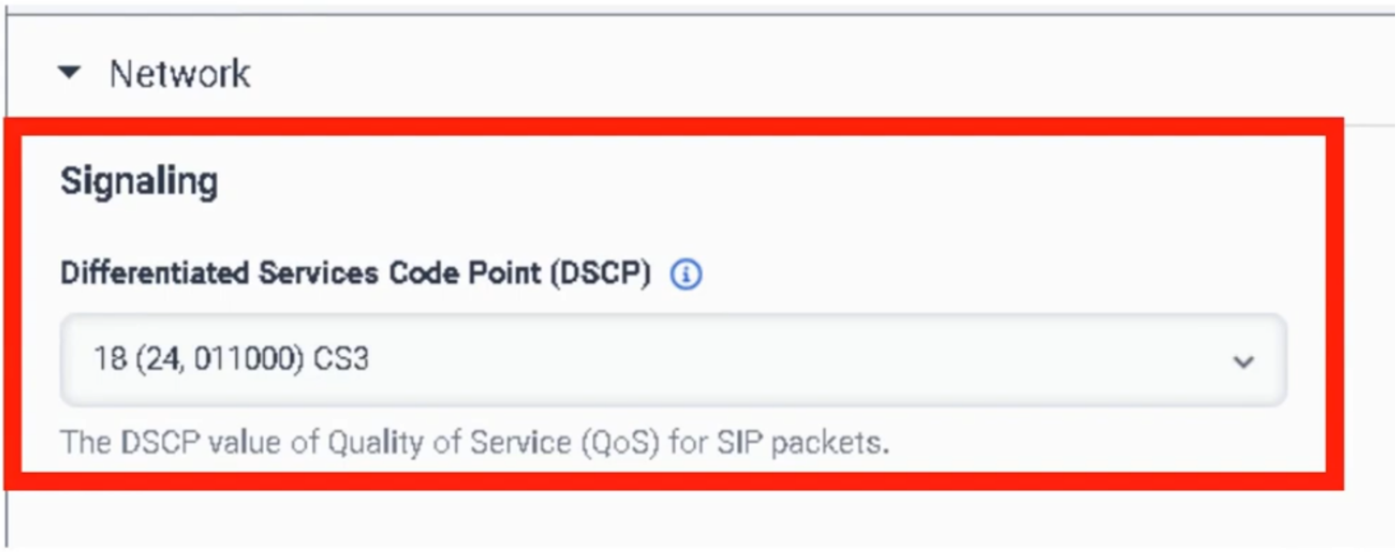
Screenshots

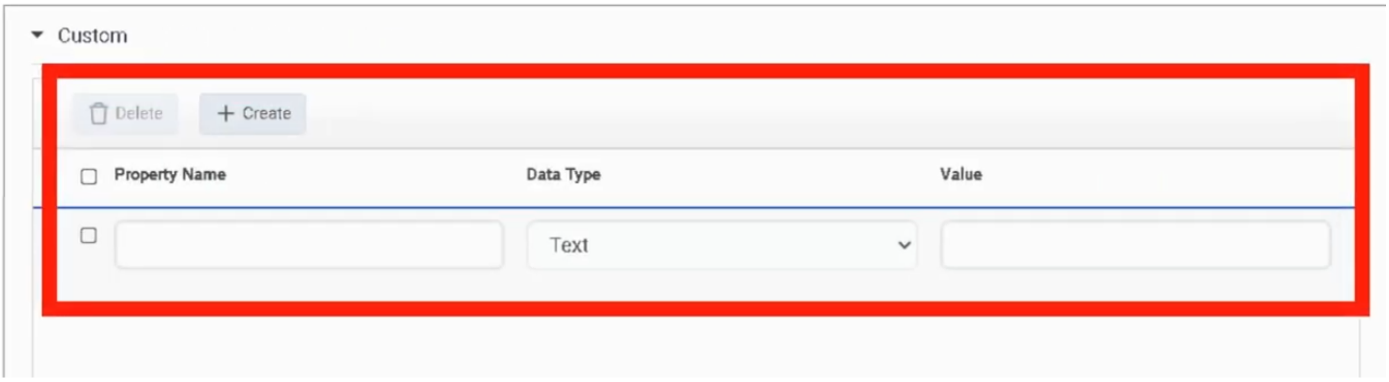


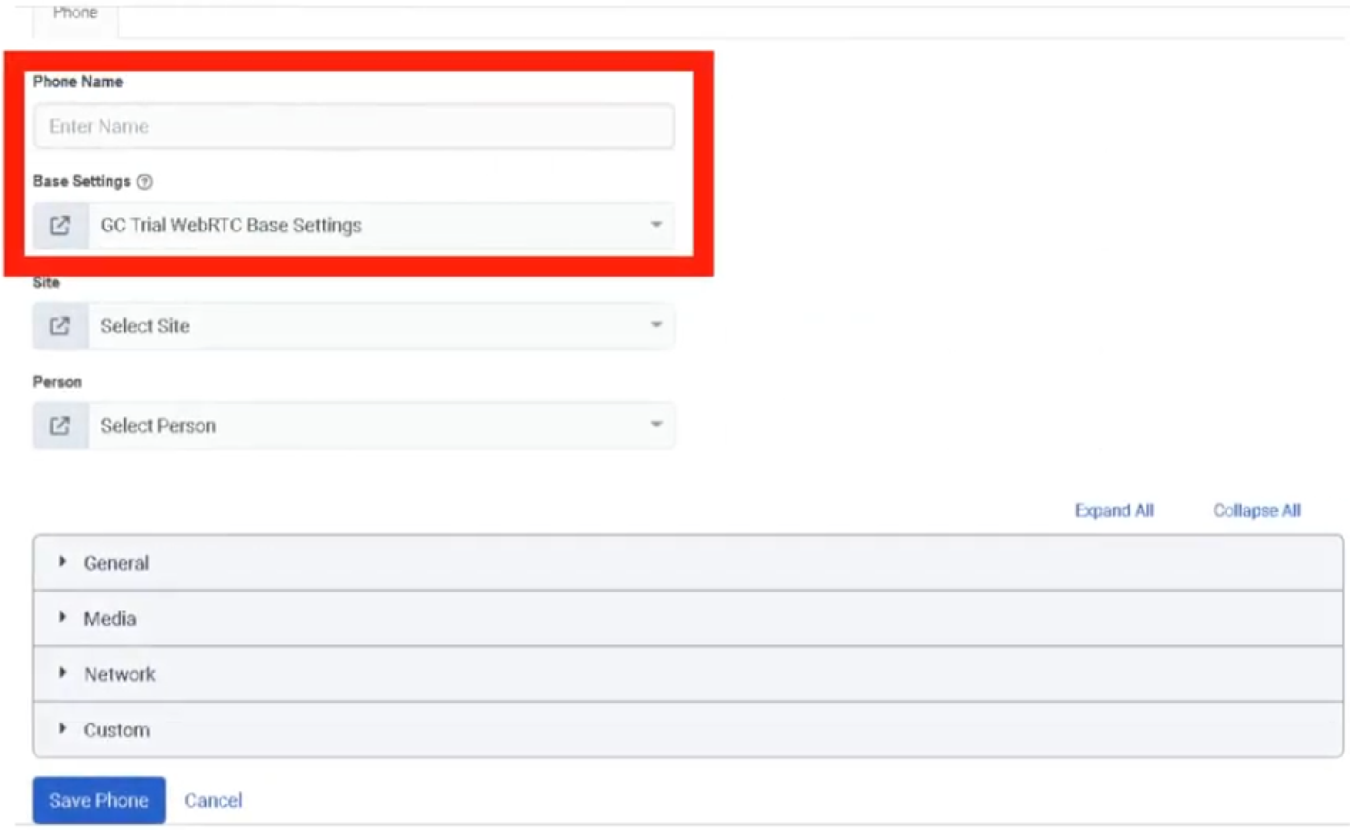



General
Persistent Connection
Maintain a persistent connection
When this setting is changed, WebRTC clients will need to be restarted in order to apply the new setting.
Timeout
600
seconds
The maximum length of time that a persistent connection may remain idle before being closed automatically.


Media
TURN Behavior for WebRTC calls
When enabled, force relay for WebRTC stations through TURN with reserved IP Addresses. Refer to the resource center for more details
Only use TURN when necessary
Always use TURN
DSCP
2E (46, 101110) EF
Preferred Codec List
audio/opus
Select a Codec


Network
Signaling
Differentiated Services Code Point (DSCP)
18 (24, 011000) CS3
The DSCP value of Quality of Service (QoS) for SIP packets.


Custom
Delete
+ Create
Property Name
Data Type
Value
Text


Phone
Phone Name
Enter Name
Base Settings
GC Trial WebRTC Base Settings
Site
Select Site
Person
Select Person
Expand All
Collapse All
General
Media
Network
Custom
Save Phone
Cancel

Phone

Phone Name

Test Remote

Base Settings

Remote

Site

Test Site

Remote Address

555-555-5555

Expand All Collapse All

- General
- Custom

Save Phone Cancel

Base Phone

Base Settings Name

Enter Name

Phone Make and Model

Remote

Expand All Collapse All

General

Incoming Caller Address

Determines what caller address is sent to this phone.

- (•) Use the trunk's prioritized caller selection rules
- ( ) Use the original caller address

  If this phone is reached over the PSTN, be sure your carrier supports sending an ANI you don't own.

Persistent Connection

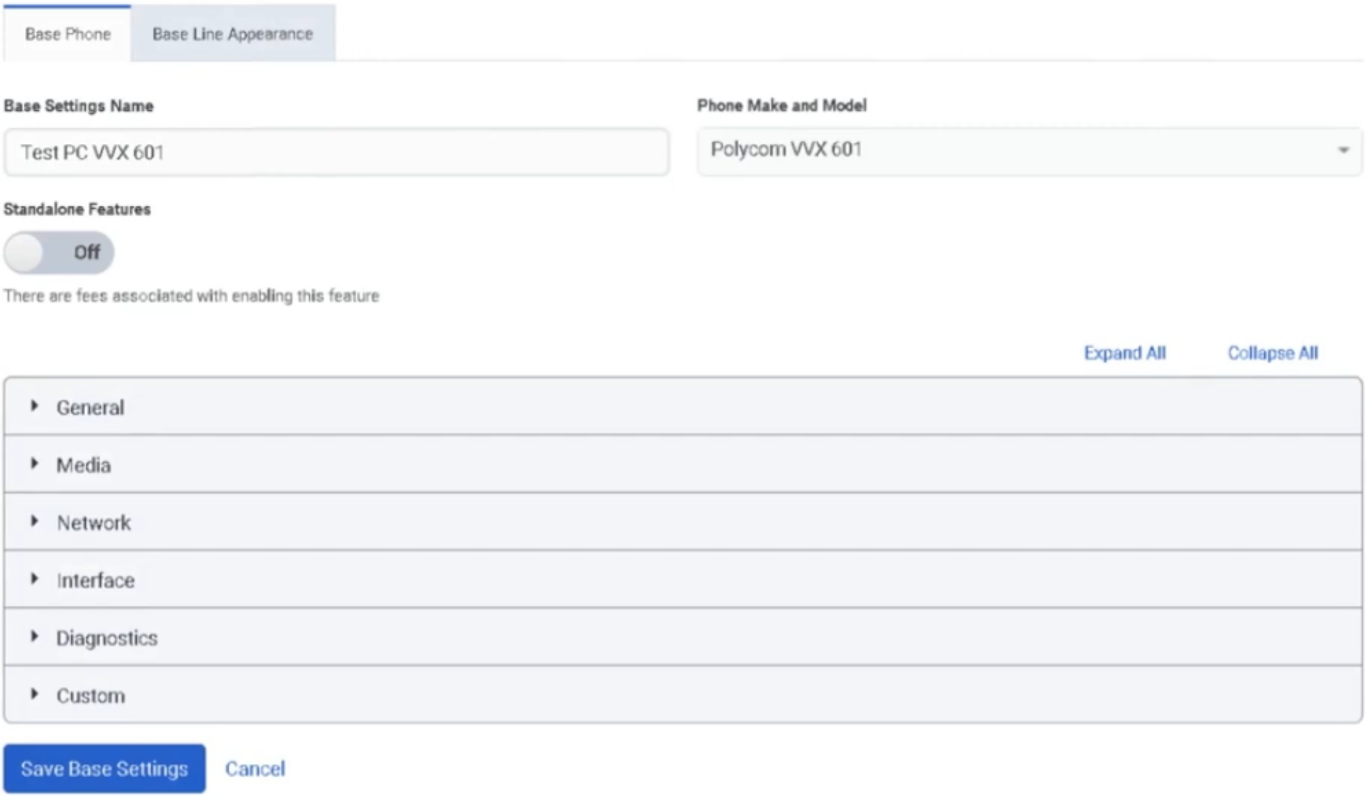
[ ] Maintain a persistent connection

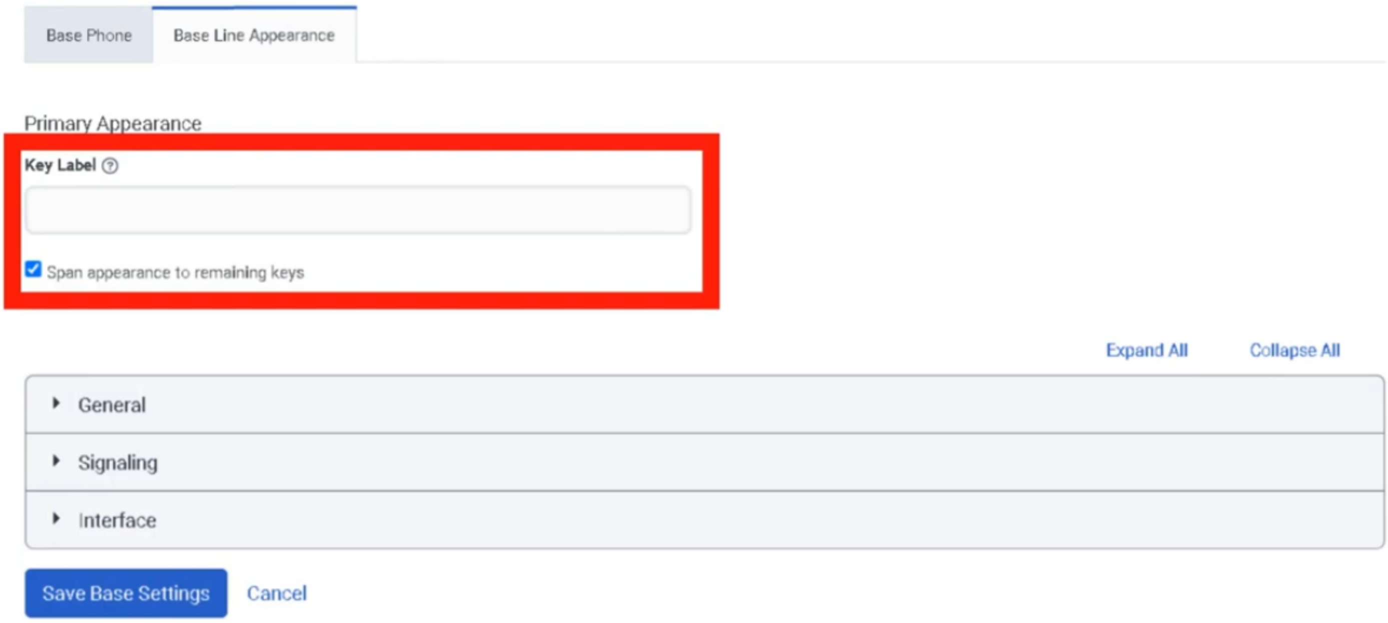
Timeout

600 seconds

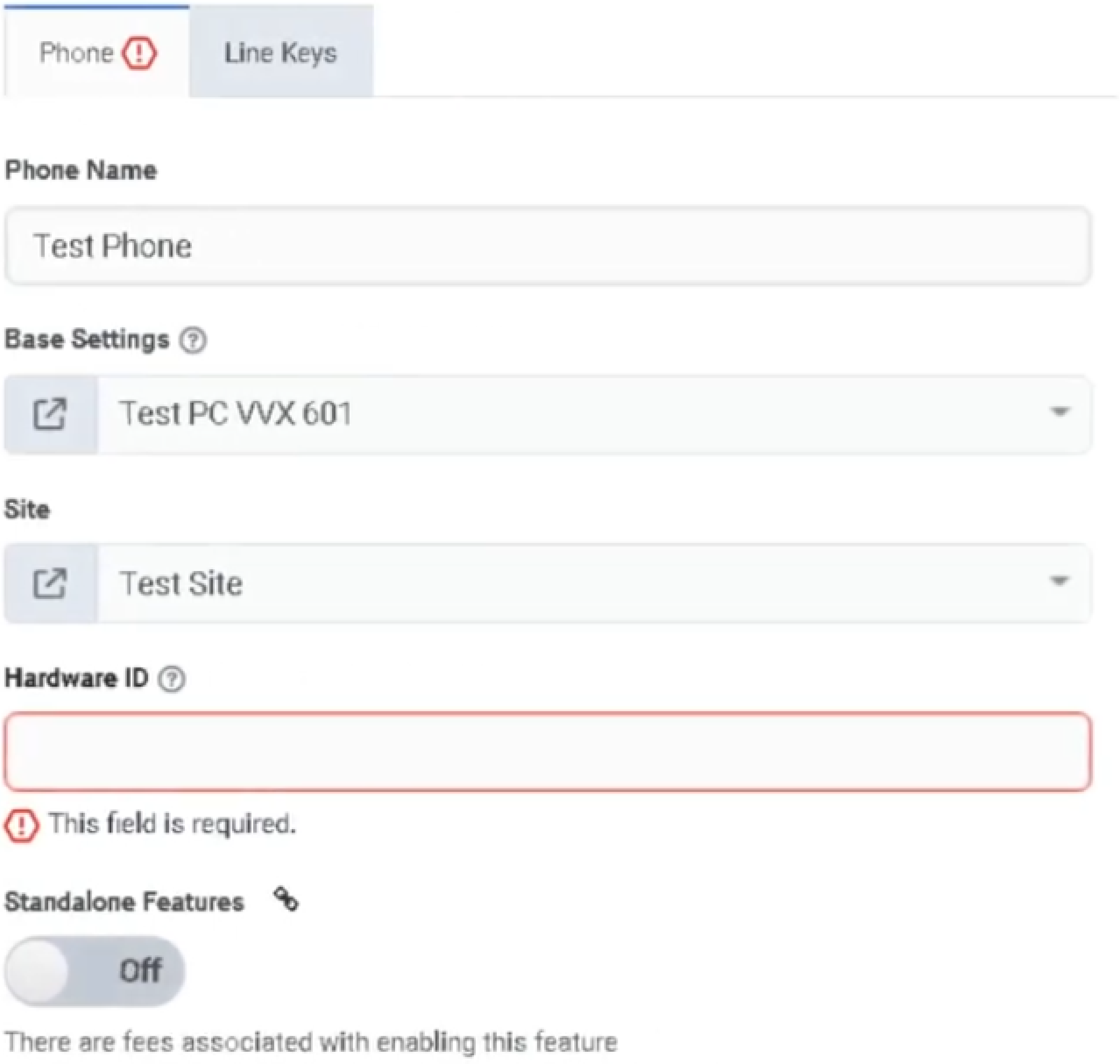
The maximum length of time that a persistent connection may remain idle before being closed automatically.

Physical Phone Hands-On





Now Add a phone



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