Call Management Portal

SIP Trunk Administration for Tenant Admins

Telesystem IT's About Trust

Blue Platform

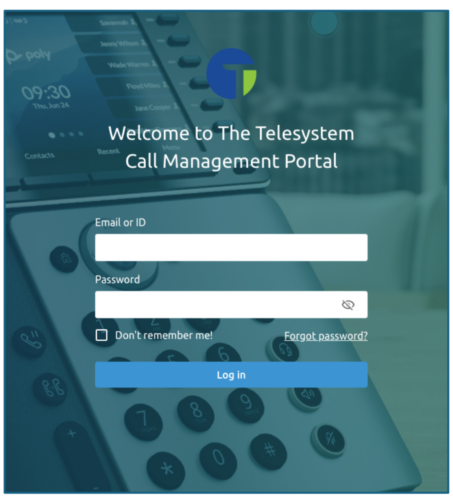
Introduction

This guide is designed for Telesystem customers using the **Blue Calling Platform** with **SIP Trunks** configured. It will help you navigate the **Blue Platform Call Management Portal** to locate and understand the settings and features associated with your SIP Trunks.

Step 1: Accessing the Blue Platform Call Management Portal

To access the Blue Platform Call Management Portal, follow these steps:

1. **Open a Supported Web Browser:**
    - Navigate to https://uccallportal.trusttelesystem.com/sign-in.
    - Supported browsers include:
        - Microsoft Edge
        - Google Chrome (recommended for best results)
        - Mozilla Firefox
        - Safari
2. **Log In:**
    - Enter your Tenant Administrator Credentials to log into the portal.



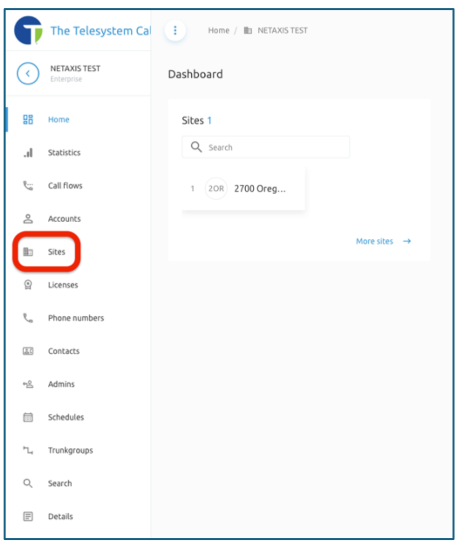
3. **If You Don't Have Credentials:**
    - An authorized contact on your account can request credentials by contacting the Telesystem Support Team:
        - Phone: 888.808.6111
        - Email: support@trusttelesystem.com

By logging in, you will gain access to the tools and features needed to manage your SIP Trunk settings and configurations.

Step 2: Locating your SIP Trunk

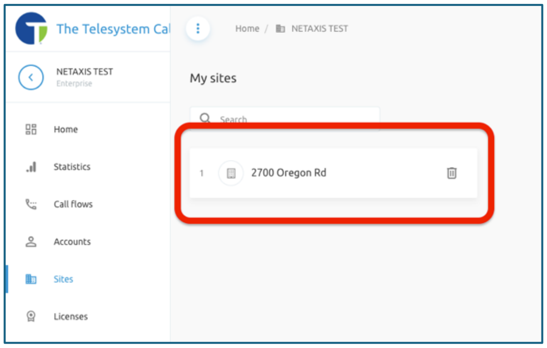
To find your SIP Trunk, follow these steps:

1. **Understand SIP Trunk Assignment:**
    - SIP Trunks are assigned at the **Group** or **Site** level within your organization.
    - Groups/Sites are often organized by:
        - **Geographical location** or **address**
        - **Affiliates**, **departments**, or other logical groupings
2. **Navigate to the Correct Site:**
    - After logging into the portal, locate the navigation pane on the **left-hand side** of the page.
    - Click the **menu button** (three horizontal lines) at the top-left corner to expand the navigation pane. This will display labels for each navigation icon.



3. **Access the "Sites" Section:**
    - Click on the “**Sites**” icon or menu item.
4. **Select the Appropriate Site:**
    - In the “**My Sites**” menu, click the button corresponding to the Site or Group where the SIP Trunk is assigned.

- Note: Some organizations may have only one site listed.

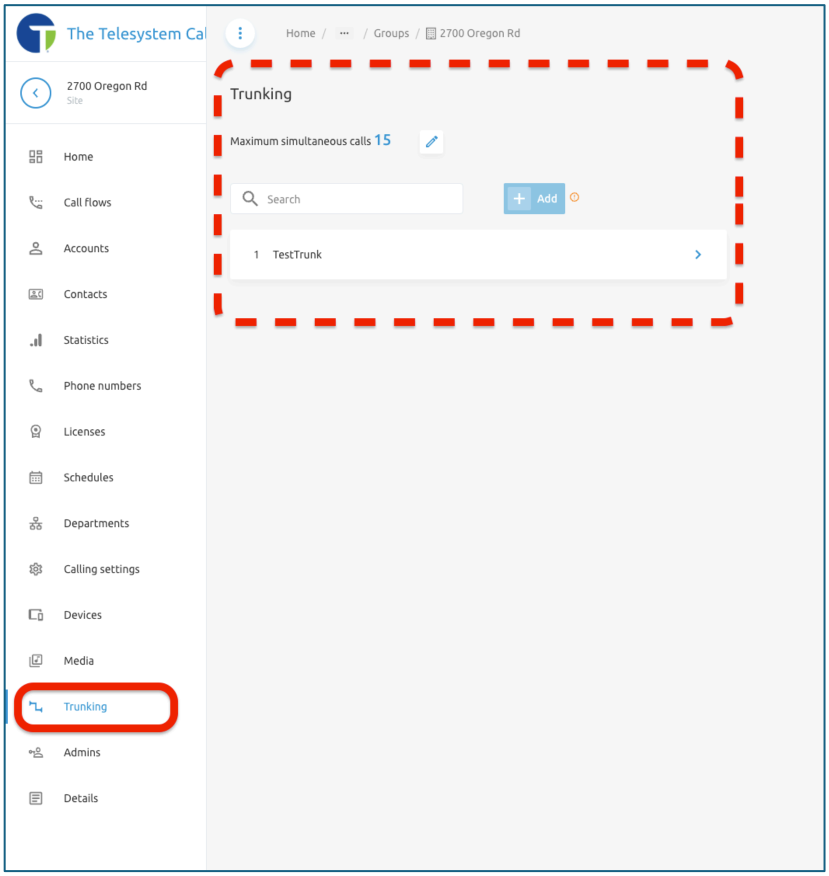


Once you select the site, you will be able to view and manage the SIP Trunk settings for that specific location.

Step 3: Access the Trunking menu

To view and manage SIP Trunks for a specific Site/Group, follow these steps:

1. **Navigate to the Trunking Menu:**
   - From the **Site/Group level** of the **Call Management Portal**, use the **left-hand navigation pane**.
   - Click on the "**Trunking**" menu item.
2. **View SIP Trunks:**
   - The SIP Trunks configured for this Site/Group will be displayed on the **right-hand side** of the page.
3. **Review Trunk Capacity:**
   - On this page, you can see the **maximum number of simultaneous calls** supported by all trunks assigned to this Site/Group.
4. **Access Specific Trunk Settings:**
   - Click on the **button or box** for the SIP Trunk name you wish to view to access its detailed configuration.



This menu allows you to review and manage settings for the SIP Trunks associated with the selected Site/Group.

Access Details

The **Access Details** menu of the SIP Trunk provides information about the authentication and network configuration for your trunk. This menu includes two dropdown sections:

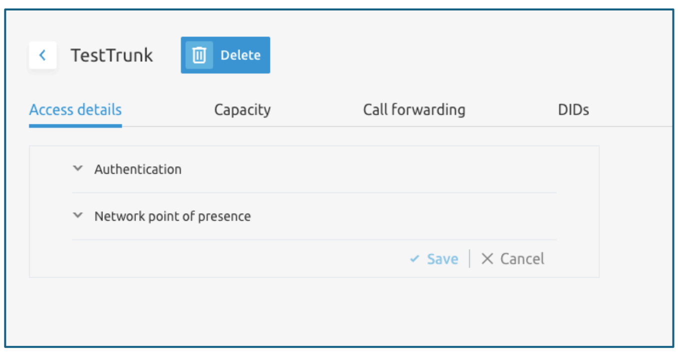
1. Authentication
2. Network Point of Presence

***Note**: The settings on this page are read-only for Tenant Administrators. To request changes, an authorized contact must reach out to the Telesystem Support Team or your Telesystem Account Manager. Any adjustments will be made by a Telesystem voice engineer.*

Authentication

Expanding the **Authentication** section reveals the following details:

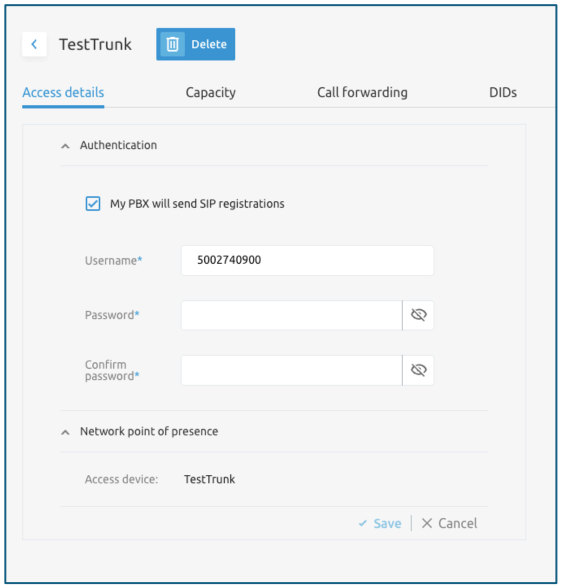
1. **My PBX Will Send SIP Registration (Checkbox):**
   - Indicates whether your PBX phone system is configured to manage its own SIP registration process with Telesystem's switching platform. When enabled, the PBX handles the initiation and management of SIP protocol registrations.
2. **Username:**
   - Displays the **authentication username** for the SIP Trunk. This username is used for registering the trunk with the Enterprise Session Border Controller (ESBC) or PBX phone system during registration-based authentication.
3. **Password and Confirm Password:**
   - Shows the **password** associated with the authentication username. These fields are used for registration-based authentication with the SIP Trunk.



Network Point of Presence

The **Network Point of Presence** section defines the connection interfaces between your PBX phone system and Telesystem's switching platform. This will vary depending on your configuration:

1. **Using an ESBC:**
   a. If Telesystem has installed an Enterprise Session Border Controller (ESBC) at your location, the device will interface with your PBX. The access device name will typically follow the format: `***address + ESBC***`
      i. For example, if your location is 123 Main St, the device may be labeled as *123MainESBC*.
2. **Direct Peering with Telesystem:**
   a. If your PBX is directly peering with the Telesystem switch (without an ESBC), you will see a name for the trunk instead of an access device.



This menu is intended for reviewing key configuration details of your SIP Trunk, ensuring that all connection points and authentication settings are accurately displayed. For any modifications, please contact Telesystem Support or your Account Manager.

Capacity

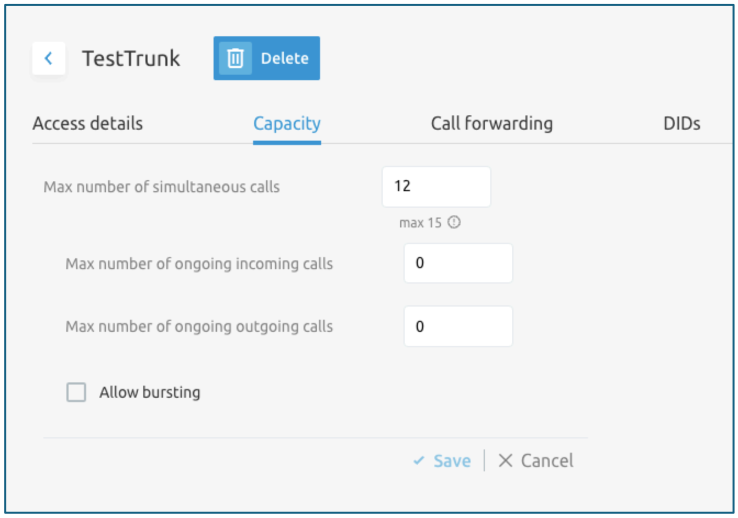
The **Capacity** menu displays the configuration details for how many simultaneous call sessions your SIP Trunk can support.

Note: *The items on this page are **read-only** for Tenant Administrators. If changes are required, an **authorized contact** must reach out to **Telesystem Support** or your **Telesystem Account Manager**. Adjustments to these settings will require a voice engineer and may impact your billing with Telesystem.*

Simultaneous Call Sessions

- The **Capacity** menu shows:
    1. The **total maximum number of simultaneous calls** the SIP Trunk can support.
    2. The **maximum number of ongoing incoming calls** (if configured).
    3. The **maximum number of ongoing outgoing calls** (if configured).
- By default:
    1. Only the **total maximum number of simultaneous calls** is configured.
    2. This setting is not limited by call direction (inbound vs. outbound).
        - For example, if the SIP Trunk is configured for **12 total calls**, it can handle any combination of inbound and outbound calls, as long as

the total does not exceed 12 at any given time.



3. **Incoming and Outgoing Call Limits:**
   - If no specific limits are configured for incoming or outgoing calls, these fields will display **"0."**
   - **Important:** This does not mean the trunk will block calls. It simply means there are no directional limits configured.

Allow Bursting

Bursting is the ability to temporarily exceed the maximum number of simultaneous calls allowed by the SIP Trunk to handle unexpected surges in call traffic.

- **Default Behavior:**
  - Telesystem does not typically enable bursting by default, as it is a chargeable feature that affects monthly billing.
- **Requesting Bursting:**
  - If you believe your SIP Trunk requires bursting capability, contact your **Telesystem Account Manager** to discuss your specific needs and the best solution for your organization.

This menu is designed to provide visibility into your trunk's capacity limits and ensure that your SIP Trunk is appropriately configured to handle your organization's call volume. For adjustments, authorized contacts can work with Telesystem to implement the necessary changes.

Call forwarding

The **Call Forwarding** options allow you to redirect all inbound calls to your SIP Trunk to an alternative destination. This feature is especially useful during situations such as:

- An outage impacting your PBX system
- Temporary closure of a location

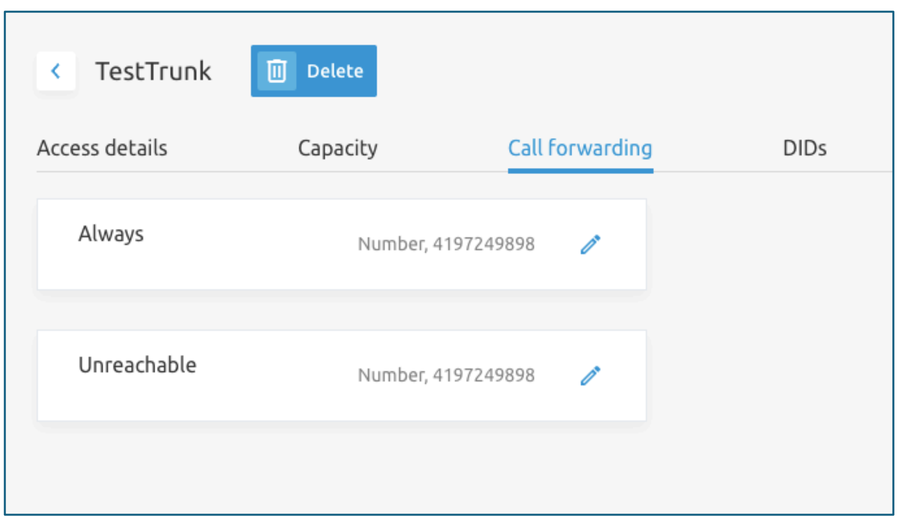
When **Call Forwarding** is enabled, all inbound calls to the DIDs associated with the SIP Trunk will be forwarded to the specified destination. Since this forwarding occurs at the **switch level**, the call will not route through your PBX phone system.

Call Forwarding Options

Two types of call forwarding are available for your SIP Trunk:

1. **Always**:
   a. This option forwards all inbound calls to the specified destination, regardless of whether your PBX or SIP Trunk is operational.
2. **Unreachable**:
   a. This option forwards inbound calls only when the SIP Trunk or PBX system is unavailable, such as during network outages or equipment failures.

These options provide flexibility to ensure business continuity and maintain communication even during unexpected disruptions.



Call Forwarding Always

The **Call Forwarding Always** feature forwards all inbound calls to a specified destination immediately upon activation and remains active until manually disabled.

- When forwarding is active, the button will display the current forwarding destination.

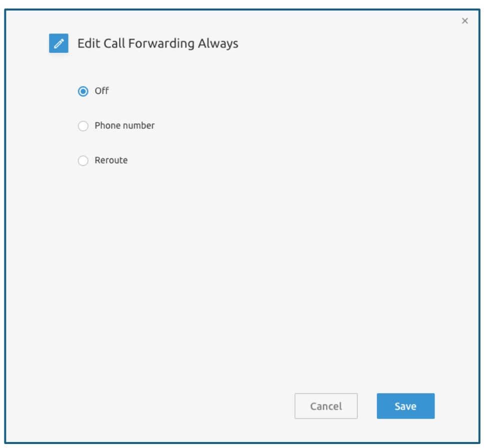
Modifying Call Forwarding Always

To change the **Call Forwarding Always** settings:

1. **Access the Edit Menu:**
   1. Click the **edit/pencil icon** next to the **Always** button.
2. **Select a Forwarding Option:**

   A pop-up window will display the following configuration options:

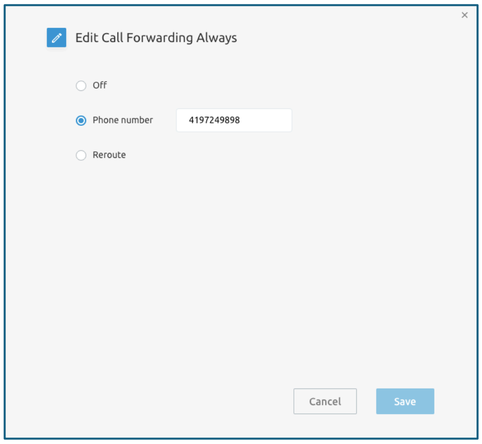
   1. **Off:**
      - Disables call forwarding, allowing inbound calls to route normally through the switch to your PBX phone system.
      - Click **Save** to confirm and apply the change.

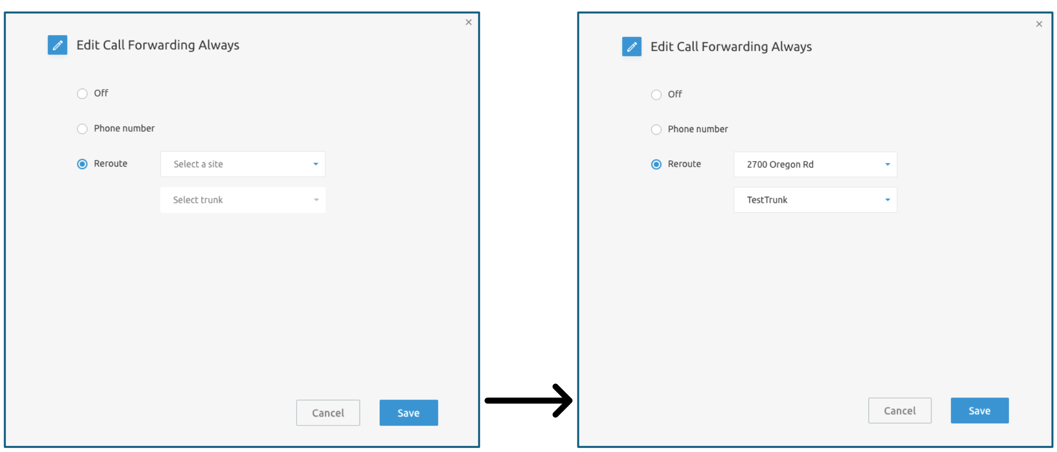
   2. **Phone Number:**
      - Select this option and enter the phone number where all inbound calls to this SIP Trunk will be forwarded.
      - **Important Notes:**
        a. The system forwards all calls to a **single phone number** for the entire trunk.
        b. For forwarding specific numbers (DIDs), configure forwarding at the DID/line level in your PBX system.
        c. To configure forwarding for individual DIDs at the switch level, additional licenses and configurations are

required. This setup is managed under the **Accounts** menu at the Group/Site level. Contact Telesystem for assistance with this option.

- Click **Save** to enable and save the forwarding.



3. **Reroute:**
   - Use this option if your organization has multiple SIP Trunks configured within the same Tenant/Organization.
   - To configure rerouting:
     a. Select **Reroute** from the menu.
     b. Use the **Select a Site** dropdown to choose the site/group containing the destination trunk.
     c. Use the **Select a Trunk** dropdown to choose the specific trunk by name.
- Click **Save** to confirm and enable the rerouting.



This feature provides flexible options for managing inbound call traffic, ensuring calls can be redirected based on your organization's needs. For more advanced configurations, contact your Telesystem Account Manager or Support Team.

Call forwarding Unreachable

The **Call Forwarding Unreachable** feature automatically redirects incoming calls to a predetermined phone number or destination when the SIP Trunk is unavailable due to issues such as:

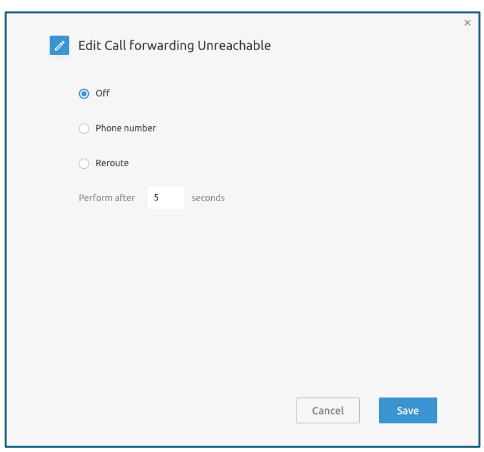
- Power outages
- Network failures
- Device malfunctions
- Registration issues

Setting up this feature in advance ensures calls are rerouted seamlessly during outages, without requiring manual intervention. Once the connection between the Telesystem switch and the PBX is restored, calls will automatically resume normal routing.

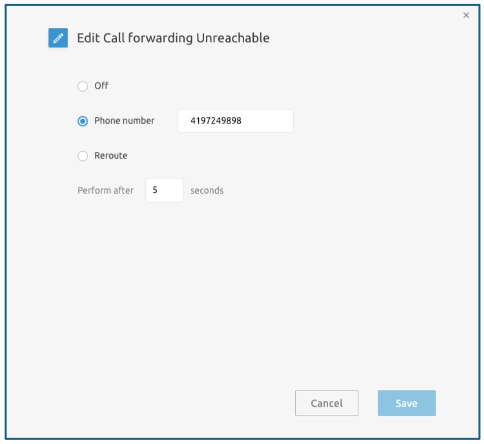
Modifying Call Forwarding Unreachable

To configure **Call Forwarding Unreachable**, follow these steps:

1. **Access the Edit Menu:**
   a. Click the **edit/pencil icon** next to the **Unreachable** button.
2. **Configure the Forwarding Option:**
   a. A pop-up window will display the following options:
      i. **Off:**
         1. Disables the feature, preventing automatic call forwarding during communication issues.
         2. Forwarding will need to be manually enabled if needed.
         3. Click **Save** to confirm and disable the feature.

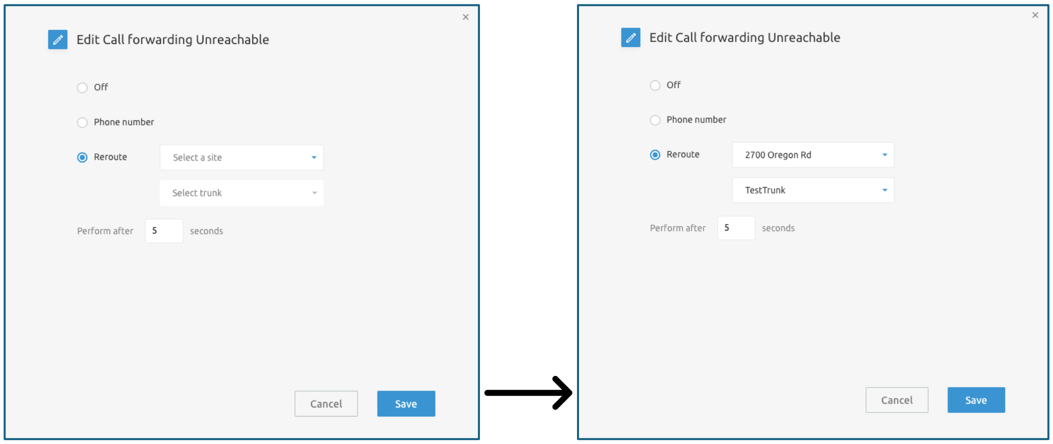


ii. **Phone Number:**
    1. Forwards all inbound calls to a single phone number when the SIP Trunk becomes unreachable.



iii. **Important Notes:**
    1. The system forwards all calls for the trunk to one destination phone number.
    2. For forwarding specific DIDs, configure this at the DID/line level in your PBX system.
    3. To forward DIDs at the switch level, additional features and licensing are required. This can be configured under the **Accounts** menu at the Group/Site level. Contact Telesystem for assistance.
    4. Click **Save** to enable and apply the forwarding configuration.

3. **Reroute:**
    a. Redirects all inbound calls to another SIP Trunk within the same Tenant/Organization.
    b. To configure rerouting:
        i. Select **Reroute** from the menu.
        ii. Use the **Select a Site** dropdown to choose the site/group where the destination trunk is configured.
        iii. Use the **Select a Trunk** dropdown to select the specific trunk by name.
        iv. Click **Save** to confirm and enable the reroute.



c. **Perform After X Seconds:**
   i. Sets the time delay (1–15 seconds) for the switch to wait for device registration before forwarding calls using the Unreachable configuration.
   ii. **Note:** Configurations outside the 1–15 second range will result in an error.
   iii. Click **Save** to apply the timing.

This feature ensures uninterrupted communication during connectivity issues and automatically restores normal routing once the SIP Trunk becomes operational. For advanced configurations or additional features, contact your Telesystem Account Manager or Support Team.

DIDs

Similar to the **Access Details** and **Capacity** menus, customer administrators cannot make changes within the **DIDs** menu. This section is designed to provide a view of the phone numbers, or **DIDs (Direct Inward Dialing numbers),** associated with the SIP Trunk.

Viewing DIDs:

Administrators can see all DIDs that have been added to the Trunk. These may appear as individual numbers or as a sequential range of numbers, depending on the configuration.

Main Number:

At the top of the page, the Main Number field displays the primary phone number assigned to the SIP Trunk. This number typically serves dual purposes:

- It acts as the primary identifier for the SIP Trunk.
- It is often the number used for authentication and may also be referred to as the "Pilot Number."

This menu is a read-only section that helps administrators quickly identify and review the DIDs associated with their trunk setup.

Enterprise Trunking/Trunkgroups Menu

At the **Tenant/Enterprise** level of the **Call Management Portal**, there is an additional menu item labeled "Trunkgroups". This menu is not configured by default for most of Telesystem's SIP Trunk customers. However, it becomes relevant for organizations using **Enterprise Trunking**.

What is the Trunkgroups Menu?

If **Enterprise Trunking** is configured, this menu displays settings designed for load balancing traffic and redundancy across multiple SIP Trunks within your organization.

Use Case for Trunkgroups:

This feature is specifically beneficial for customers who:

- Have SIP Trunks deployed across multiple sites or groups.
- Utilize multiple PBX interfaces that allow calls to be routed dynamically to any SIP Trunk within the organization.

Key Benefits and Functionality:

1. **Load Balancing**:
   - Distributes inbound and outbound call traffic across all available SIP Trunks, reducing the likelihood a single trunk reaches its maximum capacity of simultaneous call sessions.

2. **Redundancy:**

  - Provides failover capabilities by rerouting calls to other available SIP Trunks in case of outages or capacity issues on a primary trunk.
3. **Maximizing Call Capacity:**
   - The total number of simultaneous calls is calculated as an aggregate across all SIP Trunks configured for this type of routing.
   - This prevents bottlenecks on individual trunks, as calls can be distributed dynamically.
4. **Internal PBX Routing:**
   - Once calls are routed to the appropriate SIP Trunk, the PBX manages internal routing to ensure the call reaches its intended destination.

Why Use This Feature?

The **Trunkgroups** menu and **Enterprise Trunking** configuration provide organizations with a scalable, efficient, and resilient telephony solution. It is ideal for businesses with complex communication needs spanning multiple locations, offering seamless call handling and optimal resource utilization.

If your organization is interested in setting up or modifying an Enterprise Trunking configuration, please contact Telesystem through your **Account Manager** or the **Support Team** to initiate the process. Our team will guide you through the setup to ensure it meets your business requirements.