

How Incoming Calls are Handled When the Server is Unreachable

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This Best Practice describes how the SBC Edge (SBC) handles incoming calls when the Server is down or unreachable. The SBC routes the calls according to general configuration guidelines. The information below includes expected behavior for call routing when there is a problem reaching the server (i.e., registration fails, wrong configuration, etc.).

 This Best Practice assumes the following:

- Incoming call is based on Federated IP/Signaling Group listen port/transport protocol.
- Incoming call is one of the following: ISDN, SIP, or CAS.

Call Survivability when SIP Server is Down - How Calls are Routed With PSTN Backup (SIP Survivability)

 A call route to the PSTN must be created in order for calls to be re-routed to the PSTN as a backup mechanism if the SIP Server is unavailable. See [Managing SIP Server Tables](#) and [Managing Call Routing Tables](#).

When the SIP server (SIP Signaling Group) is down or unavailable, the call is routed to the PSTN as follows:

- Incoming call is allowed. Since the server is down, the Signaling Group's Server Status is set as **Down**. Instead of attempting to send calls to the server, the call proceeds to the next Destination Signaling Group (if configured based on the routing policy, which must include the PSTN). See [Managing Signaling Groups](#).
- Based on routing configuration, incoming calls are made without any post dial delay to the configured SIP or ISDN trunk.

Basic Call Mode with Agent Type as "Back-to-Back User Agent" Configuration - How Incoming Calls are handled

Below are the components of a basic configuration

- Endpoint is registered with the SBC (see [Creating and Modifying Entries in Contact Registrant Tables](#)).
- SBC's Signaling Group is configured to send requests to a server (see [Creating and Modifying Entries in SIP Server Tables](#)).

Scenario 1: Registration to server fails; Monitor using SIP Options for the corresponding SIP Server is incorrectly configured.

The SBC handles the calls as follows:

- Incoming call is allowed. Since the server is down, the Signaling Group's Server Status is set as **Down**. Instead of attempting to send calls to the server, the call proceeds to the next Signaling Group (if configured based on the routing policy).
- Based on routing configuration, incoming calls are made without any post dial delay to the configured SIP or ISDN trunk.

Scenario 2: Registration to server is not configured; Monitor using SIP Options for SIP Server fail

The SBC handles the calls as follows:

- The incoming call is allowed. Instead of attempting to send calls to the Server (since the Server is down). the call will proceed to the next Signaling Group (if configured based on the routing policy).
- Based on routing configuration, incoming calls are made without any post dial delay to the configured SIP or ISDN trunk.

Scenario 3: Registration to server succeeds; Monitor using SIP options for SIP Server fail

When a very high registration expires (**Global Time to Live TTL**) interval is configured (configuration available through the [Contact Registrant Table](#)), ensure it is sent more frequently. For example, set the registration interval less than the time set for the Registration to expire.

The SBC handles the calls as follows:

- The incoming call is allowed. If Registration is still valid, the Server and corresponding Signaling Group's Server status is **not** marked as **Down**. Calls are attempted to the Server. If no response, the call proceeds to the next Signaling Group if configured based on routing policy.
 - **Case 1.** The Far end supports Monitor using SIP options and the Registration interval is configured to be higher than the option interval. The Options response fails, which indicates the Server is down (there may be a post dial delay as the Signaling Group's Server status is not set to **Down**).
 - **Case 2.** The far end does not support Monitor using SIP options and the Registration Interval (Global Time to Live or Failed Registration Retry Timer) is configured to be higher than the registration interval. In this case, the Monitor using SIP options response failure is not an indication that the Server is down. Since the Signaling Group's Server Status is not set to Down, the call proceeds normally.

Scenario 4: Registration to server fails; Monitor using SIP Options for SIP Server are successful

The SBC handles the calls as follows:

- Incoming calls are allowed. If Registration (through [Contact Registrant Table](#)) fails, but Option responses (through the [SIP Server Table](#)) are successful, the Signaling Group's Server status is not set to **Down**. Possibilities for the registration failure could be that the Registration was misconfigured and the Server may still be up (as evident from successful Options responses).

Scenario 5: Both Registration and Monitor using SIP Options for SIP Server fail

The SBC handles the calls as follows:

- Incoming calls are allowed. Since the Server is down, the Signaling Group's Server status is set to **Down**.
- Calls proceed to the next Signaling Group if configured based on [routing policy](#).
- Based on routing configuration, incoming calls are made without any post dial delay to the configured SIP or ISDN trunk.

Scenario 6: Receiving a 3xx/4xx/5xx/6xx response to Register with a Retry after header

The SBC handles the calls as follows:

- The corresponding session fails.
- The retry-handling is relevant only for Registrations; it will still allow registrations to be retried to the corresponding server but the Invite requests will not route if all the corresponding sessions fail and the Signaling Group's Server status is set to **Down**.

Scenario 7: Receiving final 5xx/6xx (without Retry after header) failure responses to Register

The SBC handles the calls as follows:

- The corresponding session fails.
- The retry-handling is relevant only for Registrations; it will still allow registrations to be retried to the corresponding server but the Invite requests will not route if all the corresponding sessions fail and the Signaling Group's Server status is set to **Down**.

Scenario 8: Receiving 3xx/4xx (without Retry after header) response to Register

The SBC handles the calls as follows:

- The corresponding session is up.
- Invite requests are routed to the corresponding Signaling Group (the status is set to **Up**).

Forward Register after Local Processing Mode - How Incoming Calls are handled in the SBC

Below are the components of a configuration that includes Forward Register after Local Processing Mode (configured through the [SIP Signaling Group](#)).

- Endpoint is registered with the SBC.
- SBC has a Signaling Group configured to send requests to a Server.
- Signaling Group is configured with **Fwd Register After Local Processing** as the SIP Mode.
- A signaling group is associated with both call origination and termination device.

Scenario 1: Registration fails; Fwd Register Option not configured

The SBC handles the calls as follows:

- Even though the Server is down (and single Signaling group is used), the Signaling Group's Server status is **not** set to **Down**.
- The incoming call will not attempt to reach the Server and will follow the [routing policy](#) to try the next SIP/ISDN/CAS trunk using the associated Signaling Group.

Scenario 2: Fwd Register Option fails; Registration not configured

The SBC handles the calls as follows:

- Even though the Server is down (and single Signaling group is used), the Signaling Group's Server status is **not** set to **Down**.
- The incoming call will not attempt to reach the Server and will follow the [routing policy](#) to try the next SIP/ISDN/CAS trunk using the associated Signaling Group.

Scenario 3: Registration fails; Fwd Register Option succeeds

The SBC handles the calls as follows:

- Normal call flow for incoming call. In case the Server is down, there may be post dial delay in attempting to call the Server.

Scenario 4: Registration succeeds; Fwd Register Option fails

The SBC handles the calls as follows:

- Normal call flow for incoming call. In case the Server is down, there may be post dial delay in attempting to call the Server.

Troubleshoot Configuration for connection to a Server

To troubleshoot a server failure due to configuration, verify the current Registration and SIP Server Options configured on the SBC.

Verify Signaling Group

In the Signaling Group screen, the selections available in the **SIP Server Table** drop down list are derived from the entries configured in the [SIP Server Table](#). This signaling group is used in the SIP Server Table and corresponds to the Server being contacted.

Verify the Signaling Group Server Table selected for this specific signaling group as follows:

1. In the WebUI, click the **Settings** tab.
2. In the left navigation pane, go to **Signaling Groups**.
3. Click on the desired SIP Signaling Group.
4. In the **SIP Channels and Routing** portion, verify the **SIP Server Table** selected.
5. Change or update as necessary. See [Creating and Modifying SIP Signaling Groups](#).

Figure 1: Verify Signaling Group

The screenshot displays the Sonus WebUI interface for configuring a Signaling Group. The left navigation pane shows 'Signaling Groups' selected. The main content area shows the 'SIP Channels and Routing' configuration for a selected signaling group. The 'SIP Server Table' dropdown menu is highlighted with a red box, showing 'SIP Trunk Calvin ux2000' as the selected option. Other configuration options include 'Action Set Table', 'Call Routing Table', 'No. of Channels', 'SIP Profile', 'SIP Mode', 'Agent Type', 'Media Information', and 'Tone Table'.

Verify Registration/Monitor for SIP Options (for SIP Server)

Through the SIP Server Table, you choose the Contact Registrant table that will be used by a Signaling Group to register one or more contacts to a registrar. Contact Registrant Tables are used to manage contacts that are registered to a SIP server. When Monitor using SIP options is selected, an Options message is sent to the server with more detailed information, such as how often the SBC queries the server with an Options message to determine the server's availability.

When a Server is unreachable and the problem is registration failing or SIP options not being successful, you can verify the configuration as follows:

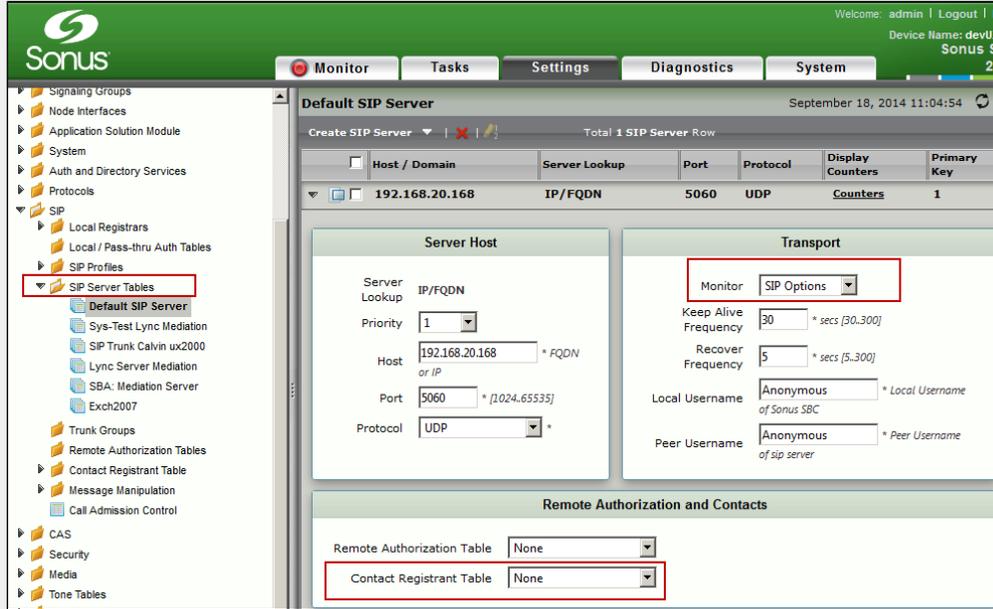
1. In the WebUI, click the **Settings** tab.
2. In the left navigation pane, go to **SIP > SIP Server Table**.
3. Select the desired **SIP Server table** that corresponds to the Server being contacted.
4. Click the **Contact Registrant Table** drop down box. All available registration tables listed in the Contact Registrant Tables are displayed.



When a challenge (401/407) is issued by the server, the table selected from the **Remote Authorization** drop down list is used by a SIP Server Table.

5. From the **Monitor** drop down list, select **SIP Options**.
6. Verify the configuration and adjust, if necessary. See [Creating and Modifying Entries in SIP Server Tables](#).

Figure 2: Verify Registration/Monitor for SIP Options



Verify Configuration for Contact Registrant

The configuration configured in the Contact Registrant Tables is used to manage contacts that are registered to a SIP server. If registration to a Server is successful, but the Monitor using SIP Options fail, verify the following configuration.

1. In the WebUI, click the **Settings** tab.
2. In the left navigation pane, go to **SIP > Contact Registrant Table**.
3. Click the table that corresponds to the Server.
4. Verify the configuration (i.e. **Global Time to Live**) and adjust, if necessary. See [Creating and Modifying Entries in Contact Registrant Tables](#).

Figure 3: Verify Configuration for Contact Registrant

The screenshot displays the Sonus SBI WebUI interface. The top navigation bar includes 'Monitor', 'Tasks', 'Settings', 'Diagnostics', and 'System'. The left sidebar shows a tree view of configuration categories, with 'SIP' expanded and 'Contact Registrant Table' selected, highlighting the 'SIP User' entry. The main content area is titled 'SIP User' and shows configuration details for a single entry, 'SIP:user'. The configuration includes a dropdown for 'Type of Address of Record' set to 'Local', an 'Address of Record URI' field containing 'SIP:user', and two timer fields: 'Global Time to Live (TTL)' set to 3600 and 'Failed Registration Retry Timer' set to 120. Below the configuration fields is a table titled 'SIP Contacts' with columns for 'Contact URI Username', 'TTL (secs)', and 'Priority (Q)'. The table is currently empty, displaying the message '-- Table is empty --'.

Verify Forward Register Option

If registration to a Server is successful, but the Forward Register option fails, verify the following configuration:

1. In the WebUI, click the **Settings** tab.
2. In the left navigation pane, go to **Signaling Groups**.
3. Click on the desired Signaling group that corresponds to the Server.
4. From the **SIP Mode** drop down list, select **Fwd Reg. after Local Processing**.

Figure 4: Verify Forward Register Option

The screenshot displays the Sonus SBC configuration interface. At the top, there is a navigation bar with tabs for Monitor, Tasks, Settings, Diagnostics, and System. The main content area is titled "Signaling Group Table" and shows a table of signaling groups. The selected group is a SIP Trunk with the following configuration:

Type	Description	Admin State	Service Status	Display	Primary Key
SIP	Lync Mediation server	Up	Up	Counters Channels	1
SIP	SIP Trunk	Up	Up	Counters Channels	2

The configuration details for the selected SIP Trunk are as follows:

- No. of Channels: 60
- SIP Profile: Default SIP Profile
- SIP Mode: Fwd Reg. After Local Processing (highlighted with a red box)
- Registrar: SIP Reg
- Registrar Min. TTL: 600
- Outbound Registrant TTL: 600
- SIP Server Table: SIP Trunk Calvin ux2000
- Load Balancing: Round Robin
- Channel Hunting: Most Idle
- Notify Lync CAC Profile: Disable

Additional settings on the right include Audio/Fax Stream DSP Mode (Enabled), Video/Application Stream Proxy Mode (Disabled), Media List ID (Lync media list w /Crypto), Play Ringback (Auto), Tone Table (Default Tone Table), Early 183 (Enable), and Music on Hold (Disabled). A Mapping Tables section is also visible at the bottom.

