SIP Server HA Deployment Guide

BC Configuration Options

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BC Configuration Options

This topic describes configuration options that are used in the deployment of SIP Business Continuity. For high-availability-specific options, see the HA Configuration Options topic for details.

For a complete list of configuration options, refer to the *SIP Server Deployment Guide*.

Application-Level Options

connect-nailedup-on-login

Setting: [TServer] section, Application and DN levels
Default Value: An empty string
Valid Values: Routing Point number, `gcti::park`
Changes Take Effect: At the next agent login session
Related Feature: Nailed-Up Connections

Specifies SIP Server actions when receiving a TAgentLogin request from a DN with the configured nailed-up connection, as follows:

- When this option is set to a DN of type Routing Point, SIP Server immediately establishes a nailed-up connection between an agent's endpoint and the specified Routing Point. After processing the TRouteCall request to the `gcti:park` device, SIP Server parks the agent on `gcti::park`, establishing the persistent SIP connection with the agent's endpoint.
- When this option is set to `gcti::park`, SIP Server parks the agent on the `gcti::park` device directly, establishing the persistent SIP connection with the agent's endpoint.
- When the value for this option is not specified (the default), SIP Server does not take any action.

At a DN level, this option must be set on agent Extension DN, or, if this DN is located behind the softswitch on the respective softswitch DN.

disconnect-nailedup-timeout

Setting: [TServer] section, Application and DN levels
Default Value: `0`
Valid Values: Any positive integer
Changes Take Effect: At the next nailed-up connection
Related Feature: Nailed-Up Connections

Specifies whether SIP Server terminates an agent's nailed-up connection because of the agent's inactivity. When set to a non-zero value, SIP Server waits this time interval, in seconds, before terminating the agent's nailed-up connection. When set to `0` (the default), SIP Server does not terminate the agent connection.

dr-forward

Setting: [TServer] section, Application level
Default Value: `off`
Valid Values:

- `off`—DR forwarding to the peer switch is disabled. SIP Server delivers calls to a DN on the local switch.
- `no-agent`—SIP Server delivers a call to a DN on the local switch when there is an agent logged in on this DN; if there is no agent logged in, SIP Server forwards the call to the peer switch. This setting applies only to DNs on "dual-registered" SIP endpoints that simultaneously register to both peers, or for DNs without any registered endpoints, such as for "remote agents".
- `oos`—SIP Server delivers a call to a DN on the local switch when the DN is in service; if the DN is out of service, SIP Server forwards the call to the peer switch. This setting applies only to DNs on "single-registered" SIP endpoints that register only to the preferred peer unless there is an error, in which case they register to the alternate peer.

Changes Take Effect: Immediately

Defines a system-wide mode of forwarding inbound and internal calls when SIP Server is operating in Business Continuity mode. This option can also be set at the DN level, in which case the setting overrides that set at the Application level.

Notes:

- The registration timing of endpoints must be carefully considered when using the `oos` setting. For maximum responsiveness in a disaster scenario, a short registration interval must be used so the phone can quickly detect when a peer is unavailable. The deployment should be properly planned to account for the corresponding load of REGISTER messages.
- For both the `no-agent` and `oos` settings, SIP Server only forwards calls targeting an Extension DN, and it will only forward each call a single time; if the other peer is unable to deliver the call, an error is generated.
- With the `oos` setting, if a desktop is unable to connect to the site where a SIP phone is registered, it might result in a phone registering a DN on one peer while the agent desktop connects to the other peer. Calls would be delivered to the phone, but the agent desktop would be unaware of these calls.

dr-peer-location

Setting: [TServer] section, Application level
Default Value: `NULL`
Valid Values: A valid name of the DR peer Switch
Changes Take Effect: on the next target detection

Specifies the location of the other SIP Server in the DR pair. If set to `NULL` (the default), SIP Server is unable to support the Dial Plan feature.

dr-peer-trunk

Setting: [TServer] section, Application level
Default Value: `NULL`
Valid Values: A valid name of a `Trunk` DN that points to the DR peer site
Changes Take Effect: Immediately

Specifies that this SIP Server is a part of a DR pair, and identifies the Trunk DN that points to the other SIP Server in the DR pair. If set to NULL (the default), SIP Server operates in the traditional single mode.

graceful-shutdown-sip-timeout

Setting: [TServer] section, Application level
Default Value: 4
Valid Values: 0–32
Changes Take Effect: Immediately

Specifies the timeout, in seconds, during which SIP Server re-transmits the BYE requests that were not confirmed with 200 OK responses. The timeout starts as soon as the last call is ended. If set to 0 (zero), no BYE requests are re-transmitted. The timeout applies only when SIP Server processes the graceful shutdown.

shutdown-sip-reject-code

Setting: [TServer] section, Application level
Default Value: 603
Valid Values: 300–603
Changes Take Effect: Immediately

Specifies the error response used for rejecting new INVITE messages received by the system that is in shutdown mode. If set to 300, 301, or 302, SIP Server first checks to see if **dr-peer-trunk** is configured, and if so, sends the contact of that Trunk DN in the 302 response.

DN-Level Options

dr-oosp-transfer-enabled

Setting: [TServer] section, DN level
Default Value: true
Valid Values: true, false
Changes Take Effect: For the next call

In Business Continuity deployments, for special circumstances where an inbound call remains on the same site where it arrives and SIP Server puts itself Out of Signaling Path. This option is supported only for Trunk DNs pointing to external destinations. It must not be configured on the trunks between SIP Servers.

If set to false on the Trunk DN from where an inbound INVITE is received, SIP Server stays in the signaling path if the call, after being processed on the Routing Point DN, is sent to the local Extension DN where the DR call forwarding procedure is applied to deliver the call to the corresponding DN on the peer SIP Server. If set to true (the default), SIP Server puts itself Out Of Signaling Path.

hg-preferred-site

Setting: [TServer] section, DN level

Default Value: No default value
Valid Values: Any string value
Changes Take Effect: For the next call
Related Feature: Hunt Groups in Business Continuity

Specifies the name of the SIP Server DR Peer application corresponding to the preferred Hunt Group site. If not set or set to an invalid application name, the preferred Hunt Group site cannot be determined, and inbound Hunt Group calls are processed at the site where they are received.

sca-preferred-site

Setting: [TServer] section, DN level
Default Value: No default value
Valid Values: Any string value
Changes Take Effect: For the next call
Related Feature: Shared Call Appearance in Business Continuity

Specifies the name of the SIP Server DR Peer application corresponding to the preferred SCA site. If not set or set to an invalid application name, the preferred SCA site cannot be determined, and inbound SCA calls are processed at the site where they are received. The option can be configured only for the Primary shared line DN, where the **shared-line** option is set to `true`.