SIP Server Deployment Guide

New or Updated Configuration Options

4/14/2025

Contents

New or Updated Configuration Options

The following configuration options have been either newly added or updated after the SIP Server Deployment Guide was last published.

> **Important**
> The SIP Server Deployment Guide PDF is no longer updated with information pertaining to recent changes. Information on recently released features, other modifications, and new or updated configuration options are available in this supplement. If you are looking for specific information, please refer to both the documents, first the SIP Server Deployment Guide and then this Supplement to SIP Server Deployment Guide.

sip-schedule-record-on-hold

Setting: **[TServer]** section, Application level or DN level
Default Value: `true`
Valid Values: `true`, `false`
Changes Take Effect: On the next call

Introduced in version 8.1.104.73, this option is used to avoid scheduling a recording for the call hold party if there is any other record-enabled party on the call. If no other record-enabled party is on the call, the recording will be scheduled once the call hold party retrieves the call. Setting the option to `false` will prevent automatic retrieving of the party on hold when creating a recording leg. The default value is true and retains the existing behaviour.

enable-rp-to-rp-dial-plan

Setting: **[TServer]** section, Application level
Default Value: `false`
Valid Values: `true`, `false`
Changes Take Effect: On the next call

Introduced in version 8.1.104.69, this option is used to apply dial-plan rules to a call moved from one Routing Point (RP) or external RP to another RP. If set to `true`, SIP Server applies dial-plan rules to a call moved from one routing point to another. If set to `false`, dial-plan rules are not applied and the call is just sent to another routing point. For multi-site ISCC calls, additionally, the **enable-iscc-dial-plan** option must be set to `true`. For regular calls, additionally, the **rp-use-dial-plan** option must be set to `full` or `partial`.

disable-tconf-agent-logins-loading

Setting: **[TServer]** section, Application level
Default Value: `false`
Valid Values: `true`, `false`
Changes Take Effect: After SIP server restart

Introduced in version 8.1.104.67, this option controls processing of agent login objects by the **tconf** library. If set to `true`, SIP Server, while deployed in SIP Cluster mode, will eliminate processing of agent login objects by the **tconf** library. This allows to speed up initialization of the SIP Cluster node even if the number of configured agent logins exceeds 100,000.

3pcc-requires-agent-session

Setting: **[TServer]** section, Application level
Default Value: `false`
Valid Values: `true`, `false`
Changes Take Effect: Immediately

Introduced in version 8.1.104.66, use this option to enable enhanced restriction of 3PCC requests. When set to `true`, SIP Server only accepts 3PCC requests from a client associated with an agent login session. Requests from other clients are rejected with `EventError error code: 118 (Requested service unavailable) 'Access restricted'`. When set to `false`, SIP Server does not restrict 3PCC requests based on an agent login association. This option is only applicable for requests with **AttributeThisDN** of types **Extension** and **ACD Position**.

The enhanced restriction is applicable only to the following requests:

- RequestAgentLogout
- RequestAgentReady
- RequestAgentNotReady
- RequestSetDNDOn
- RequestSetDNDOff
- RequestMakeCall
- RequestAnswerCall
- RequestHoldCall
- RequestRetrieveCall
- RequestInitiateConference
- RequestCompleteConference
- RequestDeleteFromConference
- RequestInitiateTransfer

- RequestMuteTransfer
- RequestSingleStepTransfer
- RequestCompleteTransfer
- RequestAlternateCall
- RequestReconnectCall
- RequestCallForwardSet
- RequestCallForwardCancel
- RequestReleaseCall
- RequestSendDTMF
- RequestRedirectCall
- RequestListenDisconnect
- RequestListenReconnect
- RequestClearCall
- RequestSingleStepConference
- RequestMonitorNextCall
- RequestCancelMonitoring
- RequestSetMuteOn
- RequestSetMuteOff

dr-back-in-service-on-invite

Setting: **[TServer]** section, Application level
Default Value: `true`
Valid Values: `true`, `false`
Changes Take Effect: Immediately

Introduced in version 8.1.104.61, use this option to specify if a non-emergency DN without an active registration, in DR mode, is set to `Back-In-Service` on the INVITE initiating a first-party call-control (1pcc) call. If the option is set to `false`, a non-emergency DN without an active registration is not set to `Back-In-Service` on the INVITE initiating a first-party call-control. Previously, setting such devices to `Back-In-Service` allowed agents to login and set their DN as ready without having their device ready for voice calls resulting in `EventError (Ivalid Called Dn)` responses to any attempt to route calls to such DNs.

partyadded-def-callstate-conf

Setting: **[TServer]** section, Application level

Default Value: `false`
Valid Values: `true`, `false`
Changes Take Effect: On the next call

Introduced in version 8.1.104.40, use this option to ensure that SIP Server distributes the **EventPartyAdded** event with **AttributeCallState** set to 2 in a multi-site single step conference scenario. If the option is not found or set to `false`, SIP Server distributes the **EventPartyAdded** event with **AttributeCallState** set to `0` instead of 2.

reuse-tls-conn

Setting: **[TServer]** section, Application level and DN level
Default Value: `true`
Valid Values: `true`, `false`
Changes Take Effect: On the next call

Introduced in version 8.1.104.27, use this option is used to specify whether SIP Server reuses the existing TLS transport for sending SIP requests. If set to `false`, SIP Server opens a new TLS connection to the SIP request destination. If set to `true`, SIP Server reuses the existing TLS transport for sending SIP requests.

Note: The option is supported at the DN level too, starting with version 8.1.104.47.