SIP Cluster Solution Guide

Configuring SIP Servers

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Configuring SIP Servers

You must configure SIP Server applications for the following purposes:

- Cluster nodes
- Virtual queues
- Historical reporting

Configuring SIP Servers for SIP Cluster

1. Deploy SIP Servers as an HA pair, Hot Standby redundancy mode, by following the standard procedure.
    - Suggested application names: **SIPS_<datacenter>_1**, **SIPS_<datacenter>_1_B**.
2. On the **Switches** tab, add the SIP Cluster Switch object to each SIP Server application.
3. On the **Connections** tab, add the following connections:
    - **confserv_proxy_<datacenter>**—Set to the following parameters:
        - Connection Protocol: addp
        - Trace Mode: Trace On Both Sides
        - Local Timeout: 60
        - Remote Timeout: 90
    - **MessageServer_<datacenter>**—Set to the following parameters:
        - Connection Protocol: addp
        - Trace Mode: Trace On Both Sides
        - Local Timeout: 7
        - Remote Timeout: 11
4. On the **Server Info** tab, configure the following ports for each SIP Server application:

ID	Listening Port	Connection Protocol
default	Any available port number	
TCport	Any available port number	TController
IPport	Any available port number	IProxy
SmartProxy	Any available port number	SmartProxy
Note: Changes in port numbers take effect after SIP Server restart.		

5. On the **Options** tab in the **[TServer]** section, configure the following mandatory options for each

SIP Server application to be run in cluster mode:

Name	Cluster Value	Description
server-role	5	For SIP Server to run in cluster mode.
sip-link-type	3	For SIP Server to run in multi-threaded mode.
geo-location	<string>	A string identifying the data center to which this SIP Server instance belongs. It is used by SIP Proxy to select SIP Server in same data center as SIP Proxy. SIP Server uses it to select geo-location for 3PCC calls. All applications deployed in the same data center use the same value for this geo-location parameter.
sip-address	<SIP Server A-Record FQDN>	For SIP Server to build the Via and Contact headers in SIP messages.
sip-address-srv	<blank>	For SIP Server to build the Via and Contact headers in SIP messages.
sip-outbound-proxy	true	
sip-enable-gdns	true	To resolve SRV contacts.
sip-enable-rfc3263	true	To resolve priority and weight in SRV tables.
dial-plan	<dial plan DN name>_<short data center name>	The name of the Dial Plan DN (the VoIP Service DN with **service-type** set to `feature-server`).
find-trunk-by-location	true	To enable selection of the trunk and softswitch by geo-location. This is required to keep SIP signalling on the correct data center.
enable-strict-location-match	all	To enable strict matching of MSML resources, which is required for the SIP Cluster. SIP Server in a particular geo-location must only use MCP resources in the same geo-location.
sip-enable-x-genesys-route	true	To enable a private X-Genesys-Route header in SIP messages towards SIP Proxy. It's exclusively used by (and not propagated beyond) the SIP Proxy.
sip-port	<SIP port>	
http-port	<HTTP port>	
management-port	<management port>	

The sample configuration:

```
[agent-reservation]
request-collection-time=300 msec

[backup-sync]
addp-remote-timeout=11
addp-timeout=7
addp-trace=full
protocol=addp

[call-cleanup]
cleanup-idle-tout=60 min
notify-idle-tout=5 min
periodic-check-tout=10 min
```

```
[extrouter]
cast-type=route direct-notoken direct-callid reroute direct-uui direct-ani dnis-pool direct-
digits pullback route-uui direct-network-callid

[Log]
all=/mnt/log/SIPS_<datacenter>_1/SIPS_<datacenter>_1
buffering=false
expire=15
segment=100 MB
spool=/mnt/log/SIPS_<datacenter>_1
standard=network
time-format=iso8601
verbose=all
x-gsipstack-trace-level=3
x-server-trace-level=3

[log-filter]
default-filter-type=hide

[TServer]
acw-persistent-reasons=false
after-routing-timeout=18
agent-emu-login-on-call=true
agent-logout-on-unreg=true
agent-no-answer-action=notready
agent-no-answer-timeout=12
call-observer-with-hold=true
consult-user-data=inherited
clamp-dtmf-allowed=true
default-dn=default_rp
default-route-point=reject=404
default-route-point-order=after-dial-plan
default-music=music/on_hold_saas
dial-plan=DialPlan
divert-on-ringing=false
emulated-login-state=not-ready
extn-no-answer-timeout=12
greeting-call-type-filter=
greeting-delay-events=false
greeting-notification=
http-port=9096
init-dnis-by-ruri=true
logout-on-out-of-service=true
management-port=5002
merged-user-data=merged-over-main
monitor-consult-calls=true
msml-record-metadata-support=true
msml-record-support=true
msml-support=true
music-in-conference-file=qtmf://music/silence
override-to-on-divert=true
posn-no-answer-timeout=12
record-consult-calls=true
record-moh=false
recording-failure-alarm-timeout=900
recording-filename=$UUID$_$DATE$_$TIME$
registrar-default-timeout=140
ring-tone=qtmf://music/ring_back
rq-expire-tmout=0
rq-expire-tout=0
server-id=
set-notready-on-busy=true
```

```
shutdown-sip-reject-code=503
sip-address=<A-record FQDN>
sip-address-srv=
sip-call-retain-timeout=1
sip-dtmf-send-rtp=true
sip-enable-100rel=false
sip-enable-call-info=true
sip-enable-ivr-metadata=true
sip-enable-moh=true
sip-enable-rfc3263=true
sip-invite-treatment-timeout=15
sip-port=5060
sip-preserve-contact=true
sip-treatments-continuous=true
timeguard-reduction=1000
unknown-gateway-reject-code=503
userdata-map-trans-prefix=X-Genesys-
```

Configuring SIP Servers for Virtual Queues

Virtual Queue (VQ) SIP Servers are used primarily to manage Virtual Queues. This eliminates the need to synchronize Virtual Queue states across SIP Cluster Nodes.

1. Deploy SIP Servers as an HA pair (one HA pair per data center), Hot Standby redundancy mode, by following the standard procedure.
    - Suggested application names: **SIPS_VQ_<datacenter>**, **SIPS_VQ_<datacenter>_B**.
2. On the **Connections** tab, add the following connections:
    - **confserv_proxy_<datacenter>**—Set to the following parameters:
        - Connection Protocol: addp
        - Trace Mode: Trace On Both Sides
        - Local Timeout: 60
        - Remote Timeout: 90
    - **MessageServer_<datacenter>**—Set to the following parameters:
        - Connection Protocol: addp
        - Trace Mode: Trace On Both Sides
        - Local Timeout: 7
        - Remote Timeout: 11
3. On the **Switches** tab, add the **VQ-switch** object to each VQ SIP Server application. All VQ SIP Servers must be associated with the same **VQ-switch**.
4. On the **Server Info** tab, must be only the default port.
5. VQ SIP Server sample configuration:

```
[agent-reservation]
request-collection-time=300 msec

[backup-sync]
addp-remote-timeout=11
addp-timeout=7
addp-trace=full
protocol=addp

[call-cleanup]
cleanup-idle-tout=60 min
notify-idle-tout=5 min
periodic-check-tout=10 min

[extrouter]
cast-type=route direct-notoken direct-callid reroute direct-uui direct-ani dnis-pool direct-
digits pullback route-uui direct-network-callid

[Log]
all=/mnt/log/SIPS_VQ_<datacenter>/SIPS_VQ_<datacenter>
buffering=false
expire=15
segment=100 MB
spool=/mnt/log/SIPS_VQ_<datacenter>
standard=network
time-format=iso8601
verbose=all
x-gsipstack-trace-level=3
x-server-trace-level=3

[TServer]
acw-persistent-reasons=false
after-routing-timeout=18
agent-emu-login-on-call=true
agent-logout-on-unreg=true
agent-no-answer-action=notready
agent-no-answer-timeout=12
call-observer-with-hold=true
consult-user-data=inherited
clamp-dtmf-allowed=true
default-dn=default_rp
default-route-point=reject=404
default-route-point-order=after-dial-plan
default-music=music/on_hold_saas
dial-plan=DialPlan
divert-on-ringing=false
emulated-login-state=not-ready
extn-no-answer-timeout=12
greeting-call-type-filter=
greeting-delay-events=false
greeting-notification=
http-port=9096
init-dnis-by-ruri=true
logout-on-out-of-service=true
management-port=5002
merged-user-data=merged-over-main
monitor-consult-calls=true
msml-record-metadata-support=true
msml-record-support=true
msml-support=true
music-in-conference-file=qtmf://music/silence
override-to-on-divert=true
posn-no-answer-timeout=12
```

```
record-consult-calls=true
record-moh=false
recording-failure-alarm-timeout=900
recording-filename=$UUID$_$DATE$_$TIME$
registrar-default-timeout=140
ring-tone=qtmf://music/ring_back
rq-expire-tmout=0
rq-expire-tout=0
server-id=
set-notready-on-busy=true
shutdown-sip-reject-code=503
sip-address=<A-record FQDN>
sip-address-srv=
sip-call-retain-timeout=1
sip-dtmf-send-rtp=true
sip-enable-100rel=false
sip-enable-call-info=true
sip-enable-moh=true
sip-enable-rfc3263=true
sip-invite-treatment-timeout=15
sip-port=5060
sip-preserve-contact=true
sip-treatments-continuous=true
timeguard-reduction=1000
userdata-map-trans-prefix=X-Genesys-
```

You will add VQ SIP Servers to the following applications:

- Stat Servers in each data center
- A dedicated HA pair of Interaction Concentrator instances to monitor an HA pair of VQ SIP Servers in the same data center
- Each URS and ORS located in the same data center

Configuring SIP Servers for Historical Reporting

When operating in cluster mode, Interaction Concentrator server (ICON) must connect to two ports of SIP Server: T-Controller (TCport) and Interaction Proxy (IPport). For this purpose, a dummy SIP Server application must be created. When configuring ICON for Voice details, add a connection to the **IPport** of the actual SIP Server application, and add a connection to the **TCport** of the dummy SIP Server application. Each connection represents a session. Genesys Info Mart requires each session to be associated with a SIP Server application.

1. Deploy a SIP Server application, by following the standard procedure.
2. On the **Switches** tab, add the SIP Cluster Switch, the same Switch as in the actual SIP Server application.
3. On the **Server Info** tab, add the same listening ports as in the actual SIP Server application. The Server Info tab must not contain HA configuration.
4. On the **Connections** tab, don't add anything. It must be empty.

5. On the **Options** tab in the **[TServer]** section, don't make any changes.
6. On the **Start Info** tab, clear the Auto-Restart box to avoid SCS restarting the application.
7. On the **Annex** tab in the **[sml]** section, set **autostart**=`false` to avoid SCS restarting the application.

Important

The dummy SIP Server application must *not* be added to the **applications** option of the SIP Cluster DN.