Genesys Application Note

Oracle Enterprise SBC With Genesys SIP Server

Document version 1.4

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1 Summary

This application note details the supported features, and includes reference configuration examples.

The following Oracle Enterprise Session Border Controller versions were tested and supported:

- Oracle Enterprise Session Border Controller AP 1100/3900/4600/6300/6350/VME/3950/4900 version SCZ9.0.0
- Oracle Enterprise Session Border Controller AP 1100/3900/4600/6300/6350 version E-CZ8.1.0
- Oracle Enterprise Session Border Controller 3820 version E-CZ7.2.0 (7.2 and later)
- Acme Net-Net Hardware-based SBCs (38xx/4xxx/9xxx series) version 6.x (6.1.0 MR2 and later)

The supporting versions of Genesys components include SIP Server v8.1.1, SIP Feature Server v8.1.2, Media Server (v8.1.x and v8.5.x), and SIP Proxy v8.1.1.

As noted in section 2, most test calls/cases were successfully executed.

2 Feature Support

2.1 Feature Chart

Feature Name	
General Features Supported By SBC	**Supported**
Inbound Calls - Standard	Yes
Inbound Calls - Contact Center/Routed	Yes
Outbound Calls - Standard	Yes
Outbound Calls - Automated Dialer Campaign, CPD by Genesys	Yes
Remote Agent, not REGISTERed to SIP Server	Yes
Call Qualification & Parking	Yes
GVP - Advanced IVR (VXML, TTS, ASR, etc), Conferencing, & more	Yes
Technical Features	**Supported**
"Single Site"	Yes
"Multisite"	Yes
SIP Business Continuity	Yes
Transfer with re-INVITE	Yes
Transfer with 3xx	Yes
Transfer with REFER	Yes
Ad Hoc Conference	Yes
SIP Authentication	No
SIP Over TLS	Yes
SRTP	Yes
Service Monitoring	Yes
SIP Server High Availability - with Virtual IP Address	Yes
SIP Trunk/SBC/Gateway High Availability - with Virtual IP Address	Yes
SIP Trunk/SBC/Gateway High Availability – List of IP Addresses	N/T
SIP Server High Availability - DNS-based Redundancy with SIP Proxy	N/T
SIP Trunk/SBC/Gateway High Availability - DNS-based Redundancy	N/T
Audio Codec Support	Yes
Video Support	N/T
SBC-Specific Features	**Supported**
Inbound & Outbound Calls	Yes
SIP Agent 3PCC Control	Yes*
Remote Agent - Transfer with REFER (SIP Phone via SBC)	Yes
Transfer with REFER	Yes
Transfer with re-INVITE	Yes

* See the Known Issues section.

2.2 Test Cases Chart

Functional Test Cases		
#	**Scenario Description**	**Supported**
1	Inbound Call to Agent released by caller	Yes
2	Inbound Call to Agent released by agent	Yes
3	Inbound Calls rejected	Yes
4	Inbound Call abandoned	Yes
5	Inbound Call to Route Point with Treatment	Yes
6	Interruptible Treatment	Yes
7	IVR (Collect Digit) Treatment	Yes
8	Inbound Call routed by using 302 out of SIP Server signaling path	Yes
9	1PCC Outbound Call from SIP Endpoint to external destination	Yes
10	3PCC Outbound Call to external destination	Yes
11	1PCC Outbound Call Abandoned	Yes
12	Caller is put on hold and retrieved by using RFC 2543 method	Yes
13	T-Lib-Initiated Hold/Retrieve Call with MOH using RFC 3264 method	Yes
14	3PCC 2 Step Transfer to internal destination by using re-INVITE method	Yes
15	3PCC Alternate from consult call to main call	Yes
16	1PCC Unattended (Blind) transfer using REFER	Yes
17	1PCC Attended Transfer to external destination	Yes
18	3PCC Two Step Conference to external party	Yes
19	3PCC (same as 1PCC) Single-Step Transfer to another agent	Yes
20	3PCC Single Step Transfer to external destination using REFER	Yes
21	3PCC Single Step Transfer to internal busy destination using REFER	Yes
22	Early Media for Inbound Call to Route Point with Treatment	Yes
23	Early Media for Inbound Call with Early Media for Routed to Agent	Yes
24	Inbound call routed outbound (Remote Agent) using INVITE without SDP	Yes
25	Call Progress Detection	Yes
26	Out of Service detection; checking MGW live status	Yes
27	SIP Authentication for outbound calls	No
28	SIP Authentication for incoming calls	No
SBC-Specific Test Cases		
29	T-Lib-Initiated Answer/Hold/Retrieve Call for Remote SIP endpoint which supports the BroadSoft SIP Extension Event Package	Yes
30	3PCC Outbound Call from Remote SIP endpoint to external destination	Yes
31	3PCC Two Step Transfer from Remote SIP endpoint to internal destination	Yes
32	1PCC Attended Transfer from Remote SIP endpoint to external destination	Yes

2.3 General Features

SIP Trunk or Gateway - Feature Compatibility	Description	Supported	Test Cases
Inbound Calls - Standard	Direct calls to a phone/user with a DID #	Yes	1, 2, 3, 4, 12
Inbound Calls - Contact Center / Routed	Contact Center calls; may be queued or played some announcements before being routed to an agent	Yes	5, 6, 7, 13, 22, 23
Outbound Calls - Standard	Manually Dialed, or Forwarded to external destination	Yes	9, 10, 11
Outbound Calls - Automated Dialer Campaign, CPD by Genesys	Automated dialing by Genesys OCS or similar application Call Progress Detection by Genesys Media Server*	Yes	25
Remote Agent, not REGISTERed to SIP Server	Typically using a PSTN phone behind the gateway or SIP Trunk	Yes	24
Call Qualification & Parking	Simple IVR controlled by a routing strategy, and queuing of calls with announcements or music	Yes	5, 6, 7, 22, 23
GVP – Advanced IVR (VXML, TTS, ASR, etc), Conferencing, & more	Same SIP signaling as qualification & parking	Yes	6, 7
Call Recording	No meaningful impact to SIP signaling		No dedicated test cases

* CPD may also be performed by the gateway if it returns results in a format compatible with Genesys SIP. Please note such capabilities if they are available.

2.4 Technical Features

Technical Compatibility – Architecture & SIP Protocol	Description	Supported	<u>Test Cases</u>
"Single Site"	One instance of Genesys SIP Server	Yes	All Test cases apply
"Multisite"	Two or more instances of Genesys SIP Server, behind a single Trunk and/or SBC	Yes	No "dedicated" test cases
SIP Business Continuity	The SIP Business Continuity Architecture across two active data centers	[not tested – requires supplemental testing]	Not covered by standard test plan
Transfer with re-INVITE	Transfer method reflects the signaling sent to the SIP Trunk or gateway	Yes	14,15
Transfer with 3xx	Redirect prior to call connection	Yes	8
Transfer with REFER	Transfer method reflects the signaling sent to the SIP Trunk or gateway	Yes	16,17,19,20,21
Ad Hoc Conference	Conference controlled on Genesys SIP Server & Media Server	Yes	18
SIP Authentication		No	27, 28
SIP Over TLS	Please refer to the SIP Server Deployment Guide	Yes	No dedicated test cases
SRTP		Yes	No dedicated test cases
Service Monitoring	Monitoring with OPTIONS messages	Yes	26
SIP Server High Availability - with Virtual IP Address	Effectively transparent to external devices	Yes	No dedicated test cases
SIP Trunk/SBC/Gateway High Availability - with Virtual IP Address	Effectively transparent to external devices	Yes	No dedicated test cases
SIP Trunk/SBC/Gateway High Availability – List of IP Addresses	Support for a highly available SBC or SIP Trunk with either multiple active nodes or primary/backup; SIP Server is configured with the IP address of each node (typically using the backup contact setting on SIP Server)	[not tested – requires supplemental testing]	Not covered by standard test plan
SIP Server High Availability - DNS-based Redundancy with SIP Proxy	Architectures with SIP Proxy used to manage high availability	[not tested – requires supplemental testing]	Not covered by standard test plan
SIP Trunk/SBC/Gateway High Availability - DNS-based Redundancy	Support for an SBC or SIP Trunk with DNS-based redundancy (the contact of the DN on SIP Server would be hostname/FQDN)	[not tested – requires supplemental testing]	Not covered by standard test plan
Audio Codec Support	The test plan does not include dedicated tests for each codec; codecs are supported by Media Server/GVP, and by the SIP endpoints	Yes	All test cases utilize the "default" codec

Video Support	The test plan does not include dedicated tests for video; video is supported by Media Server/GVP, and by the SIP endpoints		No dedicated test cases

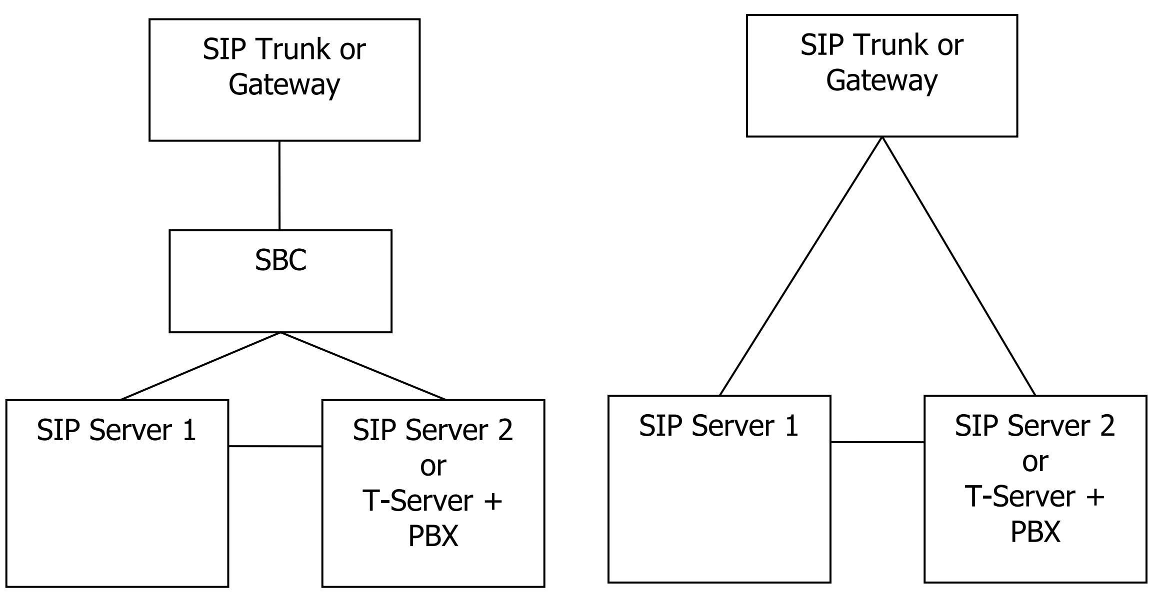
2.5 SBC-Specific Features

SBC Feature Compatibility for Agent REGISTERed to SIP Server through SBC	Supported	Test Cases
Inbound & Outbound Calls	Yes	29,30
SIP Agent 3PCC Control	Yes	29
Remote Agent - Transfer with REFER (SIP Phone via SBC)	Yes	30
Transfer with REFER	Yes	30
Transfer with reINVITE	Yes	31

See section 6 for known limitations.

2.6 Feature Details

2.6.1 Multisite



Note:
This application note uses the term "multisite" to cover architectures with transfers with ISCC, which conform to either option on the left: a SIP Trunk/Gateway through a single SBC, or a SIP Trunk/Gateway connected directly.

Either REFER or reINVITE may be tested and supported.

Architectures with 2 or more SBCs are beyond the scope of this app note.

2.6.2 High Availability

This Application Note and the Test Plan provide coverage and support for High Availability accomplished with a "Virtual IP Address". This is also referred to as "IP Address Takeover" or a "Floating IP Address."

The general approach is that the "active" instance of a component utilizes this special IP address. It is typically transparent at the SIP signaling layer which instance is active. This method of high availability may be employed by Genesys SIP Server, an SBC, or by the components that provide the interface for a standard "SIP Trunk."

Other methods of high availability do exist. These methods require more advanced logic on the part of each SIP component to monitor multiple instances of another component, and select the appropriate instance. For example, SIP Server supports configuring a primary and backup IP address for a component (using the contact and contacts-backup options). This method is referred to as a "List of IP Addresses" in this Application Note. In another example, SIP Server does support using an FQDN to reach another component, and can utilize multiple DNS records to help choose the best component instance. This method is referred to as "DNS-based HA."

Both the "List of IP Addresses" and "DNS-based HA" methods are beyond the scope of this Application Note (and this limitation applies in both directions, from SIP Server towards an external component, and vice versa from an external component towards SIP Server).

3 Software and Hardware Versions Validated

The following Genesys components and Oracle Acme Packet Enterprise SBC were validated for reference configuration examples.

3.1 Genesys Components

Genesys Components		
Component	**Version**	**Notes**
SIP Server	8.1.1	Genesys SIP Server performs call switching and control. SIP Server communicates via SIP with SIP Endpoints.
Media Server	8.1.7	Used to handle media interactions such as call treatments (ring back, busy tones and music on hold); also used as MCU.
SIP Proxy	8.1.1	Optionally can be used for DNS-based HA deployment.
SIP Feature Server	8.1.2	Used as a SIP Voicemail Server.

3.2 Gateway/SBC

3rd Party Hardware Components		
Model	**Version**	**Notes**
Oracle Enterprise SBC AP4600	SCZ9.0.0	SCZ9.0.0 or later are supported
Oracle Enterprise SBC 4600	E-CZ8.1.0	E-CZ8.1.0 or later are supported
Oracle Enterprise SBC 3820	E-CZ7.2.0	E-CZ7.2.0 or later are supported

For a full listing of 3rd party hardware/software supported by Genesys, see the Genesys Supported Media Interface (SMI) Guide.

4 Features Configuration in Genesys Configuration Environment

This section describes how to configure features presented in the Feature Chart in Genesys configuration environment.

Features can be configured in the SIP Server Switch on a DN object of type Trunk representing the SIP Trunk pointed to the SBC, a DN of type Extension (or ACD Position) representing SIP Endpoint devices, and/or on an Agent Login object, and/or in a SIP Server Application.

Note: It is assumed that the reader has Genesys knowledge and is familiar with deploying a basic Genesys environment.

<table>
<tr><th colspan="2">Genesys SIP Configuration</th></tr>
<tr><th colspan="2">Features Supported By Gateway/SBC</th></tr>
<tr><th>General Features</th><th>Key Actions and Procedures</th></tr>
<tr><td>Inbound Calls – Standard

Test cases:
1, 2, 3, 4, 5, 6, 7, 12, 13, 22, 23</td><td>1. In the Genesys configuration environment, under Switch -> DNs, create a DN object of type Trunk. This object represents the SIP Trunk pointing to the SBC.
2. In the Trunk DN -> TServer section, configure:
contact=<the contact URI that SIP Server uses for communication with the SBC>
3. If needed, enable support of Early media for inbound calls. In the Trunk DN -> TServer section, configure:
sip-early-dialog-mode=1
4. If needed, specify the method of hold media SDP (RFC 3264 "inactive" SDP) to be used by SIP Server for third-party call control (3pcc) hold operations.
In the SIP Server Application -> TServer section, configure:
sip-hold-rfc3264=true

Note: By default, SIP Server uses "black hole" RFC 2543 method (c=0.0.0.0).</td></tr>
<tr><td>Inbound Calls - Contact Center/Routed</td><td>Same configuration as for Inbound Calls - Standard, above.</td></tr>
</table>

Outbound Calls - Standard Test cases: 9, 10, 11	1. In the Genesys configuration environment, under Switch -> DNs, create a DN object of type Trunk. This object represents the SIP Trunk pointing to the SBC. 2. In the Trunk DN -> TServer section, configure: **contact =<the contact URI that SIP Server uses for communication with the SBC>** 3. To activate required features described in this Table, configure options in the Trunk DN object as described in Inbound Calls - Standard, above. 4. Configure the SBC to support inbound/outbound calls to/from SIP Server. 5. Configure a phone to make basic calls (incoming, outgoing) with SIP Server. 6. If needed, specify the REFER method that SIP Server will use to make 3pcc outbound calls. In the DN object of type Extension -> TServer section, configure: **refer-enabled=true** 7. Start SIP Server. 8. After successful SIP registration, the phone is ready for making outgoing calls and receiving incoming calls. 9. Run your desktop client to make a test call.
Outbound Calls - Automated Dialer Campaign, CPD by Genesys Test case: 25	1. Enable call progress detection to be done by Media Server. In the Trunk Group DN (pointing to RM) -> TServer section, configure:**cpd-capability=mediaserver** 2. Instruct SIP Server to use the re-INVITE method for 3pcc calls with call flow 1:**refer-enabled=false** **make-call-rfc3725-flow=1** 3. Disable a ring tone for scenarios that might include CPD by specifying: **ring-tone-on-make-call=false** 4. Specify the Request-URI in the following format: **sip:msml@<RMHost>:<RMPort>;media-service=cpd;gvp-tenantid=[<tenant name>]** 5. Specify the Tenant name where SIP Server is deployed: **subscription-id=<Tenant where SIP Server is deployed>** 6. Specify the Resource Manager IP address and SIP port: **contact =sip:<RM_ip_address>:<RM_sip_port>**
Remote Agent, not REGISTERed to SIP Server Test cases: 24	No configuration is required
Call Qualification & Parking Test cases: 5, 6, 7, 22, 23	No configuration is required

GVP – Advanced IVR (VXML, TTS, ASR, etc), Conferencing, & more	Deploy Genesys Media Server with required capabilities. See the Genesys 8.1 SIP Server Deployment Guide for details.
Technical Features	**Key Actions and Procedures**
"Single Site"	Deploy one instance of SIP Server. See "Inbound Calls" and "Outbound Calls" features, above.
"Multisite"	Deploy two or more instances of Genesys SIP Server behind a single Trunk and/or SBC. See Multisite and the Genesys 8.1 SIP Server Deployment Guide for details.
SIP Business Continuity	Refer to the Genesys SIP Server High-Availability Deployment Guide.
Transfer with re-INVITE Test cases: 14, 15	Specify the re-INVITE method to be used for 3pcc Attended transfer. In the DN type Extension (transfer controller) -> TServer section, configure: **refer-enabled=false**
Transfer with 3xx Test case: 8	Force SIP Server to put itself in the Out Of Signaling Path (OOSP) after the Unattended transfer (Genesys Single-Step Transfer) or routing to the external destination has been completed. In the Trunk DN object (representing the SBC) -> TServer section, configure: **oosp-transfer-enabled=true**
Transfer with REFER Test cases: 16, 17, 19, 20, 21	Specify the REFER method to be used for 3pcc transfer operations. In the Trunk DN object (representing the SBC) -> TServer section, configure: **refer-enabled=true**
Ad Hoc Conference Test case: 18	Deploy Genesys Media Server with MCU capabilities. See the Genesys 8.1 SIP Server Deployment Guide for details.
SIP Over TLS	Refer to the Genesys 8.1 SIP Server Deployment Guide.
SRTP	No configuration is required.

Service Monitoring Test case: 26	Specify how often (in seconds) SIP Server should check a device for out-of-service status. In the Trunk DN object (representing the SBC) - > TServer section, configure: **oos-check=10** Specify when SIP Server should place a non-responding device into out-of-service status. In the Trunk DN object (representing the SBC) - > TServer section, configure: **oos-force=5**
SIP Server High Availability - with Virtual IP Address	Refer to the Genesys SIP Server High-Availability Deployment Guide.
SIP Server High Availability - DNS-based Redundancy with SIP Proxy	Requires HA deployment using SIP Proxy. SIP Proxy can be used in the SIP Server standalone deployment or Genesys Business Continuity with SIP Proxy deployment. Refer to the Genesys SIP Proxy Deployment Guide and Genesys SIP Server High-Availability Deployment Guide.
Audio Codec Support	No configuration is required.
SBC-Specific Features	**Key Actions and Procedures**
Inbound & Outbound Calls Test cases: 29, 30	1. Deploy one instance of SIP Server. See the "Inbound Calls" and "Outbound call" features, above. 2. Point a phone to the SBC IP address.
SIP Agent 3PCC Control Test case: 29	In the DN object of type Extension -> TServer section, specify support for the BroadSoft Extension Event Package: **sip-cti-control=talk, hold** See the Known Issues section. **Note:** If required, specify the method of hold media SDP (RFC2543-compliant implementation) to be used by SIP Server for third-party call control (3pcc) hold operations. In the SIP Server Application -> TServer section, configure: **sip-hold-rfc3264=false**
Transfer with REFER Test case: 30	Specify the REFER method to be used for 3pcc transfer operations. In the Trunk DN object (representing the SBC) - > TServer section and in the DN object of type Extension -> TServer section, configure: **refer-enabled=true**
Transfer with re-INVITE Test case: 31	Specify the re-INVITE method to be used for the 3pcc Attended transfer. In the DN object of type Extension (transfer controller) -> TServer section, configure: **refer-enabled=false**

5 Oracle Enterprise Session Border Controller Configuration

This section provides general guidelines for configuring the Oracle Enterprise SBC. Genesys recommends consulting Oracle Enterprise SBC documentation for more information.

Oracle Enterprise Session Border Controller Configuration	
Features Supported By SBC	
Feature	**Key Actions and Procedures**
Standard Configuration for SBC	Configure the basic setup including media-manager, network-interface for all physical connections, phy-interface, sip-config, system-config, steering-pool, and sip-interface for all connections.
Realm configuration	realm-config identifier Cisco description addr-prefix 0.0.0.0 network-interfaces M00:0 mm-in-realm enabled mm-in-network enabled mm-same-ip enabled mm-in-system enabled bw-cac-non-mm disabled msm-release disabled qos-enable disabled generate-UDP-checksum disabled max-bandwidth 0 fallback-bandwidth 0 max-priority-bandwidth 0 max-latency 0 max-jitter 0 max-packet-loss 0 observ-window-size 0 parent-realm dns-realm media-policy media-sec-policy srtp-msm-passthrough disabled class-profile in-translationid out-translationid in-manipulationid out-manipulationid average-rate-limit 0 access-control-trust-level none invalid-signal-threshold 0 maximum-signal-threshold 0 untrusted-signal-threshold 0

	nat-trust-threshold 0 max-endpoints-per-nat 0 nat-invalid-message-threshold 0 wait-time-for-invalid-register 0 deny-period 30 cac-failure-threshold 0 untrust-cac-failure-threshold 0 ext-policy-svr diam-e2-address-realm subscription-id-type END_USER_NONE symmetric-latching disabled pai-strip disabled trunk-context device-id early-media-allow enforcement-profile additional-prefixes restricted-latching none restriction-mask 32 user-cac-mode none user-cac-bandwidth 0 user-cac-sessions 0 icmp-detect-multiplier 0 icmp-advertisement-interval 0 icmp-target-ip monthly-minutes 0 options spl-options accounting-enable enabled net-management-control disabled delay-media-update disabled refer-call-transfer disabled refer-notify-provisional none dyn-refer-term disabled codec-policy codec-manip-in-realm disabled codec-manip-in-network enabled rtcp-policy constraint-name session-recording-server session-recording-required disabled manipulation-string manipulation-pattern stun-enable disabled stun-server-ip 0.0.0.0 stun-server-port 3478 stun-changed-ip 0.0.0.0 stun-changed-port 3479 sip-profile sip-isup-profile match-media-profiles qos-constraint

	block-rtcp disabled hide-egress-media-update disabled tcp-media-profile monitoring-filters node-functionality default-location-string alt-family-realm pref-addr-type none Similar configuration should be in place for all other network connections.
Session agent configuration	hostname 192.168.2.228 ip-address 192.168.2.228 port 23846 state enabled app-protocol SIP app-type transport-method UDP realm-id core egress-realm-id description carriers allow-next-hop-lp enabled constraints disabled max-sessions 0 max-inbound-sessions 0 max-outbound-sessions 0 max-burst-rate 0 max-inbound-burst-rate 0 max-outbound-burst-rate 0 max-sustain-rate 0 max-inbound-sustain-rate 0 max-outbound-sustain-rate 0 min-seizures 5 min-asr 0 time-to-resume 0 ttr-no-response 0 in-service-period 0 burst-rate-window 0 sustain-rate-window 0 req-uri-carrier-mode None proxy-mode redirect-action loose-routing enabled send-media-session enabled response-map ping-method ping-interval 0 ping-send-mode keep-alive ping-all-addresses disabled ping-in-service-response-codes out-service-response-codes

	load-balance-dns-query hunt options spl-options media-profiles in-translationid out-translationid trust-me disabled request-uri-headers stop-recurse local-response-map ping-to-user-part ping-from-user-part in-manipulationid out-manipulationid manipulation-string manipulation-pattern p-asserted-id trunk-group max-register-sustain-rate 0 early-media-allow invalidate-registrations disabled rfc2833-mode none rfc2833-payload 0 codec-policy enforcement-profile refer-call-transfer disabled refer-notify-provisional none reuse-connections NONE tcp-keepalive none tcp-reconn-interval 0 max-register-burst-rate 0 register-burst-window 0 sip-profile sip-isup-profile kpml-interworking inherit monitoring-filters session-recording-server session-recording-required disabled Similar configuration should be in place for all endpoints.
Local policy configuration	local-policy from-address * to-address * source-realm Cisco description activate-time deactivate-time state enabled policy-priority none policy-attribute next-hop 192.168.2.228

	realm core action none terminate-recursion disabled carrier start-time 0000 end-time 2400 days-of-week U-S cost 0 state enabled app-protocol SIP methods media-profiles lookup single next-key eloc-str-lkup disabled eloc-str-match

6 Known Issues and Limitations

6.1 Issues and Limitations Identified with Genesys Products

An external caller might not hear Music on Hold when a call was placed on hold by an agent using the 3pcc hold operation. This issue occurs when an agent phone is REGISTERed to SIP Server through the SBC and the hold operation was done through the RFC-3264-compliant implementation (sip-hold-rfc3264=true).