Genesys Application Note

Cisco Unified Border Element (CUBE) SBC With Genesys SIP Server

Document Version 1.0

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Trademarks

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1 Summary

Cisco Unified Border Element (CUBE) as an enterprise session border controller is recommended for integration with the Genesys SIP solution.
This application note details the supported features, and includes reference configuration examples.

The following version of the Cisco Unified Border Element (CUBE) as an enterprise session border controller was tested and supported:

- Cisco Unified Border Element (CUBE) SBC version 11.0.0, software 15.5.2.S2, platform ASR 1001

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), GVP (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	Not Tested
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	Not Tested
SIP Server High Availability - DNS-based Redundancy with SIP Proxy	Yes
SIP Trunk/SBC/Gateway High Availability - DNS-based Redundancy	Not Tested
Audio Codec Support	Yes
Video Support	Not Tested
SBC-Specific Features	**Supported**
Inbound & Outbound Calls	No
SIP Agent 3PCC Control	No
Remote Agent - Transfer with REFER (SIP Phone via SBC)	No
Transfer with REFER	No
Transfer with re-INVITE	No

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	Not tested
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	No
30	3PCC Outbound Call from Remote SIP endpoint to external destination	No
31	3PCC Two Step Transfer from Remote SIP endpoint to internal destination	No
32	1PCC Attended Transfer from Remote SIP endpoint to external destination	No

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	Test Cases
"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

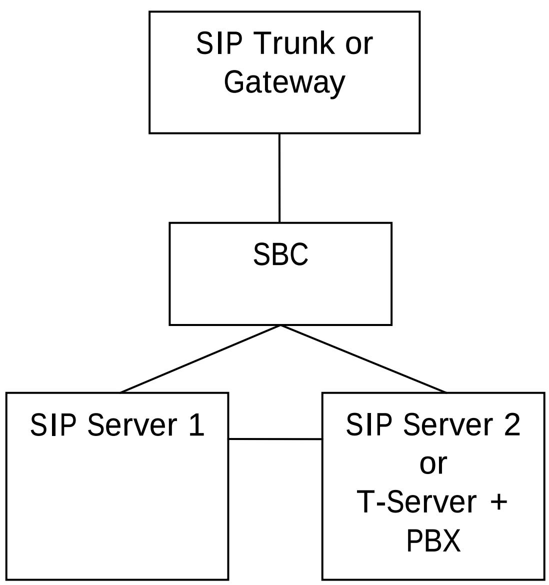
2.5 SBC-Specific Features

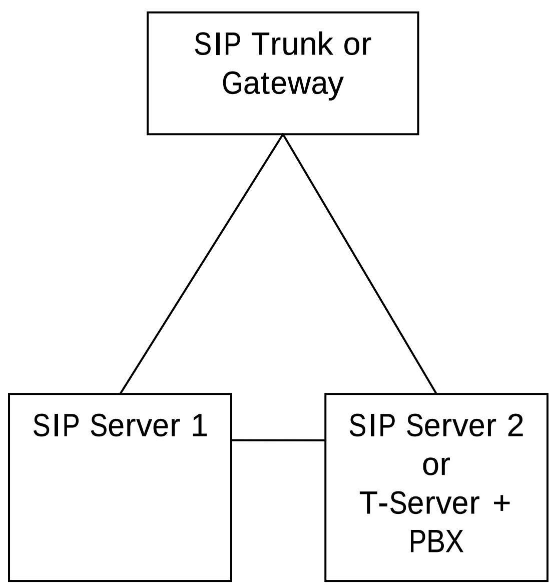
SBC Feature Compatibility for Agent REGISTERed to SIP Server through SBC	Description	Supported	Test Cases
Inbound & Outbound Calls		No	29,30
SIP Agent 3PCC Control		No	29
Remote Agent - Transfer with REFER (SIP Phone via SBC)		No	30
Transfer with REFER		No	30
Transfer with reINVITE		No	31

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 CUBE 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	8.5	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**
Cisco Unified Border Element (CUBE) SBC	11.0.0 software 15.5.2.S2 platform ASR 1001	

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

4 Features Configuration in Genesys Configuration Environment

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

Genesys SIP Configuration	
Features Supported By Gateway/SBC	
General Features	**Key Actions and Procedures**
Inbound Calls – Standard Test cases: 1, 2, 3, 4, 5, 6, 7, 12, 13, 22, 23	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).
Inbound Calls - Contact Center/Routed	Same configuration as for Inbound Calls - Standard, above.
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 DN (representing the SBC) -> TServer section, configure: **cpd-capability=mediaserver** 2. Specify the re-INVITE method to be used for 3pcc operations (outgoing calls, consultation calls, transfer completion). In the Trunk DN (representing the SBC) -> TServer section, configure: **refer-enabled=false**
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 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 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 SIP Server Deployment Guide for details.
SIP Over TLS	Please refer to the 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.

SIP Server High Availability - DNS-based Redundancy with SIP Proxy	Requires HA deployment using SIP Proxy deployment. SIP Proxy can be used in 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.

5 CUBE Enterprise Session Border Controller Configuration

This section describes how to configure the CUBE SBC.

CUBE SBC Configuration	
Features Supported By SBC	
Feature	**Key Actions and Procedures**
Standard Configuration for SBC	Current configuration: 4814 bytes ! version 15.5 service timestamps debug datetime msec service timestamps log datetime msec no platform punt-keepalive disable-kernel-core ! hostname cubea ! boot-start-marker boot system bootflash:asr1001-universalk9.03.15.02.S.155-2.S2-std.bin boot-end-marker ! vrf definition Mgmt-intf ! address-family ipv4 exit-address-family ! address-family ipv6 exit-address-family ! enable password telecom ! no aaa new-model ! ip domain name mycompany.com ip name-server 192.168.2.123 ip name-server 192.168.2.124 !

	ipv6 multicast rpf use-bgp
	ipv6 multicast vrf Mgmt-intf rpf use-bgp
	!
	subscriber templating
	!
	multilink bundle-name authenticated
	!
	voice rtp send-recv
	!
	voice service voip
	 ip address trusted list
	  ipv4 192.168.3.241
	  ipv4 192.168.3.242
	 address-hiding
	 mode border-element license capacity 200
	 allow-connections sip to sip
	 redirect ip2ip
	 sip
	  header-passing
	  error-passthru
	  registrar server
	  asymmetric payload dtmf
	  early-offer forced
	  midcall-signaling passthru
	  privacy-policy passthru
	  g729 annexb-all
	!
	voice class codec 1
	 codec preference 1 g711ulaw
	 codec preference 3 g723r63
	 codec preference 4 g729r8
	 codec preference 5 g729br8
	!
	redundancy
	 mode none
	!
	interface GigabitEthernet0/0/0
	 ip address 192.168.6.243 255.255.255.0
	 ip route-cache same-interface
	 negotiation auto
	!
	interface GigabitEthernet0/0/1
	 ip address 135.17.64.11 255.255.255.0
	 ip route-cache same-interface
	 negotiation auto
	!
	interface GigabitEthernet0
	 vrf forwarding Mgmt-intf
	 ip address 192.168.6.246 255.255.255.0
	 negotiation auto
	!
	ip forward-protocol nd

```
!
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route profile
ip route 0.0.0.0 0.0.0.0 192.168.6.1
ip route 0.0.0.0 0.0.0.0 135.17.64.1
ip route vrf Mgmt-intf 0.0.0.0 0.0.0.0 192.168.6.1
!
control-plane
!
dial-peer voice 1 voip
 description Incoming from PSTN Gateway
 session protocol sipv2
 incoming called-number 16..
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
dial-peer voice 163 voip
 tone ringback alert-no-PI
 description Outgoing to Genesys SIP Proxy
 destination-pattern 163.
 session protocol sipv2
 session target dns:SPCube.qa.sipcluster.genesyslab.com:5060
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
dial-peer voice 166 voip
 tone ringback alert-no-PI
 description Outgoing to Genesys SIP-Server
 destination-pattern 166.
 session protocol sipv2
 session target ipv4:10.10.18.32:23848
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
dial-peer voice 2 voip
 tone ringback alert-no-PI
 description Incoming from Genesys SIP-Server
 session protocol sipv2
 incoming called-number 8340....
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
```

	dtmf-relay rtp-nte codec g711ulaw no vad ! dial-peer voice 8340 voip tone ringback alert-no-PI description Outgoing to PSTN Gateway destination-pattern 8340.... session protocol sipv2 session target ipv4:192.168.6.31 voice-class sip bind control source-interface GigabitEthernet0/0/0 voice-class sip bind media source-interface GigabitEthernet0/0/0 dtmf-relay rtp-nte codec g711ulaw no vad ! sip-ua ! end

6 Known Issues and Limitations

6.1 Issues and Limitations Identified with Genesys Products

Support is limited to the trunk side only.