



Genesys Application Note

AudioCodes Mediant SBC With Genesys SIP Server

Document version 1.1

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1 Summary

AudioCodes Mediant Session Border Controller (SBC) is recommended for integration with the Genesys SIP solution.

As noted in [section 2](#) and [4](#) below, all test calls/cases were successfully executed.

This application note is applicable to the following AudioCodes products:

- Mediant 500 SBC
- Mediant 800B SBC
- Mediant 1000B SBC
- Mediant 2600 SBC
- Mediant 3000 SBC
- Mediant 4000 SBC
- Mediant 9000 SBC
- Mediant Software and Virtual SBC

The supporting versions of Genesys components include SIP Server 8.1.1, SIP Feature Server 8.1.2, Media Server (8.1.x and 8.5.x), and SIP Proxy v8.1.1.

2 Feature Support

2.1 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 (CPD) 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.

Note: Support for Answering Machine Detection (AMD) and Call Progress Tone Detection (CPD) on the Mediant 4000, Mediant 9000, and the Mediant Software and Virtual SBC is planned for the AudioCodes 7.0 software release. It is supported on other AudioCodes SBC devices.

2.2 Technical Features

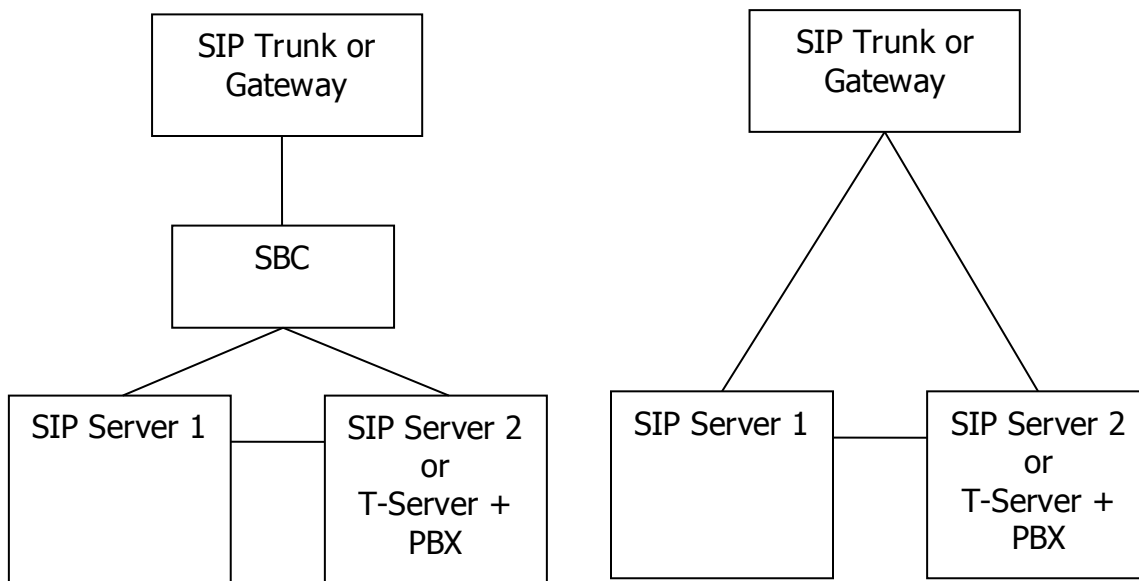
Technical Compatibility – Architecture & SIP Protocol	Description	Supported	Test Cases
"Single Site"	One instance of Genesys SIP Server	Yes	All test cases apply
"Multisite"	Two or more instances of Genesys SIP Server, behind a single Trunk and/or SBC		No "dedicated" test cases
Transfer with re-INVITE	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		Yes	27, 28
SIP Over TLS	See the Genesys 8.1 SIP Server Deployment Guide for details.	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 "negotiated preferred" 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	[not tested – requires supplemental testing]	No dedicated test cases

2.3 SBC-specific Features

SBC Feature Compatibility for Agent REGISTERed to SIP Server through SBC	Description	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

2.4 Details Regarding Features

2.4.1 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.

2.4.2 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. A Genesys SIP Server, an SBC, or the components may employ this high availability on 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 back IP address for a component (using the contact and contacts-backup options). This type of method is referred to as a "list of IP Addresses" in this application note. In another example, a 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 high availability 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 AudioCodes 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.
Genesys Media Server	8.1.700	Used to handle media interactions such as call treatments (ring back, busy tones and music on hold); also used as MCU.
Genesys SIP Feature Server	8.1.2	Used as a SIP Voicemail Server
SIP Proxy	8.1.1	Optionally can be used for DNS-based HA deployment

3.2 Gateway/SBC

3 rd Party Hardware Components		
Model	Version	Notes
AudioCodes Mediant SBC	6.8	

For a full listing of 3rd party hardware/software supported by Genesys, see the [Genesys Supported Media Interface Guide \(SMI\)](#) and the [SIP Integration Reference](#).

4 Functional Test Case Scenarios

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	Yes
28	SIP Authentication for incoming calls	Yes
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 2 Step Transfer from Remote SIP endpoint to internal destination	Yes
32	1PCC Attended Transfer from Remote SIP endpoint to external destination	Yes

5 Features Configuration in Genesys Configuration Environment

Genesys SIP Configuration																																		
Features Supported By Gateway/SBC																																		
Feature	Key Actions and Procedures																																	
1, 2, 3, 4, 5, 6, 7, 9, 11, 13, 18, 30, 31	The following is the default/standard SIP Server configuration used during testing of the technical features:																																	
	<table><tr><th>SIP Server Application Options TServer section</th></tr><tr><td>sip-hold-rfc3264=true router-timeout=30 default-dn= blind-transfer-enabled=true resource-management-by-rm=true msml-support=true sip-enable-moh=true</td></tr></table>				SIP Server Application Options TServer section	sip-hold-rfc3264=true router-timeout=30 default-dn= blind-transfer-enabled=true resource-management-by-rm=true msml-support=true sip-enable-moh=true																												
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	sip-hold-rfc3264=true router-timeout=30 default-dn= blind-transfer-enabled=true resource-management-by-rm=true msml-support=true sip-enable-moh=true																																	
	The following is the default/standard for DN configuration used during testing of the technical features:																																	
	<table><tr><th>Name</th><th>Number</th><th>Name in CME</th><th>CME Options TServer section</th><th>Comment</th></tr><tr><td>MGW-TRUNK</td><td>MGW-TRUNK</td><td>MGW-TRUNK</td><td>refer-enabled=true contact=<TSE_CONTACT> oos-check=10 oos-force=5 oosp-transfer-enabled=true sip-replaces-mode=2</td><td>TSE</td></tr><tr><td>Ext-DN1 Ext-DN2</td><td>21001 21002</td><td>N/A</td><td>N/A</td><td></td></tr><tr><td>SIP-DN1 SIP-DN2</td><td>7101 7102</td><td>7101 7102</td><td>refer-enabled=false ring-tone-on-make-call=false make-call-rfc3725-flow=1 contact=*</td><td></td></tr><tr><td>SIP-RDN</td><td>7200</td><td>7200</td><td>refer-enabled=true ring-tone-on-make-call=false make-call-rfc3725-flow=1 contact=*\nsip-cti-control=talk,hold</td><td>SIP endpoint which supports the BroadSoft SIP Extension Event Package.</td></tr><tr><td>SVC_MSML</td><td>SVC_MSML</td><td>SVC_MSML</td><td>prefix=msml= contact=<MS_CONTACT> service-type=msml subscription-id= Environment</td><td>MS</td></tr></table>				Name	Number	Name in CME	CME Options TServer section	Comment	MGW-TRUNK	MGW-TRUNK	MGW-TRUNK	refer-enabled=true contact=<TSE_CONTACT> oos-check=10 oos-force=5 oosp-transfer-enabled=true sip-replaces-mode=2	TSE	Ext-DN1 Ext-DN2	21001 21002	N/A	N/A		SIP-DN1 SIP-DN2	7101 7102	7101 7102	refer-enabled=false ring-tone-on-make-call=false make-call-rfc3725-flow=1 contact=*		SIP-RDN	7200	7200	refer-enabled=true ring-tone-on-make-call=false make-call-rfc3725-flow=1 contact=*\nsip-cti-control=talk,hold	SIP endpoint which supports the BroadSoft SIP Extension Event Package.	SVC_MSML	SVC_MSML	SVC_MSML	prefix=msml= contact=<MS_CONTACT> service-type=msml subscription-id= Environment	MS
	Name	Number	Name in CME	CME Options TServer section	Comment																													
	MGW-TRUNK	MGW-TRUNK	MGW-TRUNK	refer-enabled=true contact=<TSE_CONTACT> oos-check=10 oos-force=5 oosp-transfer-enabled=true sip-replaces-mode=2	TSE																													
	Ext-DN1 Ext-DN2	21001 21002	N/A	N/A																														
	SIP-DN1 SIP-DN2	7101 7102	7101 7102	refer-enabled=false ring-tone-on-make-call=false make-call-rfc3725-flow=1 contact=*																														
SIP-RDN	7200	7200	refer-enabled=true ring-tone-on-make-call=false make-call-rfc3725-flow=1 contact=*\nsip-cti-control=talk,hold	SIP endpoint which supports the BroadSoft SIP Extension Event Package.																														
SVC_MSML	SVC_MSML	SVC_MSML	prefix=msml= contact=<MS_CONTACT> service-type=msml subscription-id= Environment	MS																														
Use the default/standard configuration indicated above																																		

8	In addition to the default/standard configuration indicated above, set the following: oosp-transfer-enabled=true
10	In addition to the default/standard configuration indicated above, set the following: refer-enabled=true
12	In addition to the default/standard configuration indicated above, set the following: sip-hold-rfc3264=false
14	In addition to the default/standard configuration indicated above, set the following: refer-enabled=false
15	In addition to the default/standard configuration indicated above, set the following: refer-enabled=true
16	In addition to the default/standard configuration indicated above, set the following: refer-enabled=true
17	In addition to the default/standard configuration indicated above, set the following: oosp-transfer-enabled=true
19	In addition to the default/standard configuration indicated above, set the following: refer-enabled=true
20	In addition to the default/standard configuration indicated above, set the following: refer-enabled=true oosp-transfer-enabled=true
21	In addition to the default/standard configuration indicated above, set the following: refer-enabled=true sip-busy-type=2
22	In addition to the default/standard configuration indicated above, set the following: sip-early-dialog-mode=1
23	In addition to the default/standard configuration indicated above, set the following: sip-early-dialog-mode=1
24	In addition to the default/standard configuration indicated above, set the following: oosp-transfer-enabled=false
25	In addition to the default/standard configuration indicated above, set the following: cpd-capability = mediaserver refer-enabled=false
26	In addition to the default/standard configuration indicated above, set the following: oos-check=10 oos-force=5

27	In addition to the default/standard configuration indicated above, on the Annex tab, configure the AuthClient section with options username=<username> password=<password>
28	In addition to the default/standard configuration indicated above, set the following: authenticate-requests=invite password=1234
29	In addition to the default/standard configuration indicated above, set the following: sip-cti-control=talk,hold authenticate-requests=REGISTER password=1234
32	In addition to the default/standard configuration indicated above, set the following: refer-enabled=false

6 Gateway/SBC Configuration

This section describes how to configure features represented in the Feature Support (see [section 2](#)).

Gateway/SBC Configuration																															
Features Supported By Gateway/SBC																															
Feature	Key Actions and Procedures																														
1, 2, 3, 4, 5, 6, 7, 8, 9, 11, 12, 13, 14, 18, 22, 23, 24, 25, 26, 27, 28, 29, 31	<p>The following sections taken in order describe the steps necessary to configure basic SBC functionality for an AudioCodes SBC device. These sections are provided only as a reference. The user should refer back to respective area within the User’s Manual for the particular AudioCodes device for complete details and explanations of all the options.</p> <p>1. Network (LAN & WAN)</p> <p>a. Define the network interface for the Genesys (Trusted or LAN) Network. Reset is required.</p> <p><i>Configuration > VoIP > Network > IP Interfaces Table</i></p> <div><div>Interface Table</div><div><div>▼ Interface Table</div><div><div>Add +Edit ↗Delete 🗑</div><div>Show/Hide 📄</div></div><table><tr><th>Index ↕</th><th>Application Type</th><th>Interface Mode</th><th>IP Address</th><th>Prefix Length</th><th>Default Gateway</th><th>Interface Name</th><th>Primary DNS</th><th>Secondary DNS</th><th>Underlying Device</th></tr><tr><td>0</td><td>OAMP + Media + C</td><td>IPv4 Manual</td><td>192.168.20.200</td><td>24</td><td>192.168.20.1</td><td>Genesys</td><td>0.0.0.0</td><td>0.0.0.0</td><td>GROUP_1</td></tr><tr><td>1</td><td>Media + Control</td><td>IPv4 Manual</td><td>203.0.113.120</td><td>26</td><td>203.0.113.65</td><td>Provider</td><td>0.0.0.0</td><td>0.0.0.0</td><td>GROUP_2</td></tr></table><div><div>⏮⏪⏩⏭</div><div>Page 1 of 1</div><div>⏭⏩⏪⏮</div><div>Show 10 records per page</div><div>View 1 - 2 of 2</div></div><div><div>Selected Row #0</div><div><div>Application Type:</div><div>OAMP + Media + Control</div><div>Interface Mode:</div><div>IPv4 Manual</div><div>IP Address:</div><div>192.168.20.200</div><div>Prefix Length:</div><div>24</div><div>Default Gateway:</div><div>192.168.20.1</div><div>Interface Name:</div><div>Genesys</div><div>Primary DNS:</div><div>0.0.0.0</div><div>Secondary DNS:</div><div>0.0.0.0</div><div>Underlying Device:</div><div>GROUP_1</div></div></div></div></div> <p>Define the network interface for the ITSP (Untrusted or WAN) Network. Reset is required.</p> <p><i>Configuration > VoIP > Network > IP Interfaces Table</i></p>	Index ↕	Application Type	Interface Mode	IP Address	Prefix Length	Default Gateway	Interface Name	Primary DNS	Secondary DNS	Underlying Device	0	OAMP + Media + C	IPv4 Manual	192.168.20.200	24	192.168.20.1	Genesys	0.0.0.0	0.0.0.0	GROUP_1	1	Media + Control	IPv4 Manual	203.0.113.120	26	203.0.113.65	Provider	0.0.0.0	0.0.0.0	GROUP_2
	Index ↕	Application Type	Interface Mode	IP Address	Prefix Length	Default Gateway	Interface Name	Primary DNS	Secondary DNS	Underlying Device																					
	0	OAMP + Media + C	IPv4 Manual	192.168.20.200	24	192.168.20.1	Genesys	0.0.0.0	0.0.0.0	GROUP_1																					
1	Media + Control	IPv4 Manual	203.0.113.120	26	203.0.113.65	Provider	0.0.0.0	0.0.0.0	GROUP_2																						

Interface Table									
▼ Interface Table									
Add +		Edit ✎		Delete 🗑		Show/Hide ⌵			
Index ↕	Application Type	Interface Mode	IP Address	Prefix Length	Default Gateway	Interface Name	Primary DNS	Secondary DNS	Underlying Device
0	OAMP + Media + C	IPv4 Manual	192.168.20.200	24	192.168.20.1	Genesys	0.0.0.0	0.0.0.0	GROUP_1
1	Media + Control	IPv4 Manual	203.0.113.120	26	203.0.113.65	Provider	0.0.0.0	0.0.0.0	GROUP_2

Page 1 of 1 Show 10 records per page View 1 - 2 of 2

Selected Row #1

Application Type:	Media + Control	Interface Name:	Provider
Interface Mode:	IPv4 Manual	Primary DNS:	0.0.0.0
IP Address:	203.0.113.120	Secondary DNS:	0.0.0.0
Prefix Length:	26	Underlying Device:	GROUP_2
Default Gateway:	203.0.113.65		

2. SBC Application Enabling

Enable the SBC Application (for a device which isn't a 'pure' eSBCs). Reset is required.

[Configuration > Applications Enabling](#)

Applications Enabling	
<div> <div>⚙</div> <div>SAS Application</div> </div>	Disable
<div> <div>⚙</div> <div>SBC Application</div> </div>	Enable

3. Media Realm

Define Media Realms for the Media interfaces (both Genesys & ITSP Provider).

[Configuration > VoIP > Network > Media Realm Table](#)

Media Realm Table

Media Realm Table

Add + Edit ✎ Delete 🗑 Show/Hide 📄

Index	Media Realm Name	IPv4 Interface Name	IPv6 Interface Name
1	MR1-SBC2Genesys	Genesys	None
2	MR2-SBC2Provider	Provider	None

Page 1 of 1 Show 10 records per page View 1 - 2 of 2

Selected Row #1

Media Realm Name:	MR1-SBC2Genesys	Port Range End:	6990
IPv4 Interface Name:	Genesys	Default Media Realm:	Yes
IPv6 Interface Name:	None	QoE Profile:	None
Port Range Start:	6000	BW Profile:	None
Number Of Media Session Legs:	100		

CpMediaRealm #1 Additional Configuration

[Media Realm Extension](#) [contains 0 entries]

[Remote Media Subnet](#) [contains 0 entries]

Media Realm Table

Media Realm Table

Add + Edit ✎ Delete 🗑 Show/Hide 📄

Index	Media Realm Name	IPv4 Interface Name	IPv6 Interface Name
1	MR1-SBC2Genesys	Genesys	None
2	MR2-SBC2Provider	Provider	None

Page 1 of 1 Show 10 records per page View 1 - 2 of 2

Selected Row #2

Media Realm Name:	MR2-SBC2Provider	Port Range End:	7990
IPv4 Interface Name:	Provider	Default Media Realm:	No
IPv6 Interface Name:	None	QoE Profile:	None
Port Range Start:	7000	BW Profile:	None
Number Of Media Session Legs:	100		

CpMediaRealm #2 Additional Configuration

[Media Realm Extension](#) [contains 0 entries]

[Remote Media Subnet](#) [contains 0 entries]

4. SRDs

Define SRDs for the networks (both Genesys & ITSP Provider).

[Configuration > VoIP > Network > SRD Table](#)

SRD Table			
<div> Add + Edit Delete </div> <div> Show/Hide </div>			
Index	Name	Media Realm Name	Media Anchoring
1	SRD1-Genesys	MR1-SBC2Genesys	Enable
2	SRD2-Provider	MR2-SBC2Provider	Enable
Page 1 of 1 Show 10 records per page View 1 - 2 of 2			
Selected Row #1			
Name:	SRD1-Genesys	Block Unregistered Users:	NO
Media Realm Name:	MR1-SBC2Genesys	Max. Number of Registered Users:	-1
Media Anchoring:	Enable	Enable Un-Authenticated Registrations:	Enable

SRD Table			
<div> Add + Edit Delete </div> <div> Show/Hide </div>			
Index	Name	Media Realm Name	Media Anchoring
1	SRD1-Genesys	MR1-SBC2Genesys	Enable
2	SRD2-Provider	MR2-SBC2Provider	Enable
<div>SRD2-Provider</div>			
Page 1 of 1 Show 10 records per page View 1 - 2 of 2			
Selected Row #2			
Name:	SRD2-Provider	Block Unregistered Users:	NO
Media Realm Name:	MR2-SBC2Provider	Max. Number of Registered Users:	-1
Media Anchoring:	Enable	Enable Un-Authenticated Registrations:	Enable

5. SIP Interfaces

Define SIP Interfaces.

Configuration > VoIP > Network > SIP Interface Table

SIP Interface Table							
▼ SIP Interface Table							
<div> Add + Edit ✎ Delete 🗑 </div> <div>Show/Hide ⌵</div>							
Index ↕	SIP Interface Name	Network Interface	Application Type	UDP Port	TCP Port	TLS Port	SRD
1	Genesys	Genesys	SBC	5060	5060	5061	1
2	ITSP	Provider	SBC	5060	5060	5061	2
<div> Page 1 of 1 Show 10 records per page View 1 - 2 of 2 </div>							
Selected Row #1							
<div> <div> SIP Interface Name: Genesys Network Interface: Genesys Application Type: SBC UDP Port: 5060 TCP Port: 5060 TLS Port: 5061 </div> <div> SRD: 1 Message Policy: None TLS Context Name: None TLS Mutual Authentication: Enable TCP Keepalive: Disable Classification Failure Response Type: 500 </div> </div>							

SIP Interface Table							
▼ SIP Interface Table							
<div> Add + Edit ✎ Delete 🗑 </div> <div>Show/Hide ⌵</div>							
Index ↕	SIP Interface Name	Network Interface	Application Type	UDP Port	TCP Port	TLS Port	SRD
1	Genesys	Genesys	SBC	5060	5060	5061	1
2	ITSP	Provider	SBC	5060	5060	5061	2
<div> Page 1 of 1 Show 10 records per page View 1 - 2 of 2 </div>							
Selected Row #2							
<div> <div> SIP Interface Name: ITSP Network Interface: Provider Application Type: SBC UDP Port: 5060 TCP Port: 5060 TLS Port: 5061 </div> <div> SRD: 2 Message Policy: None TLS Context Name: None TLS Mutual Authentication: Enable TCP Keepalive: Disable Classification Failure Response Type: 500 </div> </div>							

6. IP Groups

Define the supporting IP Groups.

[Configuration > VoIP > Network > IP Group Table](#)

IP Group Table					
<div> Add + Edit ✎ Delete 🗑 Show/Hide ⏏ </div>					
Index ↑	Type	Description	Proxy Set ID	SIP Group Name	Contact User
1	Server	IPG1-SBC2Genesys	1	svoice-99-sip.domain.com	
2	Server	IPG2-SBC2Provider	2	gw0.itsp-domain.com	
Page 1 of 1 Show 10 records per page View 1 - 2 of 2					
Selected Row #1					
<div> <div> Common </div> <div> Type: Server Description: IPG1-SBC2Genesys Proxy Set ID: 1 SIP Group Name: svoice-99-sip.domain.com Contact User: SRD: 1 Media Realm Name: MR1-SBC2Genesys IP Profile ID: 1 Local Host Name: sbc-01-sip.domain.com UUI Format: Disable QoE Profile: None Bandwidth Profile: None Media Enhancement Profile: None Always Use Source Address: No </div> <div> SBC </div> <div> Classify By Proxy Set: Enable Max. Number of Registered Users: -1 Inbound Message Manipulation Set: -1 Outbound Message Manipulation Set: -1 Registration Mode: User Initiates Registration Authentication Mode: User Authenticates Authentication Method List: SBC Client Forking Mode: Sequential Source URI Input: Destination URI Input: Username: Password: Msg Man User Defined String1: Msg Man User Defined String2: </div> </div>					

IP Group Table					
<div> Add + Edit ✎ Delete 🗑 Show/Hide ⏏ </div>					
Index ↑	Type	Description	Proxy Set ID	SIP Group Name	Contact User
1	Server	IPG1-SBC2Genesys	1	svoice-99-sip.domain.com	
2	Server	IPG2-SBC2Provider	2	gw0.itsp-domain.com	
Page 1 of 1 Show 10 records per page View 1 - 2 of 2					
Selected Row #2					
<div> <div> Common </div> <div> Type: Server Description: IPG2-SBC2Provider Proxy Set ID: 2 SIP Group Name: gw0.itsp-domain.com Contact User: SRD: 2 Media Realm Name: MR2-SBC2Provider IP Profile ID: 2 Local Host Name: UUI Format: Disable QoE Profile: None Bandwidth Profile: None Media Enhancement Profile: None Always Use Source Address: No </div> <div> SBC </div> <div> Classify By Proxy Set: Enable Max. Number of Registered Users: -1 Inbound Message Manipulation Set: -1 Outbound Message Manipulation Set: -1 Registration Mode: User Initiates Registration Authentication Mode: User Authenticates Authentication Method List: SBC Client Forking Mode: Sequential Source URI Input: Destination URI Input: Username: Password: Msg Man User Defined String1: Msg Man User Defined String2: </div> </div>					

7. Proxy Sets

Define Proxy Sets.

Add the address(es) or FQDN of the Genesys SIP Server(s) to the Proxy Set assigned to the IP Group in the previous step.

Note: Proxy Set ID 0 (zero) is reserved as a 'default proxy' and should not be used with IP groups.

Note: While a single proxy is shown below, a Proxy Set should be created for each defined IP Group.

[Configuration > VoIP > Network > Proxy Sets Table](#)

Proxy Sets Table

Proxy Set ID		1
--------------	--	---

	Proxy Address	Transport Type
1	svoice-99-sip.domain.com	
2		
3		
4		
5		
6		
7		
8		
9		
10		

Proxy Name	Genesys SIP Server
Enable Proxy Keep Alive	Using Options
Proxy Keep Alive Time	60
Proxy Load Balancing Method	Disable
Is Proxy Hot Swap	No
Proxy Redundancy Mode	Not Configured
SRD Index	1
Classification Input	IP only

8. IP-to-IP Routing

Define any required IP-to-IP Routing rules.

For this example (associated rule highlighted for each item):

0. OPTIONS are terminated at the SBC.
1. All calls from ITSP will route to Genesys SIP Server.
2. All calls from Genesys environment will route to the ITSP.

Configuration > VoIP > SBC > Routing SBC > IP-to-IP Routing Table

IP-to-IP Routing Table

Index	Route Name	Source Host	Destination Username Prefix	Destination Host	Message Condition	ReRoute IP Group ID	Call Trigger	Call Setup Rules Set ID	Destination Type	Destination SRD ID
0	OPTIONS	*	*	*	None	-1	Any	-1	Dest Address	None
1		*	*	*	None	-1	Any	-1	IP Group	2
2		*	*	*	None	-1	Any	-1	IP Group	1

Page 1 of 1 Show 10 records per page View 1 - 3 of 3

Selected Row #0

Rule		Action	
Route Name:	OPTIONS	Destination Type:	Dest Address
Source IP Group ID:	-1	Destination IP Group ID:	-1
Source Username Prefix:	*	Destination SRD ID:	None
Source Host:	*	Destination Address:	internal
Destination Username Prefix:	*	Destination Port:	0
Destination Host:	*	Destination Transport Type:	
Request Type:	All	Alternative Route Options:	Route Row
Message Condition:	None	Group Policy:	None
ReRoute IP Group ID:	-1	Cost Group:	None
Call Trigger:	Any		
Call Setup Rules Set ID:	-1		

IP-to-IP Routing Table

IP-to-IP Routing Table

<div> Add + Insert + Edit ↗ Delete 🗑 Up ↑ Down ↓ </div> <div>Show/Hide ⌵</div>										
Index ↕	Route Name	Source Host	Destination Username Prefix	Destination Host	Message Condition	ReRoute IP Group ID	Call Trigger	Call Setup Rules Set ID	Destination Type	Destination SRD ID
0	OPTIONS	*	*	*	None	-1	Any	-1	Dest Address	None
1		*	*	*	None	-1	Any	-1	IP Group	2
2		*	*	*	None	-1	Any	-1	IP Group	1

Page 1 of 1 Show 10 records per page

View 1 - 3 of 3

Selected Row #1

Rule

Route Name:
Source IP Group ID: 1
Source Username Prefix: *
Source Host: *
Destination Username Prefix: *
Destination Host: *
Request Type: All
Message Condition: None
ReRoute IP Group ID: -1
Call Trigger: Any
Call Setup Rules Set ID: -1

Action

Destination Type: IP Group
Destination IP Group ID: 2
Destination SRD ID: 2
Destination Address:
Destination Port: 0
Destination Transport Type:
Alternative Route Options:
Group Policy: None
Cost Group: None

IP-to-IP Routing Table

IP-to-IP Routing Table

<div> Add + Insert + Edit ↗ Delete 🗑 Up ↑ Down ↓ </div> <div>Show/Hide ⌵</div>										
Index ↕	Route Name	Source Host	Destination Username Prefix	Destination Host	Message Condition	ReRoute IP Group ID	Call Trigger	Call Setup Rules Set ID	Destination Type	Destination SRD ID
0	OPTIONS	*	*	*	None	-1	Any	-1	Dest Address	None
1		*	*	*	None	-1	Any	-1	IP Group	2
2		*	*	*	None	-1	Any	-1	IP Group	1

Page 1 of 1 Show 10 records per page

View 1 - 3 of 3

Selected Row #2

Rule

Route Name:
Source IP Group ID: 2
Source Username Prefix: *
Source Host: *
Destination Username Prefix: *
Destination Host: *
Request Type: All
Message Condition: None
ReRoute IP Group ID: -1
Call Trigger: Any
Call Setup Rules Set ID: -1

Action

Destination Type: IP Group
Destination IP Group ID: 1
Destination SRD ID: 1
Destination Address:
Destination Port: 0
Destination Transport Type:
Alternative Route Options:
Group Policy: None
Cost Group: None

9. Digit/SIP-URI Manipulation

Define any required IP-to-IP Inbound/Outbound Manipulations.

In this example, the leading “+” is stripped from Destination numbers from Genesys toward the ITSP as needed.

Configuration > VoIP > SBC > Routing SBC > IP-to-IP Inbound Table

IP to IP Inbound Manipulation												
▼ IP to IP Inbound Manipulation												
Add + Insert + Edit ✎ Delete 🗑 Up ↑ Down ↓ Show/Hide 📄												
Index #	Manipulation Name	Additional Manipulation	Manipulation Purpose	Source IP Group ID	Source Username Prefix	Source Host	Destination Username Prefix	Destination Host	Request Type	Manipulated URI	Prefix to Add	Suffix to Add
1	Leading +	No	Normal	1	*	*	+	*	All	Destination		
Page 1 of 1 Show 10 records per page View 1 - 1 of 1												
Selected Row #1												
Rule				Action								
Manipulation Name:				Leading +				Remove From Left:				
Additional Manipulation:				No				Remove From Right:				
Manipulation Purpose:				Normal				Leave From Right:				
Source IP Group ID:				1				Prefix to Add:				
Source Username Prefix:				*				Suffix to Add:				
Source Host:				*								
Destination Username Prefix:				+								
Destination Host:				*								
Request Type:				All								
Manipulated URI:				Destination								

10. Message Manipulation

Define any required Message Manipulations.

An explanation for this example:

1. The ITSP sends new INVITES when a SIP 603 Declined response is returned on an initial INVITE.
2. The ITSP does not send the new SIP INVITES for SIP 600 Busy Everywhere response.

In this case, the header.request-uri.methodtype changes the '603' to '600' response code preventing new SIP INVITES for the same call.

Note: The inbound/outbound manipulation set identifier should be configured against the appropriate IP Group in the IP Group table. In this case, "Outbound Message Manipulation Set" for IP Group 2 (for the ITSP/Provider) should be configured to '1'.

Note: Depending upon the ITSP/Provider, the need may exist to implement several message manipulations to include a Diversion header or to modify the Contact header (the later in cases where external callers are being referred out to the network) Additionally, there may exist, depending upon the customer, manipulation rules to do topology hiding.

[Configuration > VoIP > SIP Definitions > Msg Policy & Manipulation > Message Manipulations](#)

Message Manipulations							
<div> <div>▼ Message Manipulations</div> <div> <div>Add +</div> <div>Insert +</div> <div>Edit ↗</div> <div>Delete 🗑</div> <div>Up ↑</div> <div>Down ↓</div> </div> <div>Show/Hide 📄</div> </div>							
Index ↕	Manipulation Name	Manipulation Set ID	Message Type	Condition	Action Subject	Action Type	Action Value
1	change 603	1	Invite.Response	header.request-uri.metho	header.request-uri.metho	Modify	'600'
<div> <div>⏪ ⏩</div> <div>Page 1 of 1</div> <div>⏪ ⏩</div> <div>Show 10 records per page</div> <div>View 1 - 1 of 1</div> </div>							
<div>Selected Row #1</div> <div> <div>Manipulation Name:</div> <div>change 603</div> <div>Action Subject:</div> <div>header.request-uri.methodtype</div> </div> <div> <div>Manipulation Set ID:</div> <div>1</div> <div>Action Type:</div> <div>Modify</div> </div> <div> <div>Message Type:</div> <div>Invite.Response</div> <div>Action Value:</div> <div>'600'</div> </div> <div> <div>Condition:</div> <div>header.request-uri.methodtype=='603'</div> <div>Row Role:</div> <div>Use Current Condition</div> </div>							

Taking into account the information provided above when configuring basic SBC functionality for an AudioCodes SBC device, the following sections would be followed when the SBC interfaces an ITSP which does not support use of SIP REFER. These sections are provided only as a reference. The user should refer back to the respective area within the User’s Manual for the particular AudioCodes device in question for complete details and explanations of all options.

Note that by default the SBC device’s handling of SIP 3xx redirect responses is to send the Contact header unchanged. However, some SIP entities may support different versions of the SIP 3xx standard while others may not even support SIP 3xx.

For ITSPs, if SIP REFER isn’t handled, they usually do not support proper SIP 3xx behavior and require equal treatment. That means being handled locally by the AudioCodes SBC with a trigger to direct a SIP INVITE be sent to the ITSP and the call anchored on the SBC (but SIP Server is released from the call). See below:

1. IP Profile

Configure the IP Profile associated to the ITSP to have SIP REFER and 3xx responses handled locally by the SBC.

Configuration > VoIP > Coders and Profiles > IP Profile Settings > Select the IP Profile associated to the SIP Trunk Provider > select the SBC tab

IP Profile Settings

3 Remote Agents Profi

Extension Coders Group ID	None
Transcoding Mode	Only If Required
Allowed Media Types	audio
Allowed Coders Group ID	Coders Group 1
Allowed Video Coders Group ID	None
Allowed Coders Mode	Restriction
SBC Media Security Behavior	As Is
RFC 2833 Behavior	As Is
Alternative DTMF Method	As Is
P-Asserted-Identity	As Is
Diversion Mode	As Is
History-Info Mode	As Is
Fax Coders Group ID	None
Fax Behavior	As Is
Fax Offer Mode	All coders
Fax Answer Mode	Single coder
PRACK Mode	Transparent
Session Expires Mode	Transparent
Remote Update Support	Supported
Remote re-INVITE	Supported
Remote Delayed Offer Support	Supported
Remote REFER Behavior	Handle Locally
Remote 3xx Behavior	Handle Locally
Remote Multiple 1xx	Supported

2. IP-to-IP Routing

Establish routing rule(s) for routing the SIP re-INVITE to the ITSP if the triggering message was a SIP REFER.

[Configuration > VoIP > SBC > Routing SBC > IP-to-IP Routing Table](#) > Add route to get to SIP Trunk provider

IP-to-IP Routing Table

▼ IP-to-IP Routing Table

	Rule	Action
	Index	6
1	Route Name	3xx/REFERS
2	Source IP Group ID	2
3	Source Username Prefix	*
5	Source Host	*
6	Destination Username Prefix	*
7	Destination Host	*
8	Request Type	All
	Message Condition	None
	ReRoute IP Group ID	-1
	Call Trigger	3xx or REFER
	Call Setup Rules Set ID	-1
	<input type="button" value="Submit"/> <input type="button" value="Cancel"/>	

IP-to-IP Routing Table

▼ IP-to-IP Routing Table

Rule
Action

In			
	Index	<input type="text" value="6"/>	
1	Destination Type	<input type="text" value="IP Group"/>	
2	Destination IP Group ID	<input type="text" value="2"/>	
3	Destination SRD ID	<input type="text" value="2"/>	
5	Destination Address	<input type="text"/>	
6	Destination Port	<input type="text" value="0"/>	
7	Destination Transport Type	<input type="text"/>	
8	Alternative Route Options	<input type="text" value="Route Row"/>	
	Group Policy	<input type="text" value="None"/>	
	Cost Group	<input type="text" value="None"/>	

11. **Answering Machine Detection (AMD)/Call Progress Tone Detection (CPD)**

Note: Support for Answering Machine Detection (AMD) and Call Progress Tone Detection (CPD) on the Mediant 4000, Mediant 9000, and the Mediant Software and Virtual SBC is planned for the AudioCodes 7.0 software release.

AudioCodes Media Gateways support answering machine detection (AMD) as well as call progress tone detection capabilities that can detect whether a human voice or an answering machine is answering the call. This capability is useful for automatic dialing applications.

To enable and configure AMD:

- Using the Media Gateway web interface, open the IPMedia Settings page (**Configuration** tab > **VoIP** > **Media** > **IPMedia Settings**):

IPMedia Settings	
⚡ IPMedia Detectors	Disable
Enable Answer Detector	Disable
Answer Detector Activity Delay	0
Answer Detector Silence Time	10
Answer Detector Redirection	0
Answer Detector Sensitivity	0
Answer Machine Detector Sensitivity Parameter Suit	0
Answer Machine Detector Sensitivity	3
Answer Machine Detector Beep Detection Timeout	200
Answer Machine Detector Beep Detection Sensitivity	0

2. Enable AMD by setting the 'IPMedia Detectors' parameter to **Enable**.
3. Configure the other AMD parameters as required. See below for a description.
4. Click **Submit** then save (burn) the setting to flash memory with a device reset.
5. To enable voice detection once the AMD detects the answering machine, set the *ini* file parameter, EnableVoiceDetection to 1.

The Media Gateway supports up to four AMD parameter suites, where each parameter suite defines the AMD sensitivity levels of detection. The sensitivity levels can range from 0 to 15, depending on the parameter suite. The level is selected using the 'Answer Machine Detector Sensitivity Level' parameter (AMDSensitivityLevel). The parameter suite(s) can be loaded to the device in the Web interface as an auxiliary file or remotely through the ini file using the AMDSensitivityFileName and AMDSensitivityFileUrl parameters.

Additionally AMD can also be configured per call based on the called number or Trunk Group. This is done by configuring AMD for a specific IP Profile and then assigning the IP Profile to a Trunk Group in the Inbound IP Routing table.

The Media Gateway also supports the detection of beeps at the end of an answering machine message. This allows users of third-party,application servers to leave voice messages after an answering machine plays a “beep” sound.

The Media Gateway supports the following methods for detecting and reporting beeps:

- **Using the AMD detector:** This “beep” detector is integrated in the existing AMD feature. The beep detection timeout and beep detection sensitivity are configurable using the AMDBeepDetectionTimeout and AMDBeepDetectionSensitivity parameters, respectively. To enable the AMD beep detection, the X-Detect header in the received SIP INVITE message must include “Request=AMD”, and the AMDBeepDetectionMode parameter must be set to 1 or 2. If set to 1, the beep is detected only after Answering Machine detection. If set to 2, the beep is detected even if the Answering Machine was not detected.
- **Using the Call Progress Tone detector:** To enable this detection mode, the X-Detect header in the received SIP INVITE message must include “Request=CPT”, and one or several beep tones (Tone Type #46) must be configured in the regular CPT file.

The Media Gateway reports beep detection by sending a SIP INFO message containing a body with one of the following values:

- Type=AMD and SubType=Beep
- Type=CPT and SubType=Beep

	<p>Upon AMD activation, the device can send a SIP INFO message to an application server notifying it of one of the following:</p> <ul style="list-style-type: none">• Human voice has been detected• Answering machine has been detected• Silence (i.e., no voice detected) has been detected <p>The detected AMD type (e.g., voice) and success of detecting it correctly are also sent in CDR and Syslog messages.</p>
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