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

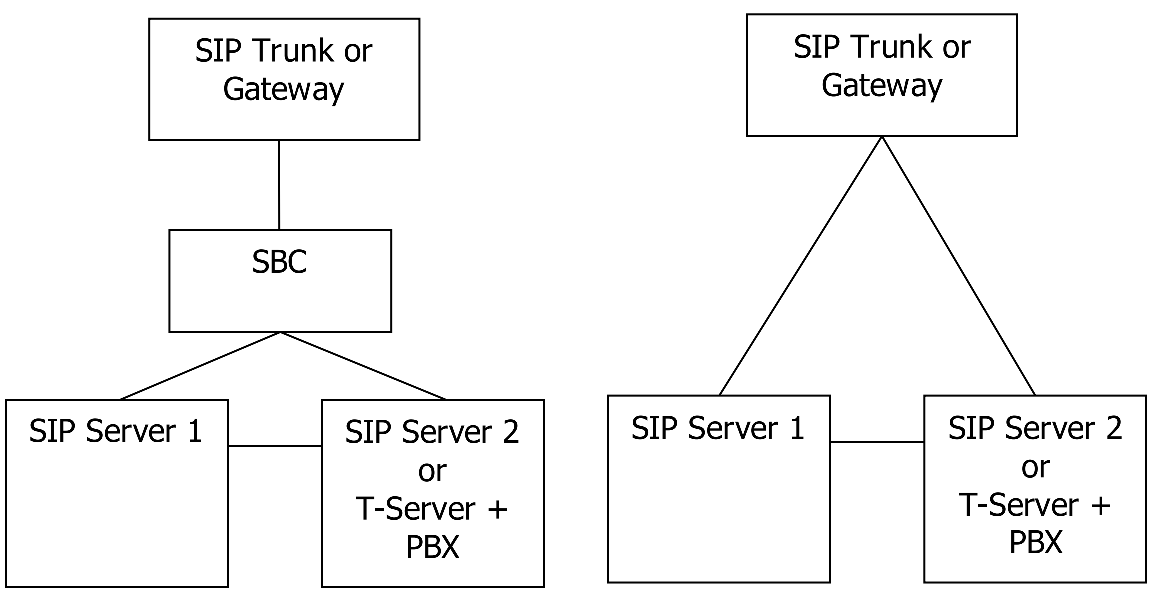
Technical Compatibility – Architecture & SIP Protocol	Description	Supported	Test Cases
"Single Site"	One instance of Genesys SIP Server	Yes	All test cases apply
"Multisite"	Two or more instances of Genesys SIP Server, behind a single Trunk and/or SBC		No "dedicated" test cases
Transfer with re-INVITE	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

3rd 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

<table>
<tr><th colspan="2">Genesys SIP Configuration</th></tr>
<tr><th colspan="2">Features Supported By Gateway/SBC</th></tr>
<tr><th>Feature</th><th>Key Actions and Procedures</th></tr>
<tr><td></td><td>
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>
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=*
sip-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>
</td></tr>
<tr><td>1, 2, 3, 4, 5, 6, 7, 9, 11, 13, 18, 30, 31</td><td>Use the default/standard configuration indicated above</td></tr>
</table>

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).

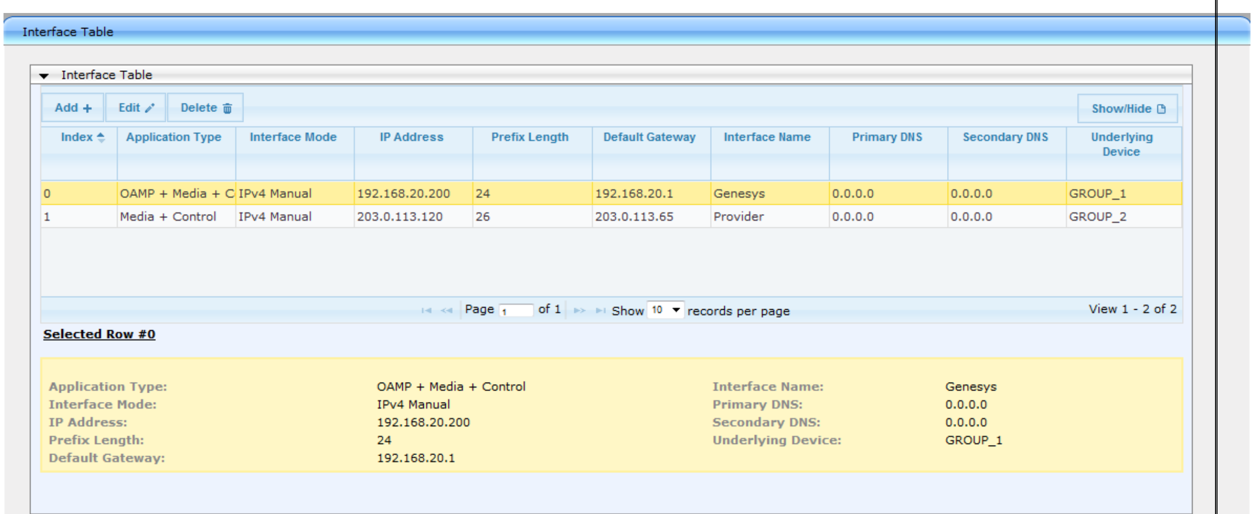
<table>
<tr><th colspan="2">Gateway/SBC Configuration</th></tr>
<tr><th colspan="2">Features Supported By Gateway/SBC</th></tr>
<tr><th>Feature</th><th>Key Actions and Procedures</th></tr>
<tr><td>1, 2, 3, 4, 5, 6, 7, 8, 9, 11, 12, 13, 14, 18, 22, 23, 24, 25, 26, 27, 28, 29, 31</td><td>

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.

1. Network (LAN & WAN)

a. Define the network interface for the Genesys (Trusted or LAN) Network. Reset is required.

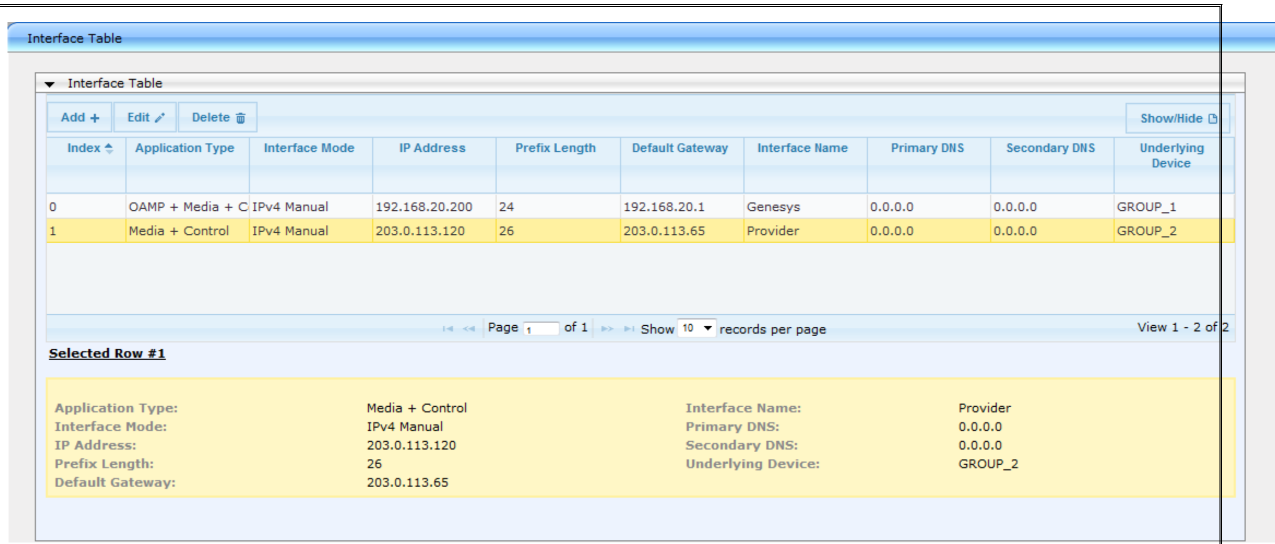
Configuration > VoIP > Network > IP Interfaces Table



Define the network interface for the ITSP (Untrusted or WAN) Network. Reset is required.

Configuration > VoIP > Network > IP Interfaces Table

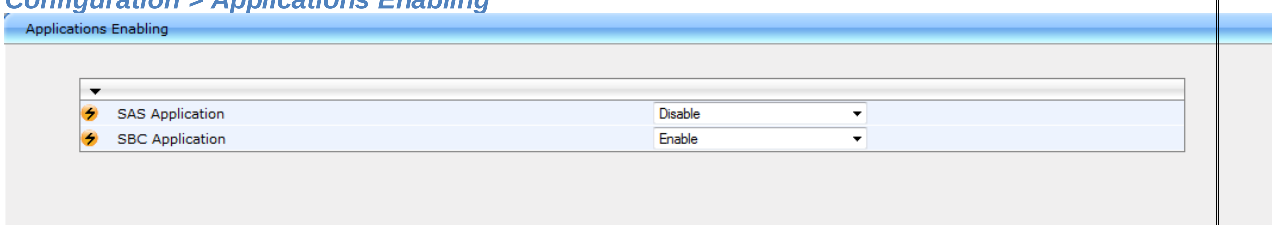
</td></tr>
</table>



2. SBC Application Enabling

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

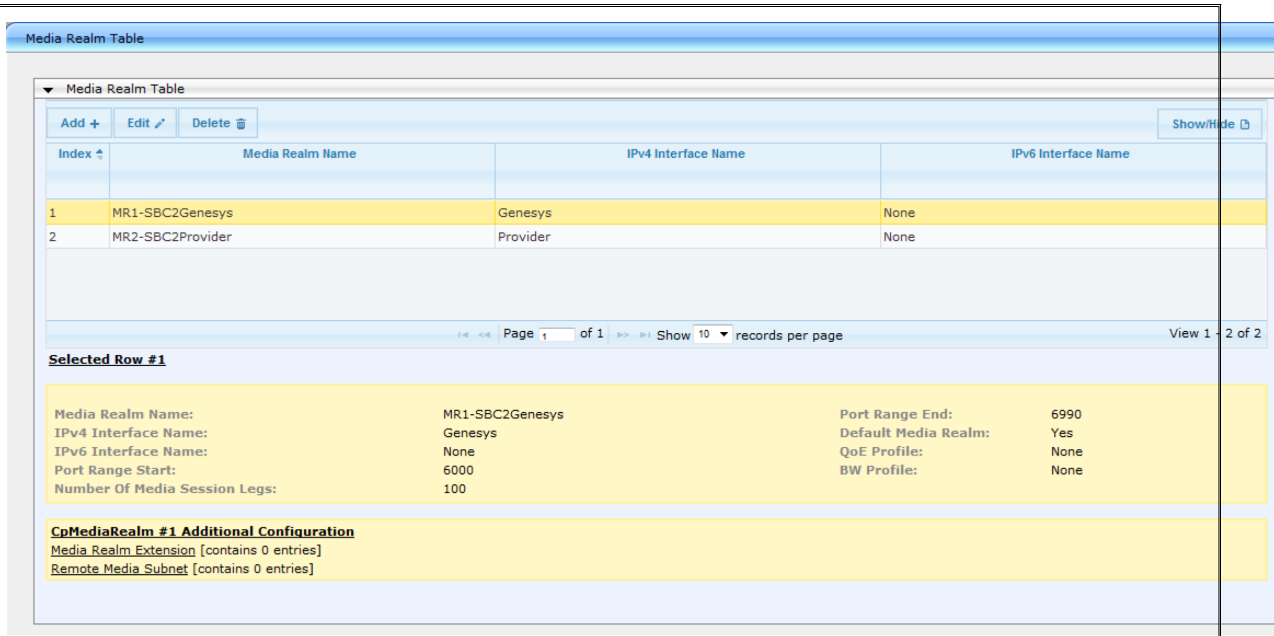
Configuration > Applications Enabling

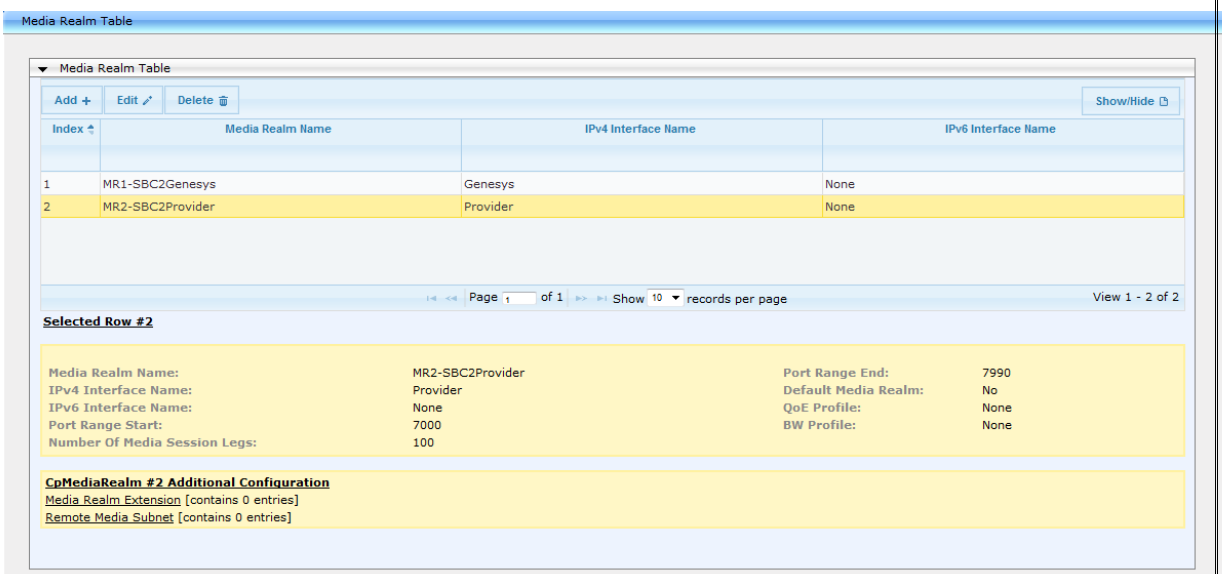


3. Media Realm

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

Configuration > VoIP > Network > Media Realm Table

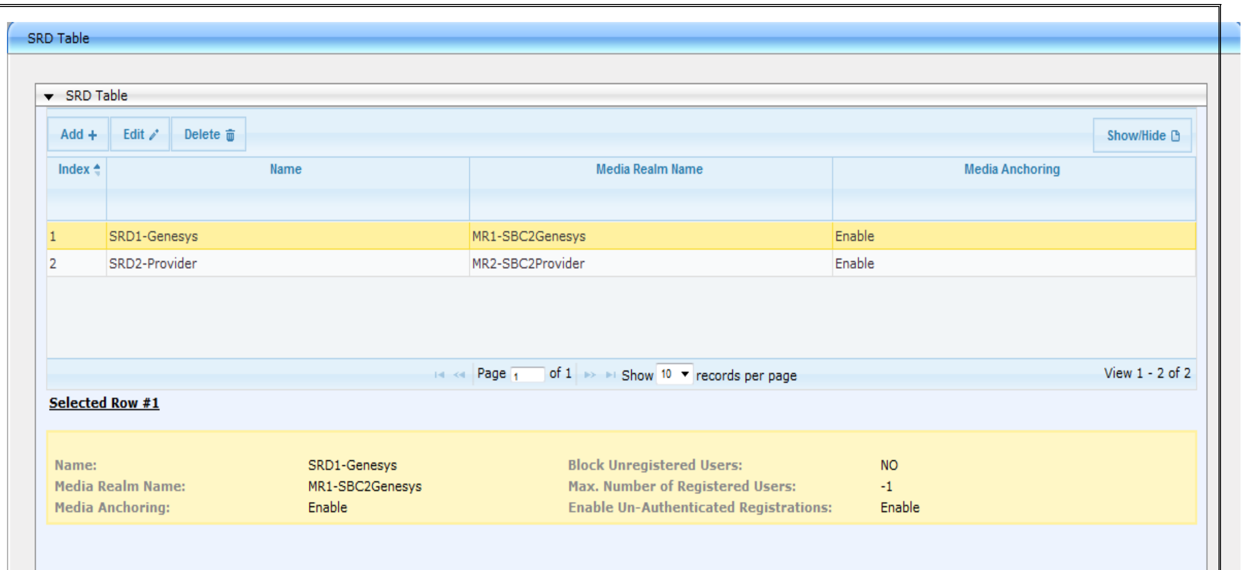


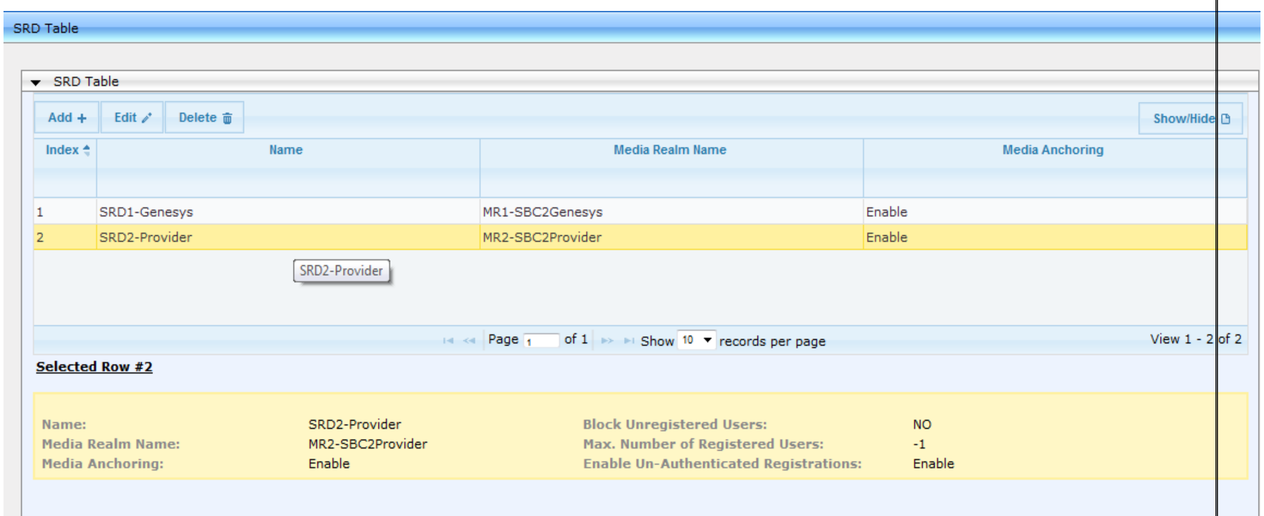


4. SRDs

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

Configuration > VoIP > Network > SRD Table

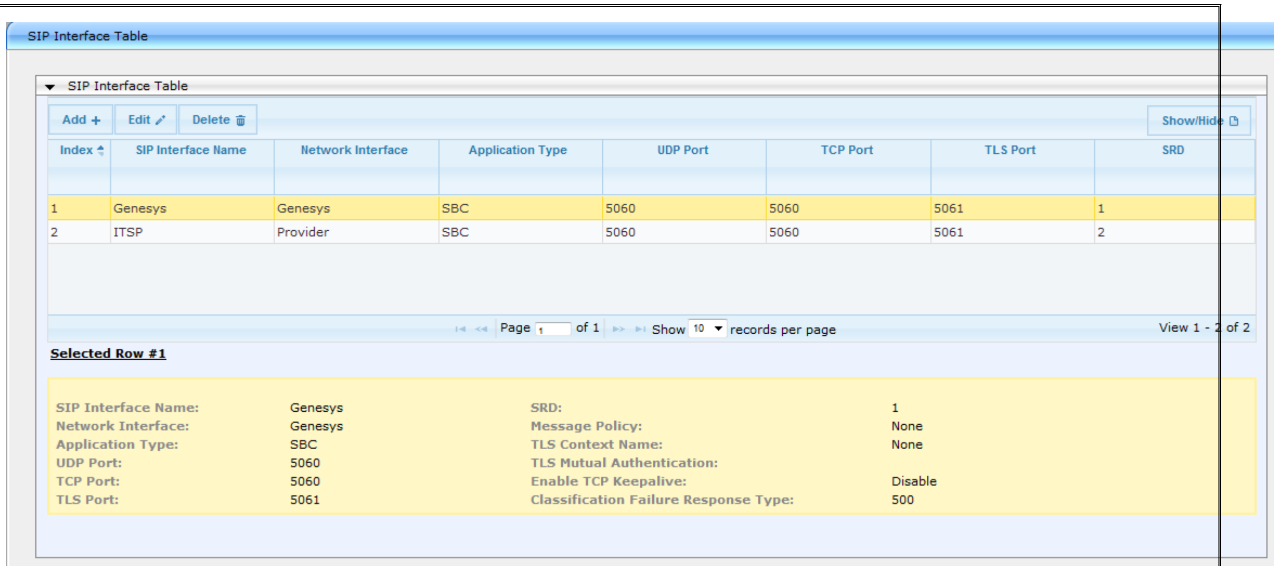


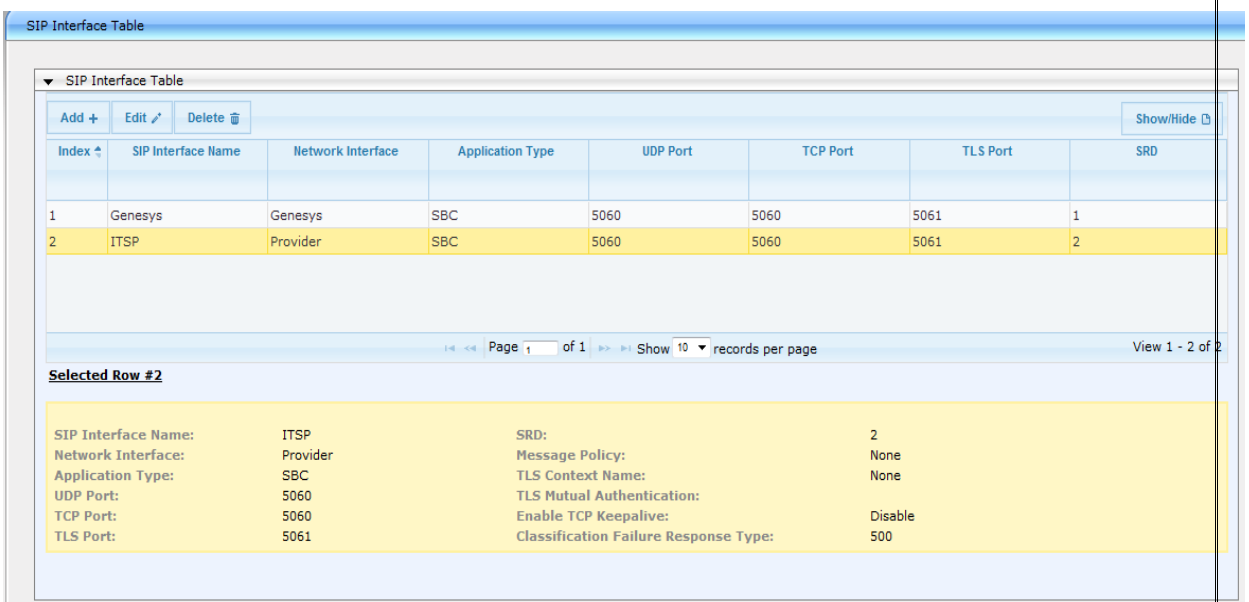


5. SIP Interfaces

Define SIP Interfaces.

Configuration > VoIP > Network > SIP Interface Table

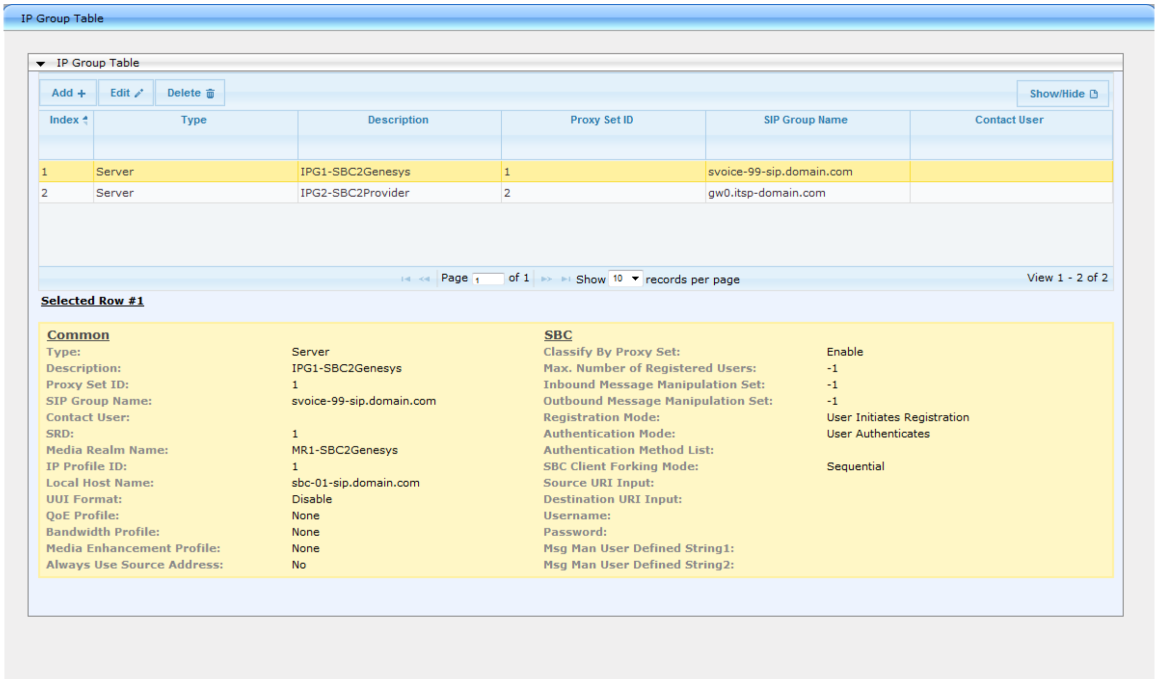


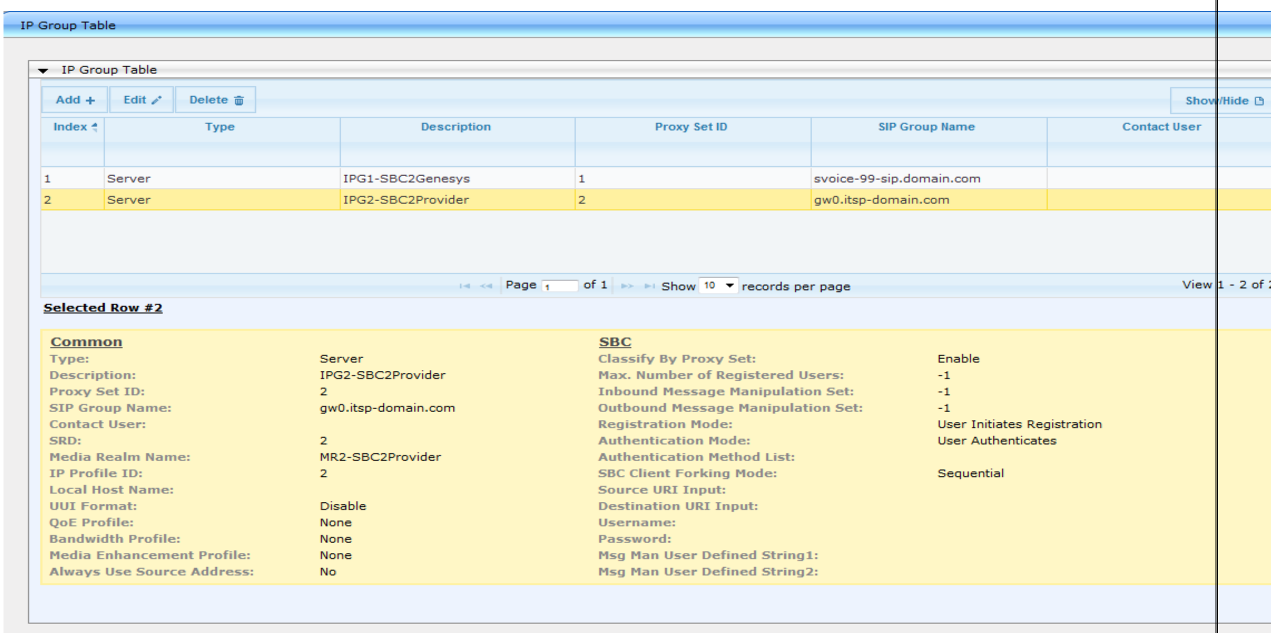


6. IP Groups

Define the supporting IP Groups.

Configuration > VoIP > Network > IP Group Table





7. Proxy Sets

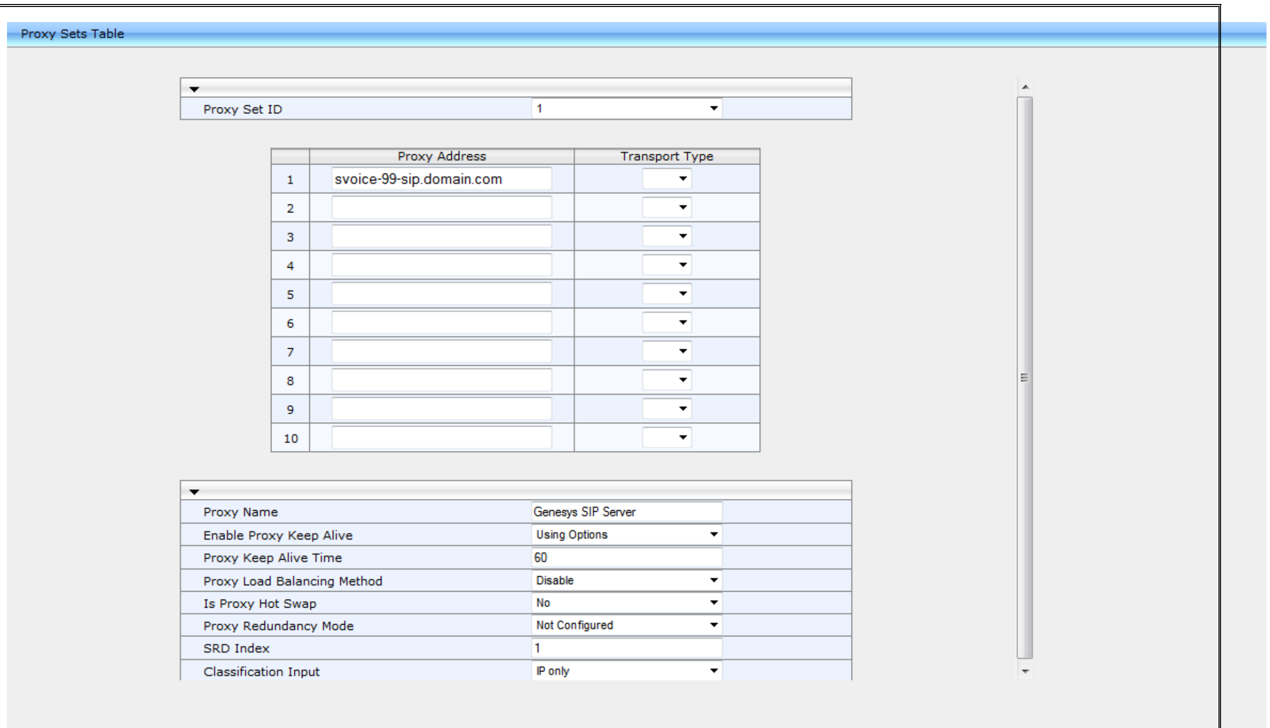
Define Proxy Sets.

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



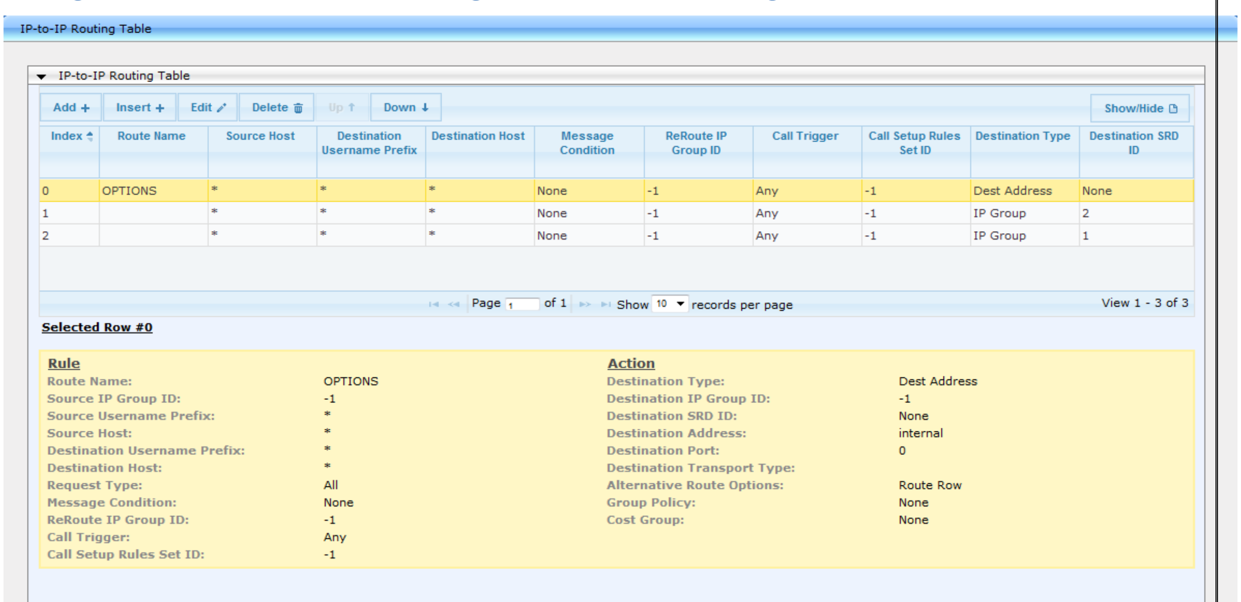
8. IP-to-IP Routing

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

IP-to-IP Routing Table

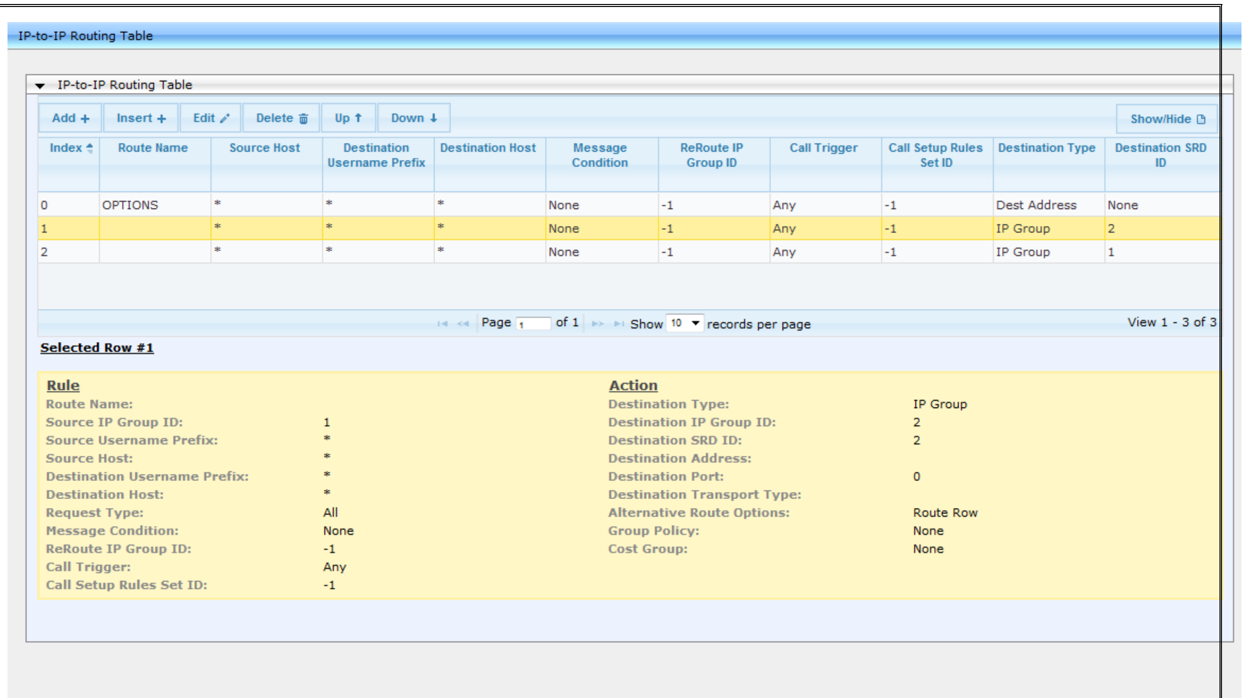
Add + Insert + Edit Delete Up Down Show/Hide

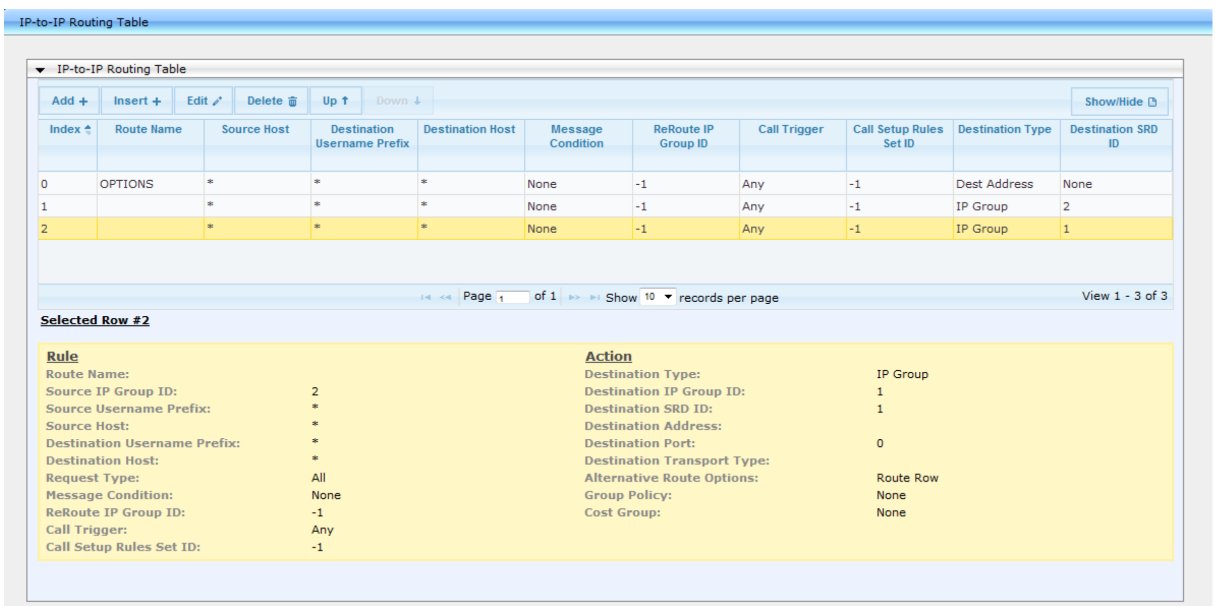
Index	Route Name	Source Host	Destination Username Prefix	Destination Host	Message Condition	ReRoute IP Group ID	Call Trigger	Call Setup Rules Set ID	Destination Type	Destination SRD ID
0	OPTIONS	*	*	*	None	-1	Any	-1	Dest Address	None
1		*	*	*	None	-1	Any	-1	IP Group	2
2		*	*	*	None	-1	Any	-1	IP Group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



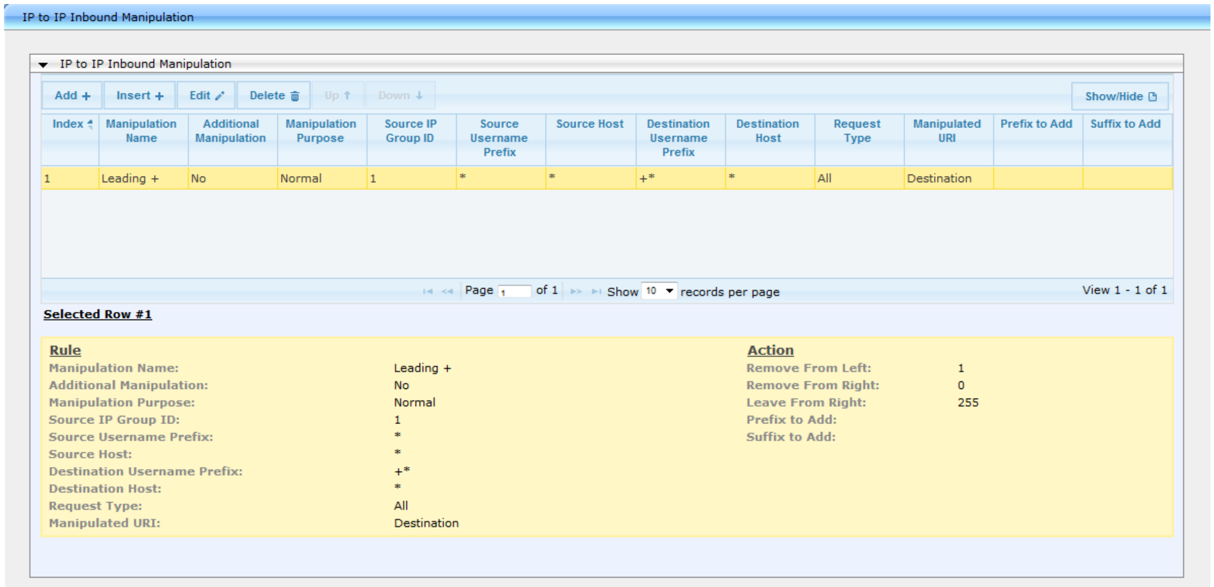


9. Digit/SIP-URI Manipulation

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



10. Message Manipulation

Define any required Message Manipulations.

An explanation for this example:

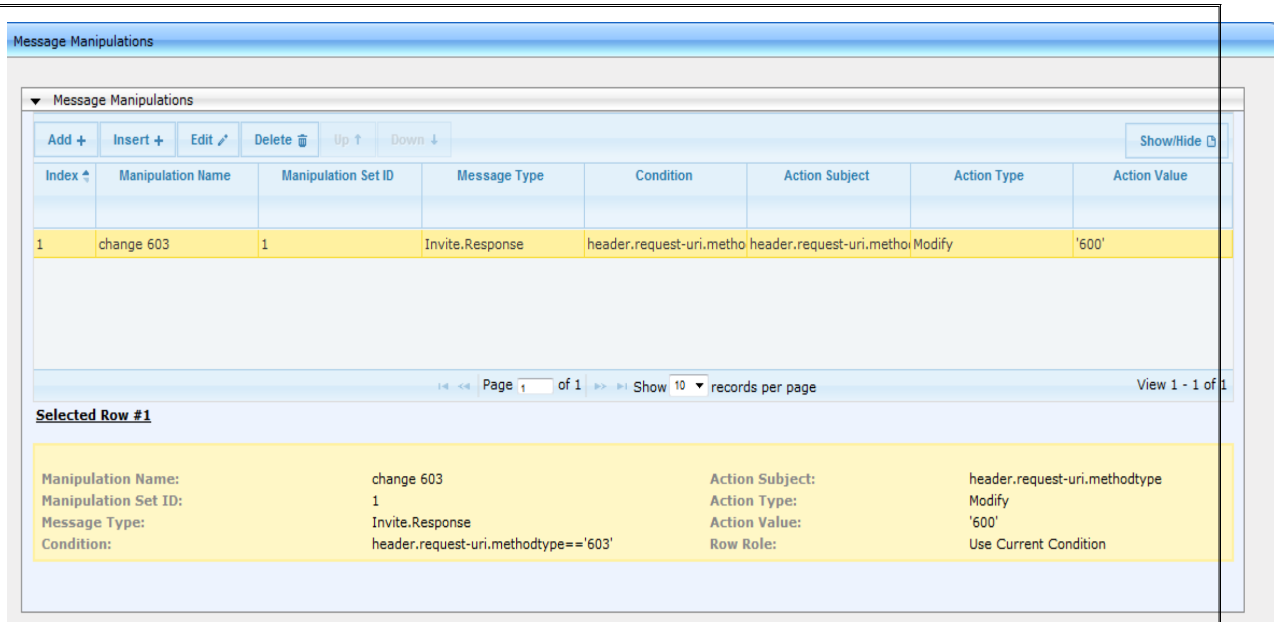
1. The ITSP sends new INVITES when a SIP 603 Declined response is returned on an initial INVITE.
2. The ITSP does not send the new SIP INVITES for SIP 600 Busy Everywhere response.

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

<table>
<tr>
<td></td>
<td>


</td>
</tr>
<tr>
<td>10,15,16,17,19, 20,21,30</td>
<td>

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.

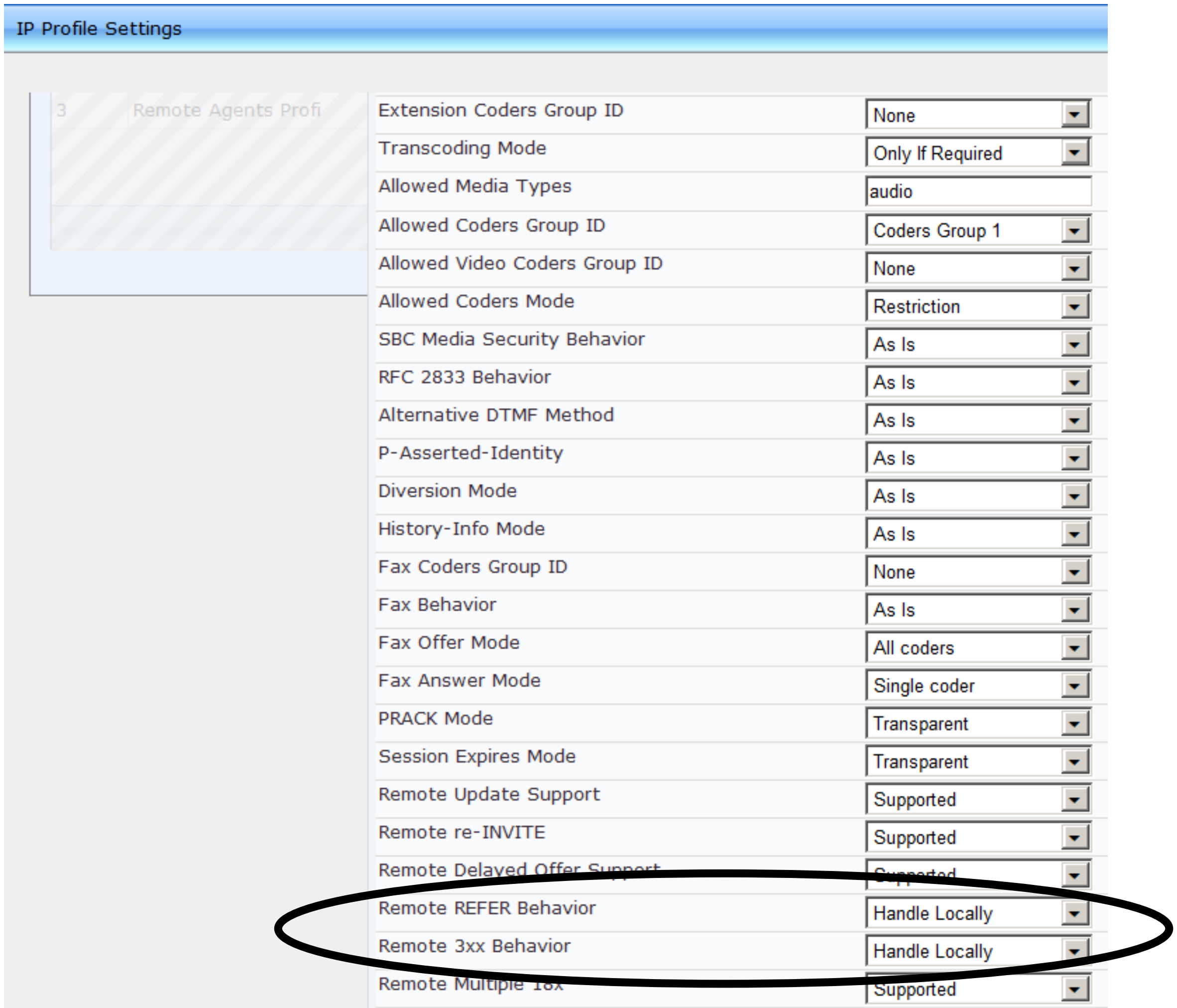
For ITSPs, if SIP REFER isn't handled, they usually do not support proper SIP 3xx behavior and require equal treatment. That means being handled locally by the AudioCodes SBC with a trigger to direct a SIP INVITE be sent to the ITSP and the call anchored on the SBC (but SIP Server is released from the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

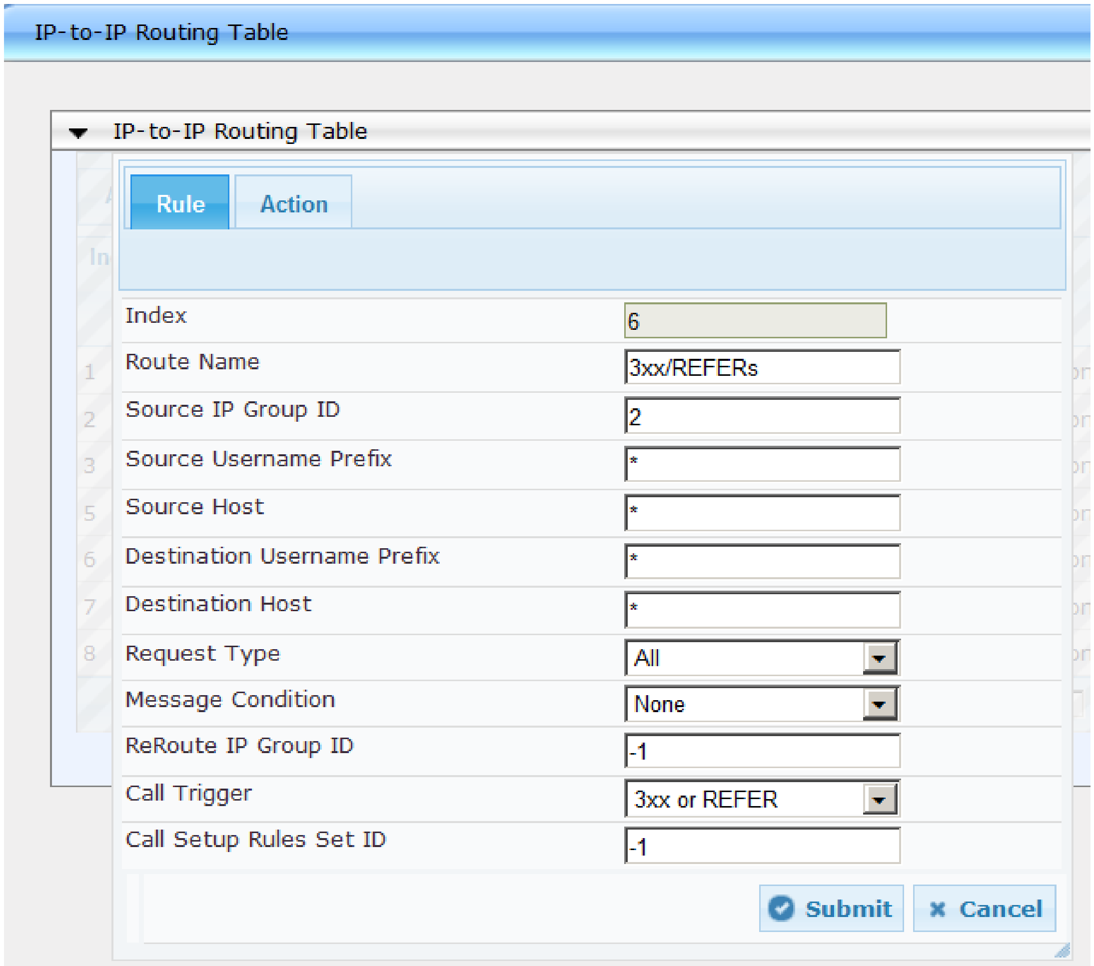
</td>
</tr>
</table>

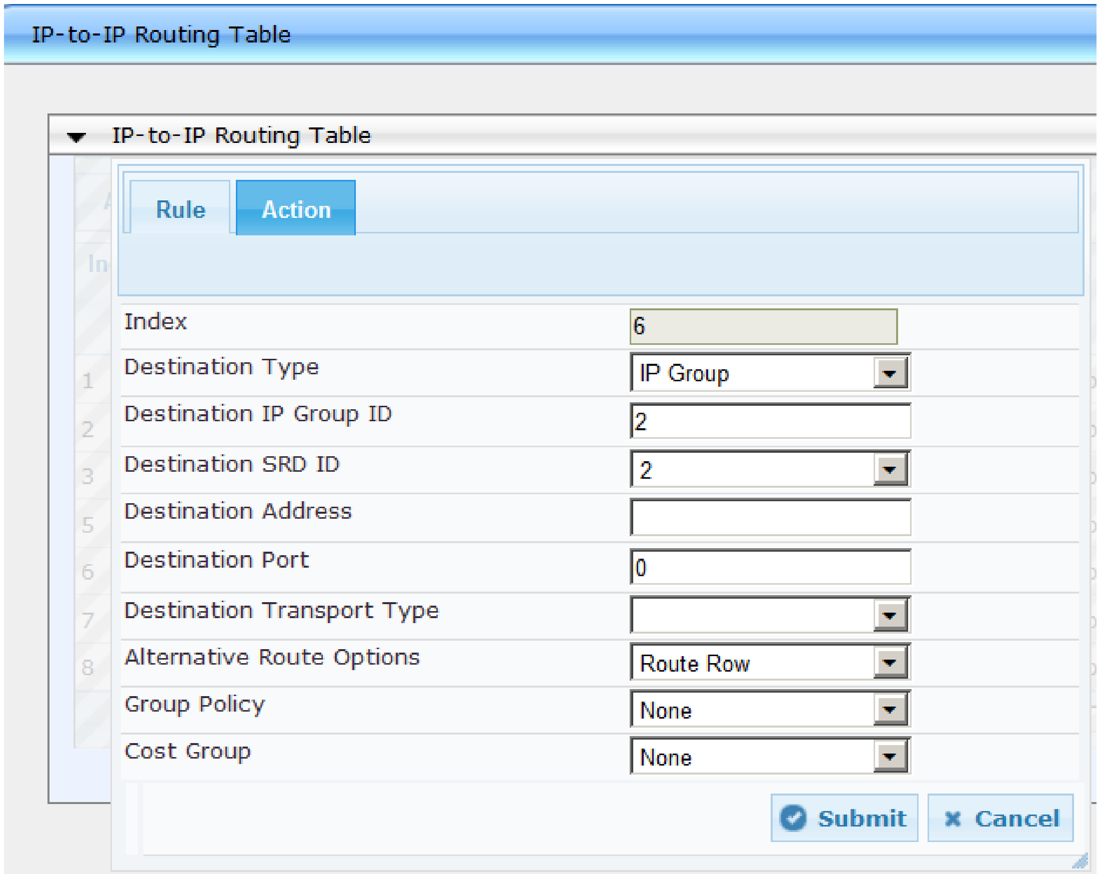


2. IP-to-IP Routing

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
Route Name
3xx/REFERs
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
3xx or REFER
Call Setup Rules Set ID
-1
Submit
Cancel



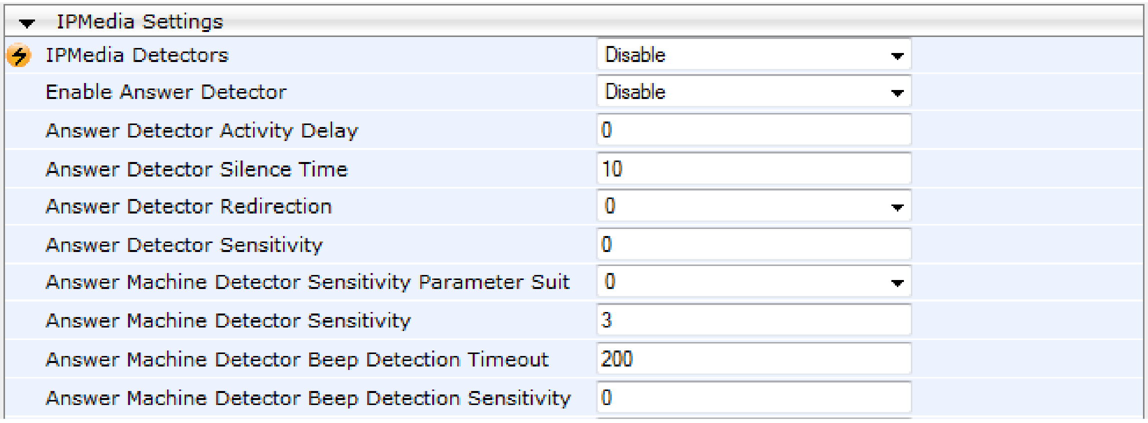
11. Answering Machine Detection (AMD)/Call Progress Tone Detection (CPD)

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:

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



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

	Upon AMD activation, the device can send a SIP INFO message to an application server notifying it of one of the following: • Human voice has been detected • Answering machine has been detected • Silence (i.e., no voice detected) has been detected The detected AMD type (e.g., voice) and success of detecting it correctly are also sent in CDR and Syslog messages.