Avaya Solution & Interoperability Test Lab

Application Notes for Configuring a SIP Trunk between AudioCodes Mediant 1000 MSBG e-SBC and Avaya IP Office - Issue 1.0

Abstract

These Application Notes describe the steps to configure a SIP trunk between AudioCodes Mediant 1000 MSBG e-SBC and Avaya IP Office.

The AudioCodes Mediant 1000 MSBG e-SBC is a SIP Session Border Controller (SBC) that manages and protects the flow of SIP signaling and related media across an untrusted IP network. The compliance testing focused on telephony scenarios between an enterprise site, where the AudioCodes Mediant 1000 MSBG e-SBC and Avaya IP Office were located, and a second site simulating a service provider service node.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe a sample configuration for a network consisting of Avaya IP Office connected to an AudioCodes Mediant 1000 MSBG e-SBC e-SBC via a SIP trunks.

The Mediant 1000 MSBG e-SBC is an all-in-one multi-service access solution for Service Providers offering managed services and distributed Enterprises. This multi-service business gateway is designed to provide converged Voice & Data services for business customers at wire speed, while maintaining SLA parameters for voice quality.

The compliance testing focused on telephony scenarios between an enterprise site, where the AudioCodes Mediant 1000 MSBG e-SBC e-SBC and Avaya IP Office were located, and a second site simulating a service provider service node.

2. General Test Approach and Test Results

The general test approach was to make calls between the main enterprise site and the 2nd site simulating a service provider service node using various codec settings and exercising common telephony features.

2.1. Interoperability Compliance Testing

The compliance testing focused on interoperability between AudioCodes Mediant 1000 MSBG e-SBC and Avaya IP Office by making calls between the enterprise site and a second site simulating a service provider service node that were connected through the Mediant 1000 MSBG e-SBC using direct SIP trunks. The following functions and features were tested:

- Calls from both SIP and non-SIP endpoints between sites.
- G.711 μLAW, G.711ALAW, and G.729(a) codec support.
- Proper recognition of DTMF transmissions by navigating voicemail menus.
- Proper operation of voicemail with message waiting indicators (MWI).
- Telephony features including Multiple Call Appearances, Hold, Transfer, Conference, Call Forwarding, Call Park, Call Pickup, and Send All Calls.
- Inbound and outbound fax calls.
- Calls using IP Office Softphone.
- Proper system recovery after a Mediant 1000 MSBG e-SBC restart and/or re-establishment of broken IP connectivity.

2.2. Test Results

The AudioCodes Mediant 1000 MSBG e-SBC passed compliance testing.

2.3. Support

For technical support on the AudioCodes Mediant 1000 MSBG e-SBC, visit their online support at http://www.audiocodes.com/support.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows two sites connected via a SIP trunk across an untrusted IP network: the main enterprise site and a second site that simulates a service provider service node. The AudioCodes Mediant 1000 MSBG e-SBC Session Border Controller (SBC) is at the edge of the main site. The public side of the Mediant 1000 MSBG e-SBC is connected to the untrusted network and the private side is connected to the trusted corporate LAN.

All SIP traffic between two sites flows through the Mediant 1000 MSBG e-SBC. In this manner, the Mediant 1000 MSBG e-SBC can protect the main site's infrastructure from any SIP-based attacks. The voice communication across the untrusted network uses SIP over TCP and RTP for the media streams.

IP Office connects to the Mediant 1000 MSBG e-SBC using a SIP trunk. IP Office endpoints include both 3rd party SIP and Avaya non-SIP endpoints.

The 2nd site (shown as a cloud), simulates a service provider service node, and comprises of a Communication Manager, System Manager, and Session Manager, with both SIP and non-SIP endpoints.

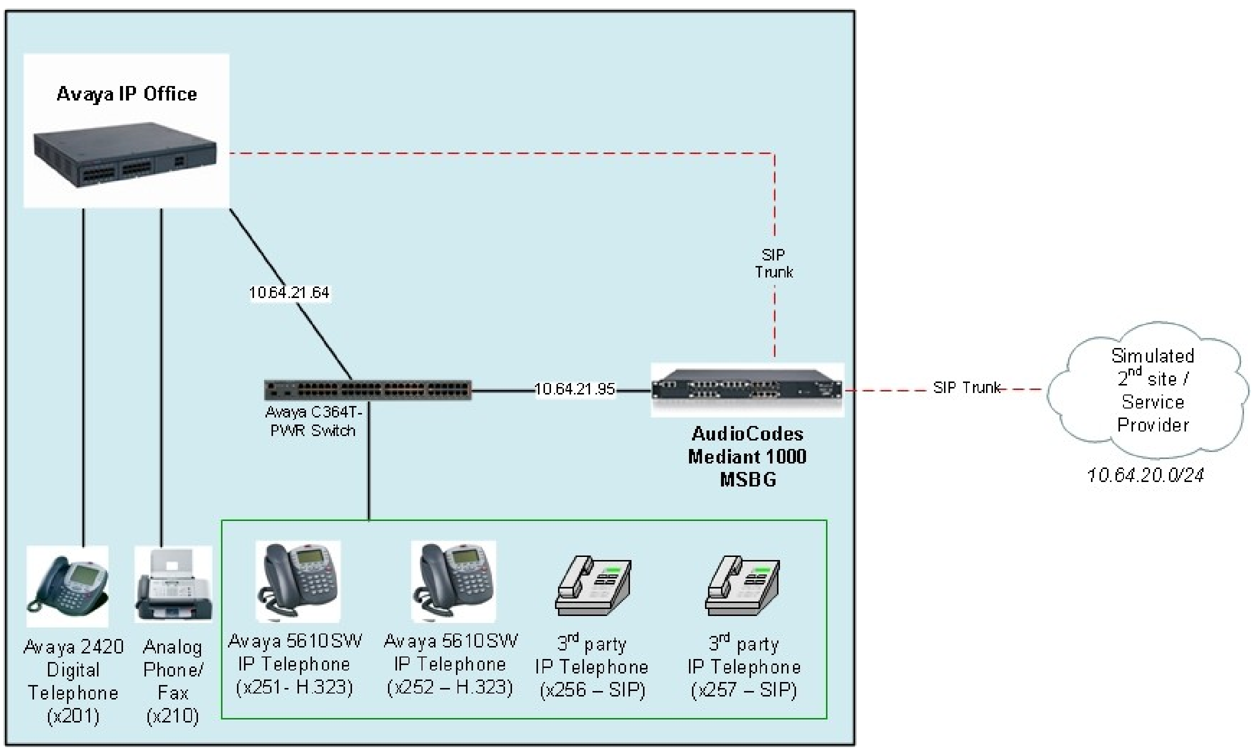


Figure 1: AudioCodes Mediant 1000 MSBG e-SBC SIP Trunking Test Configuration

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	Software
Avaya IP Office 500	6.1 (5)
Avaya IP Office 500 Phone Expansion Module Analog POTS 30 V2	6.1 (5)
Avaya IP Office Manager (Windows PC)	8.1 (5)
Avaya 5610SW IP Telephone (H.323)	2.9.1
Avaya 2420 Digital Telephone	R6 Firmware
Avaya 5420 Digital Phone	R6 Firmware
Avaya 6210 Analog Telephone	n/a
Avaya IP Office Softphone	3.0
Fax Machine	-
AudioCodes Mediant 1000 MSBG e-SBC	6.2

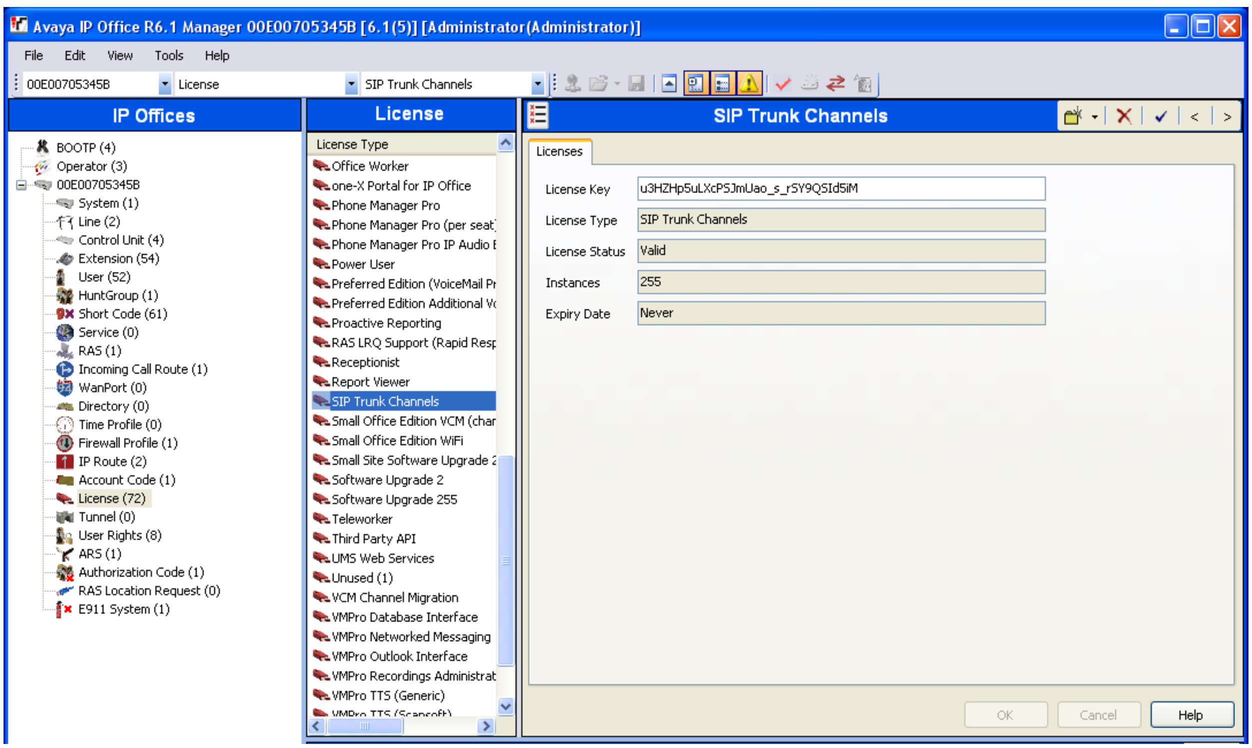
5. Configure Avaya IP Office

This section describes the steps required for configuring a static SIP trunk on IP Office.

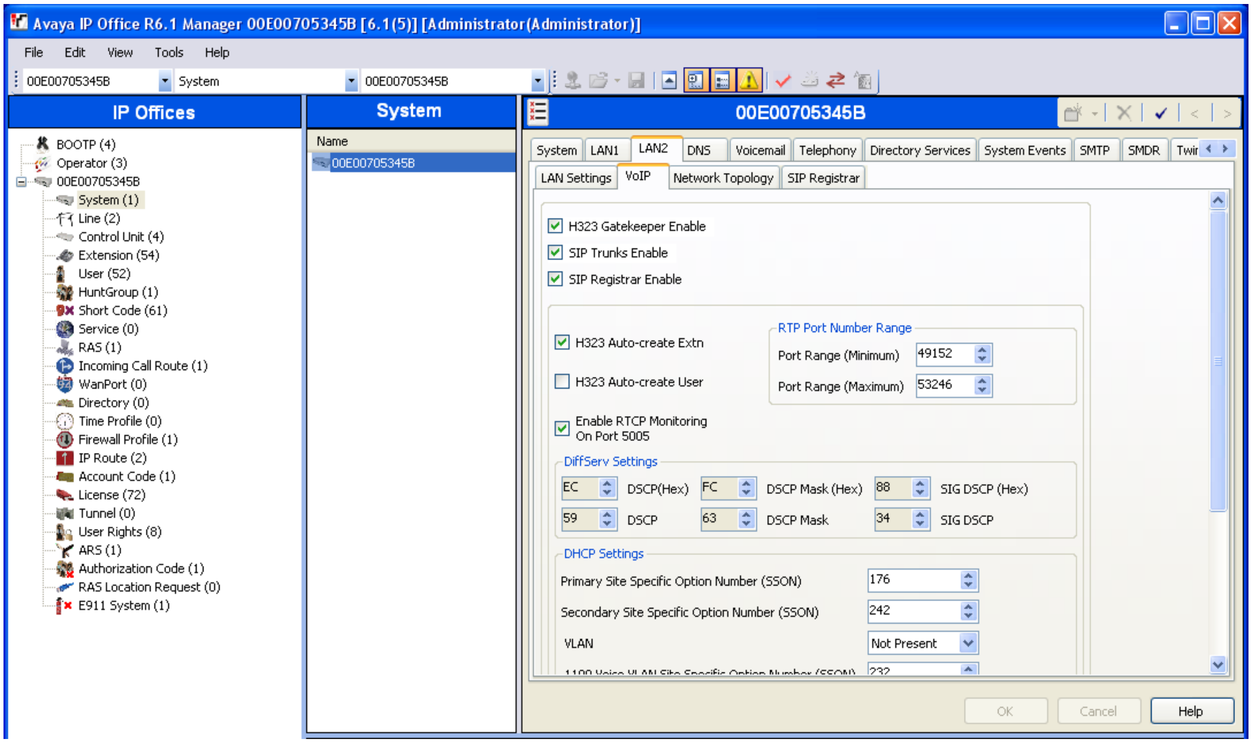
IP Office is configured via the IP Office Manager program. Log into the IP Office Manager PC and select **Start→All Programs→IP Office→Manager** to launch the Manager application. Log into the Manager application using the appropriate credentials.

1. *Verify the SIP Trunk Channels License.*
   Click on **License** in the left panel. Confirm that there is a valid **SIP Trunk Channels** entry.

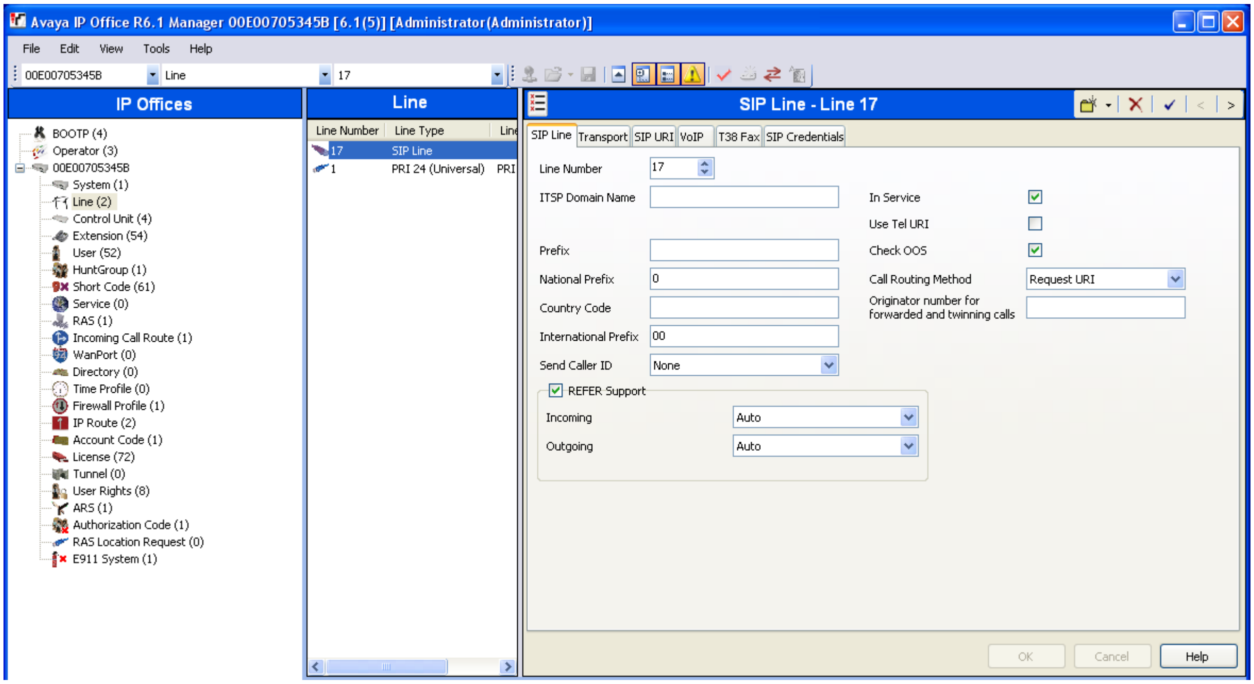
   If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.



2. *Enable SIP Trunks*.
   Select **System** in the left panel. Click the **LAN2** tab. Under the **LAN2** tab, select the **VOIP** tab, and check the **SIP Trunks Enable** box. Click the **OK** button.



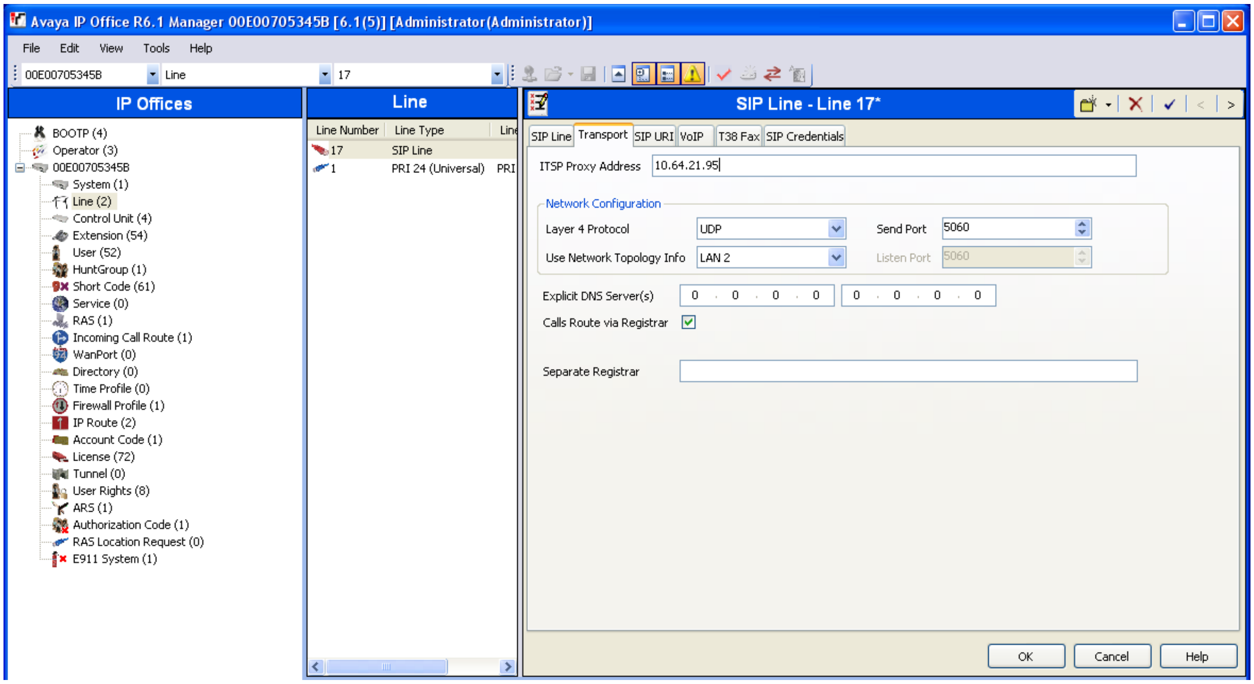
3. *Create a SIP Line.*
   Select **Line** in the left panel. Right-click and select **New→SIP Line**. The screen appears. Note the **Line Number**, or select another unused line number.



4. *Configure Transport parameters for the SIP Line*.
Select the **Transport** tab.

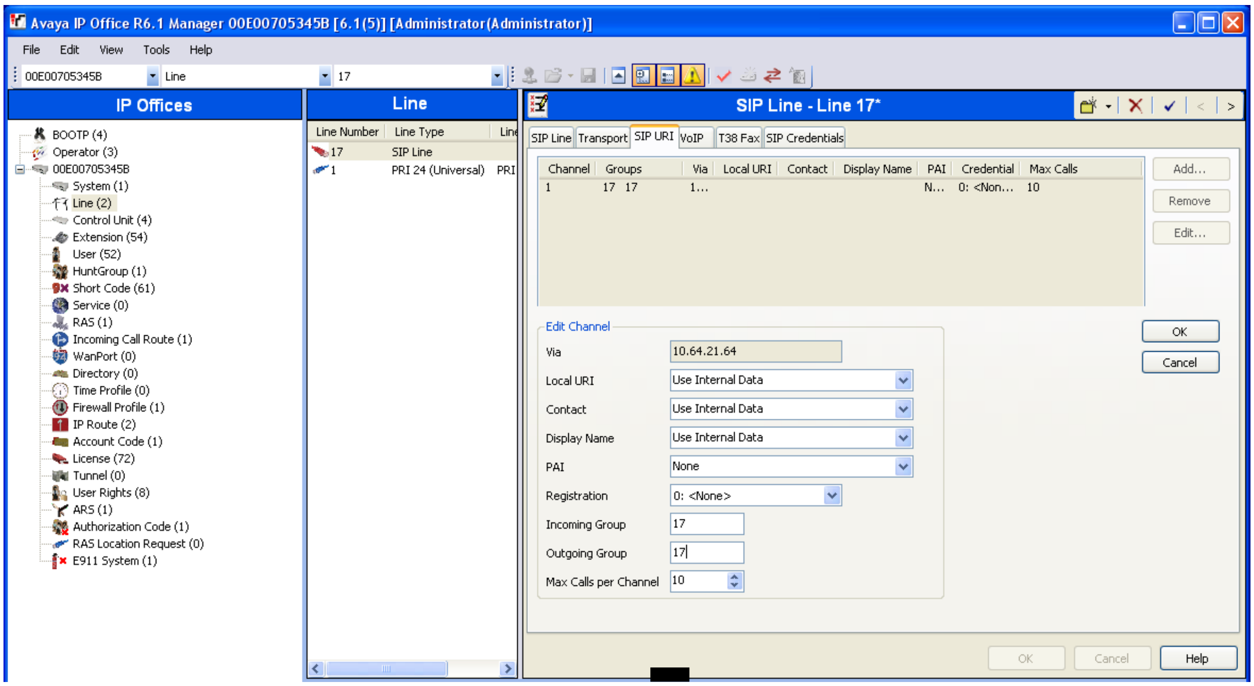
Configure the following:

- For **ITSP Proxy Address**, enter the IP address for AudioCodes Mediant 1000 MSBG e-SBC.
- For **Layer 4 Protocol** and **Send Port**, select **UDP** and **5060** respectively.
- For **Use Network Topology Info**, select **LAN 2**.
- Use default values for other fields.



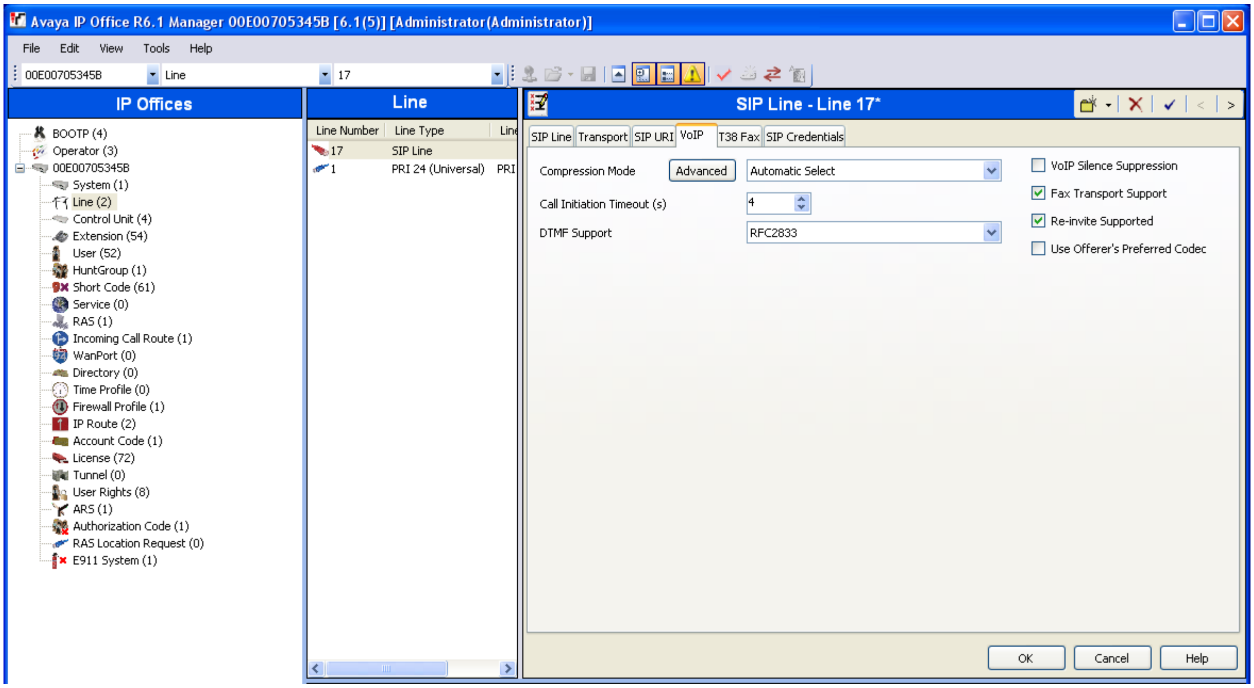
5. *Configure SIP URI parameters for the SIP Line*.
   Select the **SIP URI** tab. Click the **Add** button.

   Select **Use Internal Data** for the **Local URI**, **Contact**, and **Display Name** fields. This tells the system to use the information configured on the **SIP** tab for each individual **User** configured (not shown). Enter unused numbers for the **Incoming Group** and **Outgoing Group** fields. The **Incoming Group** field will be used for mapping inbound calls from the SIP trunk to local stations. The **Outgoing Group** will be used for routing calls externally via the Short Code configured in **Step 9**. Use default values for all other fields. Click the **OK** button.



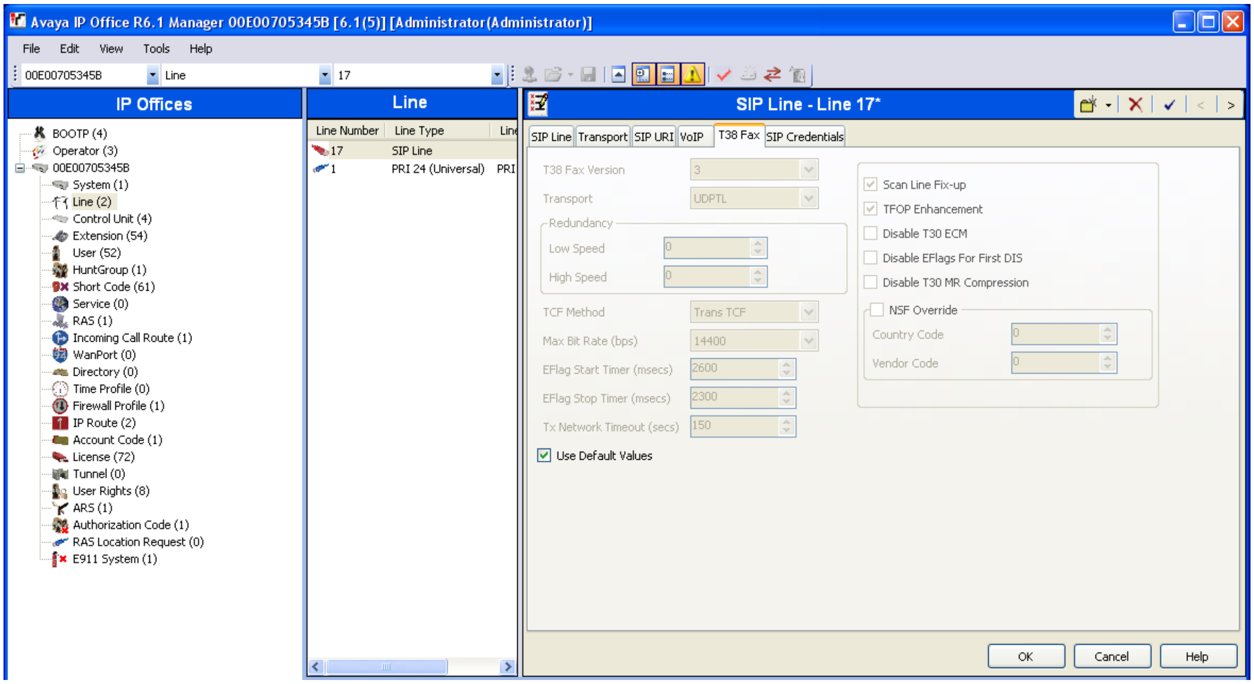
6. *Configure VoIP parameters for the SIP Line.*
   Select the **VoIP URI** tab. Click the **Add** button.

   For **Compression Mode**, select **Automatic Select** or the desired codec from the drop-down list. Check the **Fax Transport Support** and **Re-invite Supported** check boxes. Click the **OK** button.



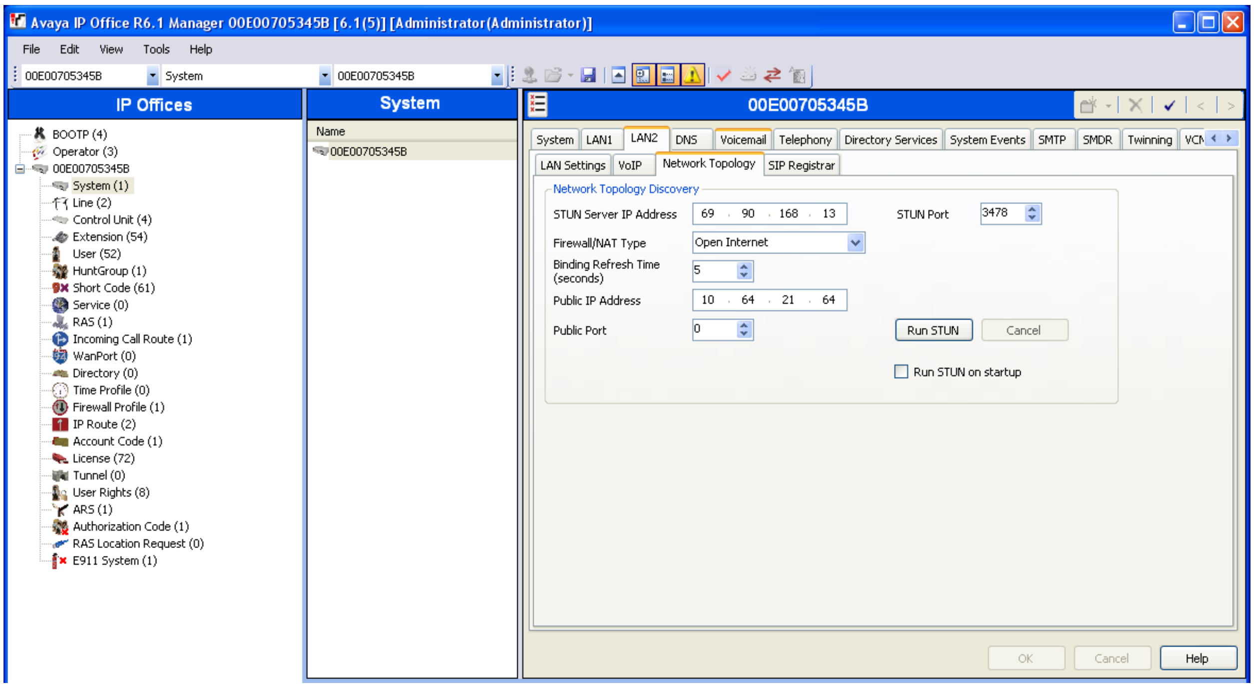
7. *Configure T38 Fax parameters for the SIP Line.*
   Select the **T38 Fax** tab.

   Check the **Use Default Values** check box. Click the **OK** button.



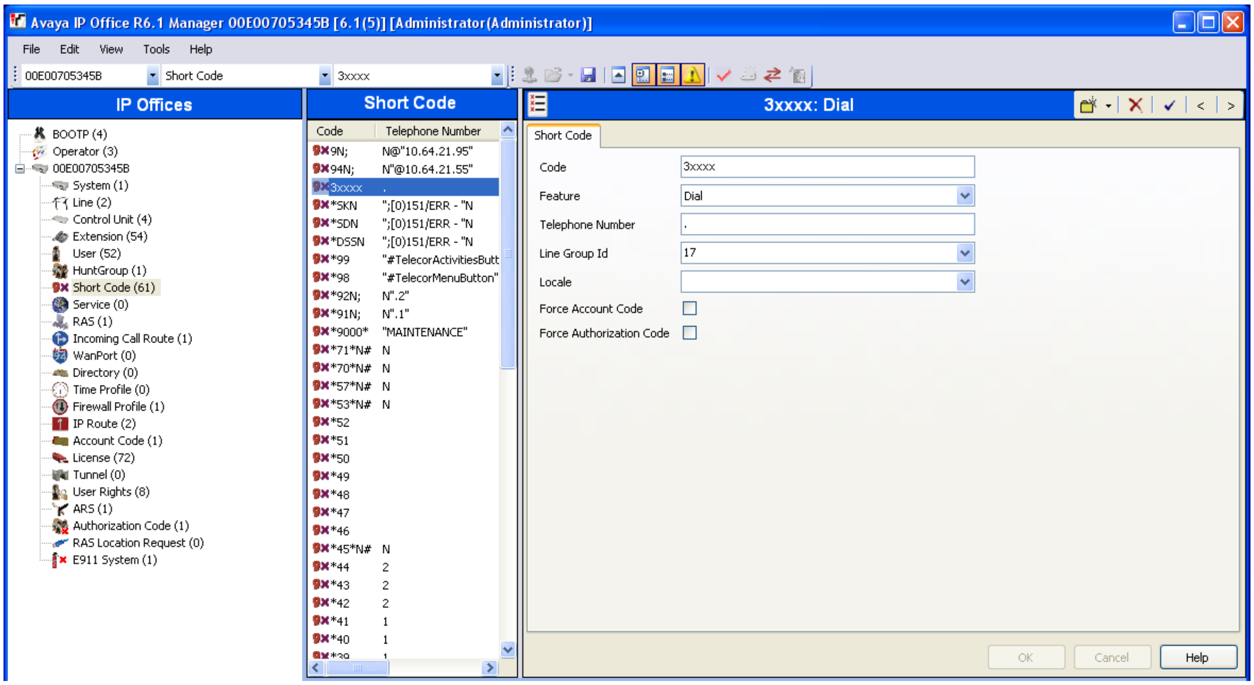
8. *Configure SIP OPTIONS timer for "keep alive" function.*
   Select **System** in the left panel. Under the **LAN2** tab, select the **Network Topology** tab.

   Set the **Binding Refresh Time** to the desired interval which determines the frequency with which OPTIONS messages will be sent to AudioCodes. For **Public IP Address**, enter the Avaya IP Office system IP address. Accept the default values for all other fields. Click the **OK** button.



9. *Configure a short code to route calls to AudioCodes.*
   Select **Short Code** in the left panel. Right click and select **New**.

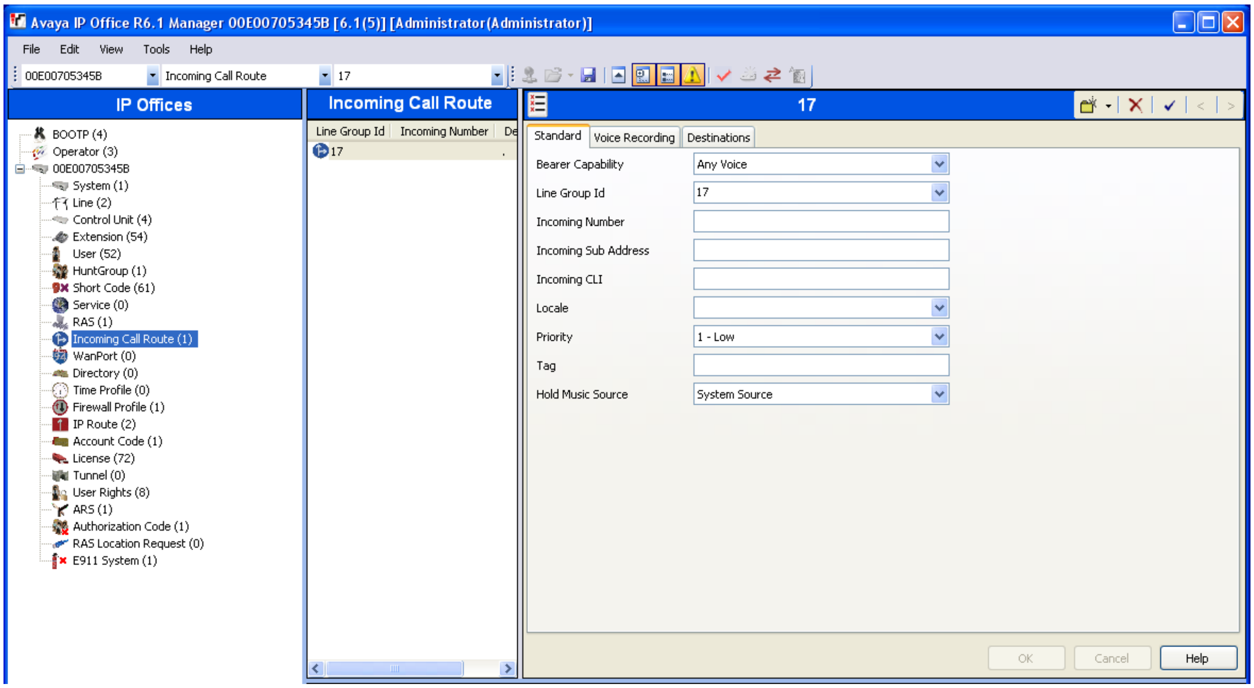
   During compliance testing, **3xxxx** was entered for the **Code** field. All calls to a 5 digit extension, beginning with a 3, were routed to AudioCodes for further routing. Select **Dial** for the **Feature** field. Enter the period symbol "**.**" for the **Telephone Number** field. Enter the **Outgoing Group Id** created in **Step 5** for the **Line Group Id** field. Use default values for all other fields. Click the **OK** button.



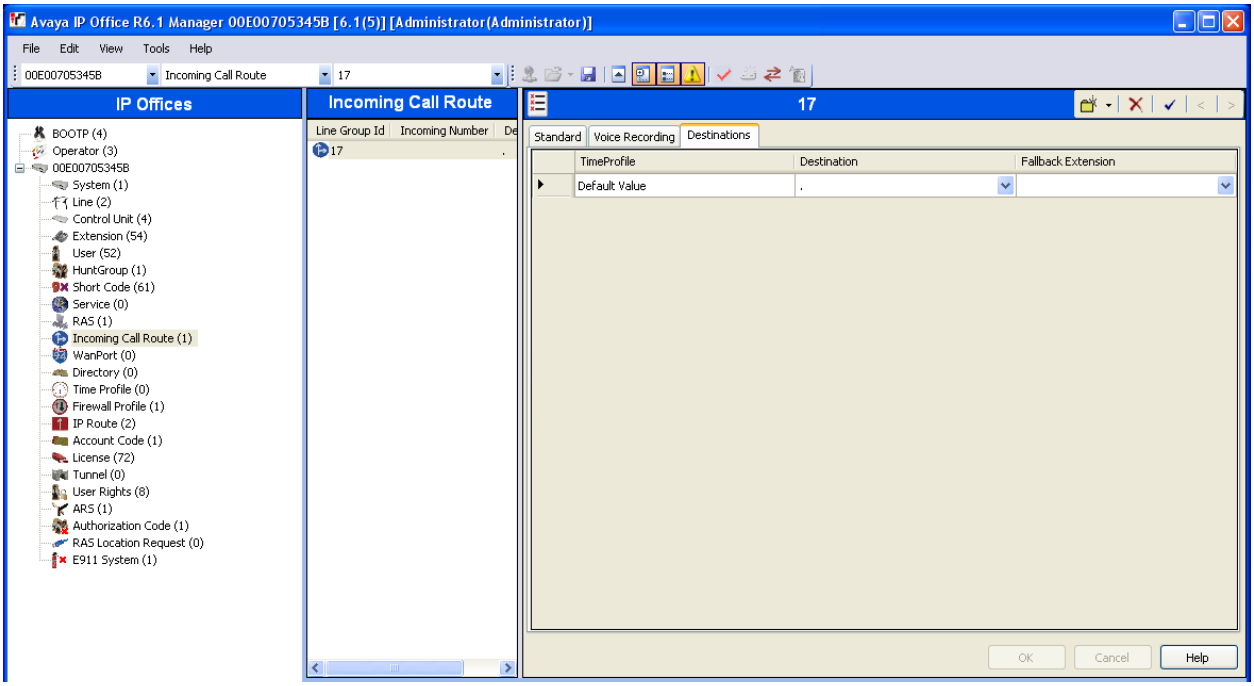
10. *Create an Incoming Call Route for the Inbound SIP calls*.
Select **Incoming Call Route** in the left panel. Right-click and select **New**.

Enter the following:

- **Any Voice** for the **Bearer Capability** field.
- The Incoming Group created for the URI in **Step 5** in the **Line Group Id** field.
- Use default values for all other fields.



- Next, navigate to the **Destinations** tab and enter the period symbol "**.**" for the **Destination** field.
- Click the **OK** button.



11. After making the changes, click on the floppy disk icon to push the changes to the IP Office system and have them take effect. **Changes will not take effect until this step is completed. ** NOTE ** This may cause a reboot of Avaya IP Office resulting in service disruption**.

6. Configure AudioCodes Mediant 1000 MSBG e-SBC

This section provides the procedures for configuring AudioCodes Mediant 1000 MSBG e-SBC. It is assumed that proper knowledge of the AudioCodes MSBG e-SBC usage, configuration, support in general is understood, and the craft person has experience with the product platform. The following information is derived from the product manuals and is referenced only as a general guide. Configuration of the e-SBC will vary for each specific customer environment; however, AudioCodes has provided screenshots (and called-out specific fields on each screen with "arrows"), to show the configuration used during compliance testing.

All of the configuration shown in this section can be completed using the AudioCodes Mediant 1000 MSBG e-SBC web interface. From a browser, enter the IP address of the e-SBC and log in with the appropriate credentials.

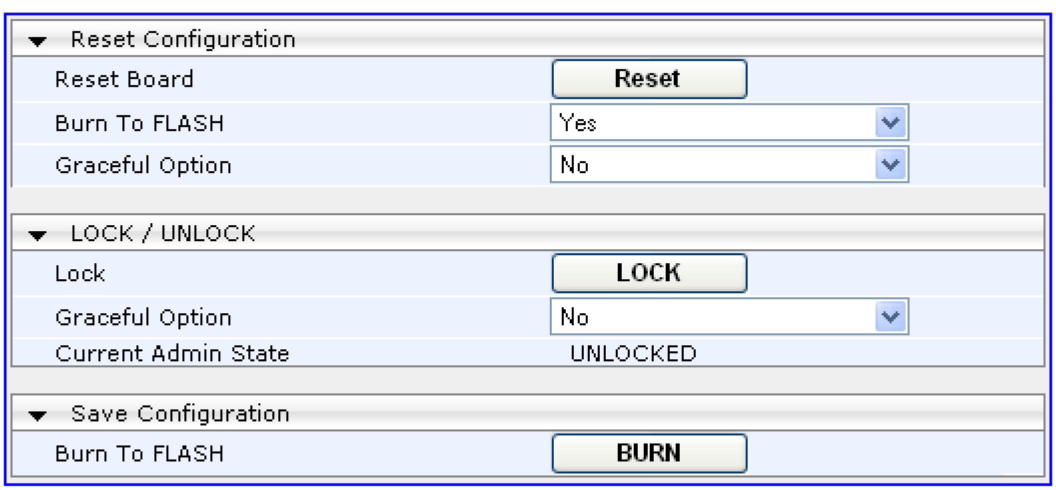
6.1. Configure Data and IP Routing Network Parameters

Ensure the IP Data Routing is set properly for support of routing for each network that is intended to interwork (these details are not shown, but they can be found in the installation manual with regards to the WAN interface setting and routing, as well as LAN side settings).

Once the administration is completed for the data segment, submit, Burn to Flash, and restart the device. Navigate to the **Maintenance Actions** page (**Management** tab > **Management Configuration** menu > **Maintenance Actions**).

- Under the **Reset Configuration** group, from the **Burn To FLASH** drop-down list, select **Yes**, and then click the **Reset** button. The Burn to flash will save the configuration and will allow the unit to recover from future resets in the configuration saved.

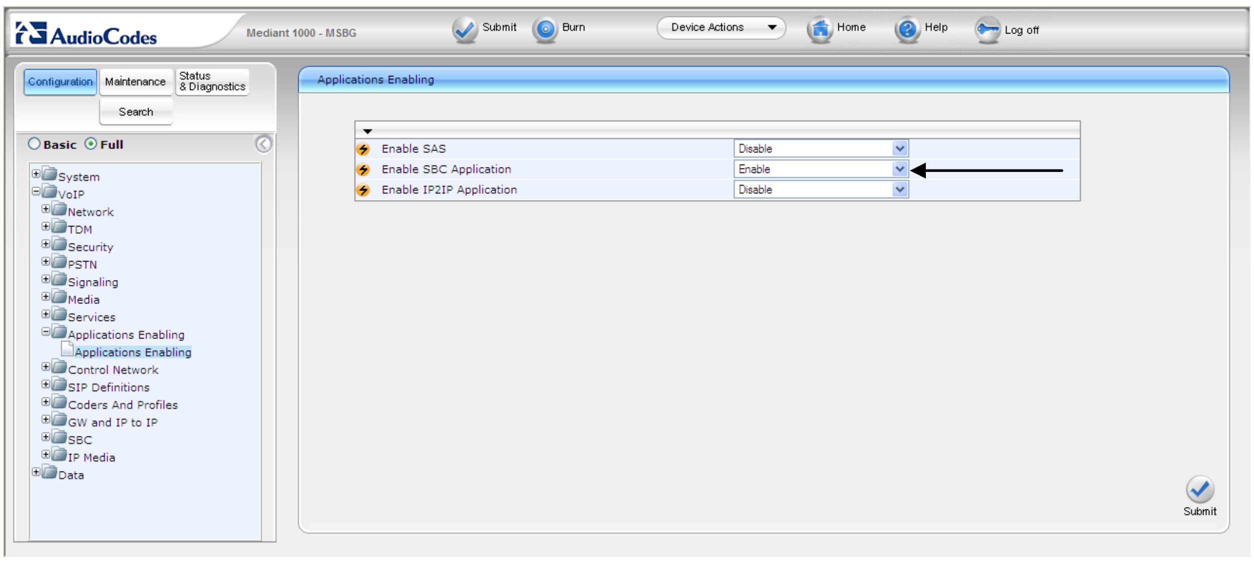
The device's new configuration (i.e., global IP address) is saved (burned) to the flash memory and the device performs a reset. The Web interface session terminates, as no longer accessible using the blade's private IP address.



6.2. Enable SBC functionality

Open the **Applications** page (**Configuration** tab > **VoIP** menu > **Applications Enabling**) to configure the SBC functionality.

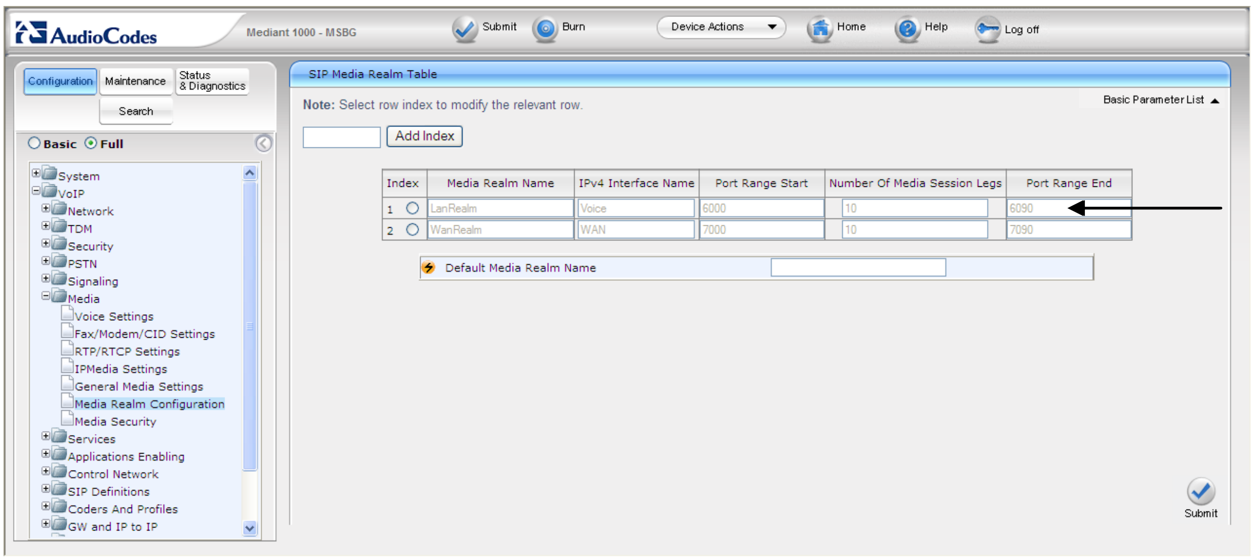
- Configure the parameter **Enable SBC Application** to **Enable**.
- Click the **Submit** button to save changes.
- Save the changes to flash memory. This is performed by selecting the **Burn** button at the top of the page. This is referred to as, "Saving Configuration", and will be referenced as such throughout this document.
- Notice the "Lightning Bolt" ⚡. All items marked with this symbol require a reset to take effect. Reset the device as noted previously in **Section 6.1**. Once the device is reset with the SBC application enabled, a submenu within VoIP menu will appear.



6.3. Configure Media Realm

Open the **Media Realm Configuration** page (**Configuration** tab > **VoIP** menu > **Media** submenu > **Media Realm Configuration** submenu) to configure the Media Realm settings.

- Configure the parameters as required. In the configuration used for compliance testing, only the **LanRealm** was used. The **Port Range Start** field indicates first RTP port of the range defined on the SBC. After the desired **Number of Media Session Legs** is entered, the SBC automatically populates the **Port Range End**.
- Click the **Submit** button to save changes.
- Save the changes to flash memory, refer to "Saving Configuration" as shown in **Section 6.2**.



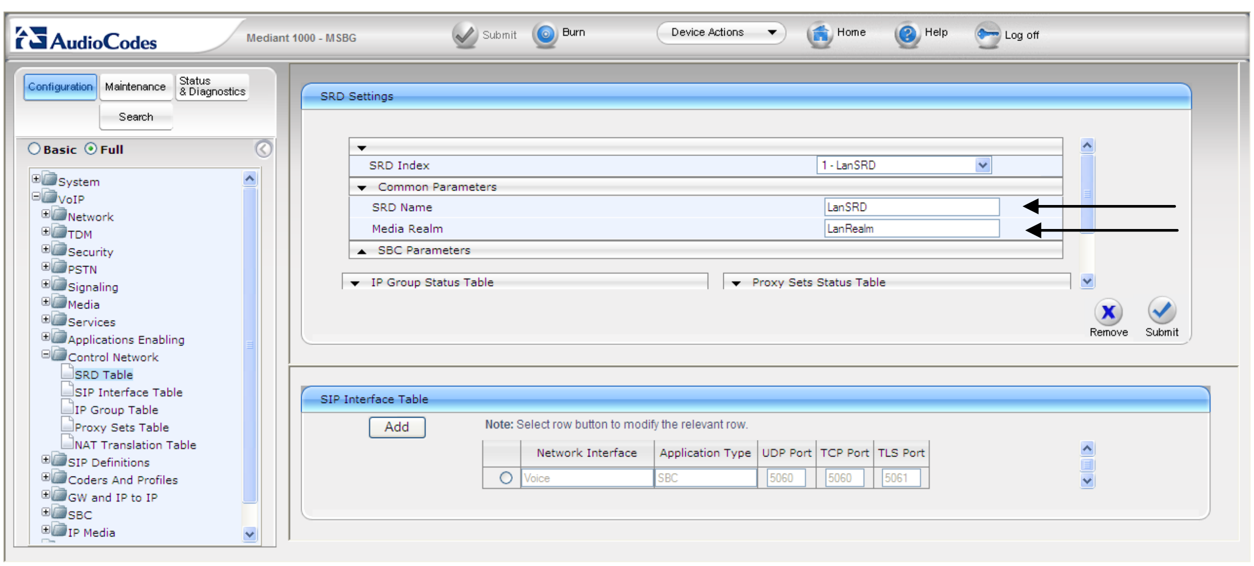
6.4. Configure SRD Table

Open the **SRD Table** page (**Configuration** tab > **VoIP** menu > **Control Network** submenu > **SRD Table** submenu) to configure the device's Signaling Routing Domain (SRD) table. An SRD is configured with a unique name and assigned a Media Realm (defined in **Section 6.3**). Once configured, SRDs can be used to do the following:

- Associate the SRD with a SIP Interface, IP Group, and Proxy Set.
- Define the SRD as a destination IP-to-IP routing rule.

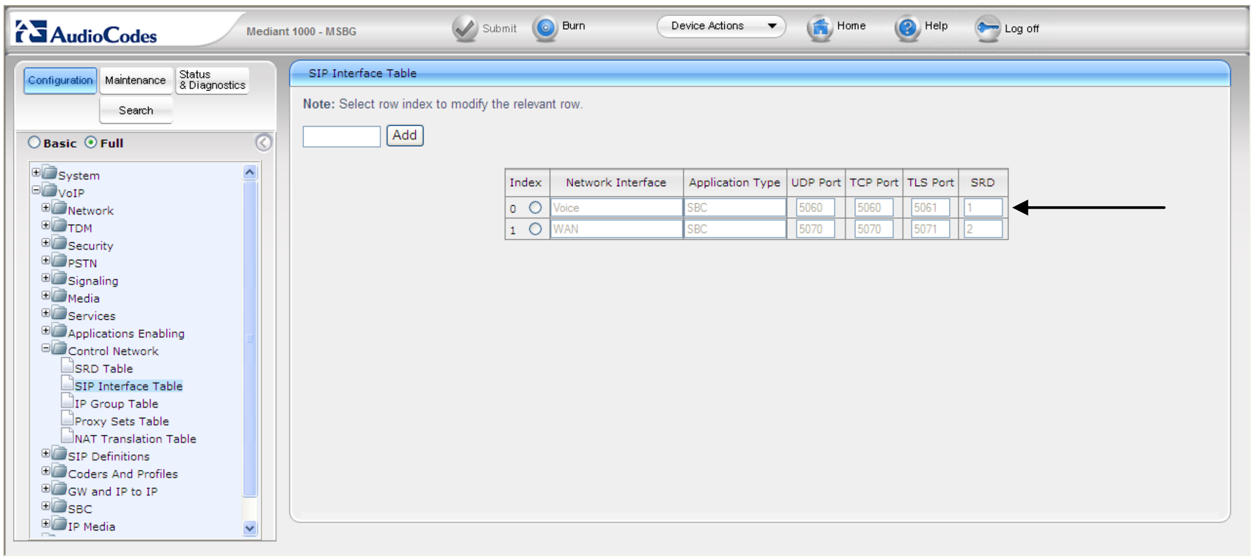
Therefore, an SRD is a set of definitions, together creating multiple, virtual multi-service IP gateways. Typically, one SRD is defined for each group of SIP User Agents (e.g. proxies, IP phones, application servers, gateways, soft switches, etc.) that communicate with each other. This provides these entities with VoIP services that reside on the same Layer-3 network (which must be able to communicate without traversing NAT devices and must not have overlapping IP addresses). Routing from one SRD to another is possible, whereby each routing destination (IP Group or destination address) indicates the SRD to which it belongs.

- Select an index that is unused.
- Configure the parameters as required. During compliance testing, **SRD Index** 1 (**LanSRD**) was mapped to **LanRealm**.
- Click the **Submit** button to save your changes.
- Save the changes to flash memory, refer to "Saving Configuration" as shown in **Section 6.2**.
- Repeat the process for the required SRD(s).
- Ensure that there is a unique SRD name which is bound to a Media Realm created previously.



6.5. Configure SIP Interfaces

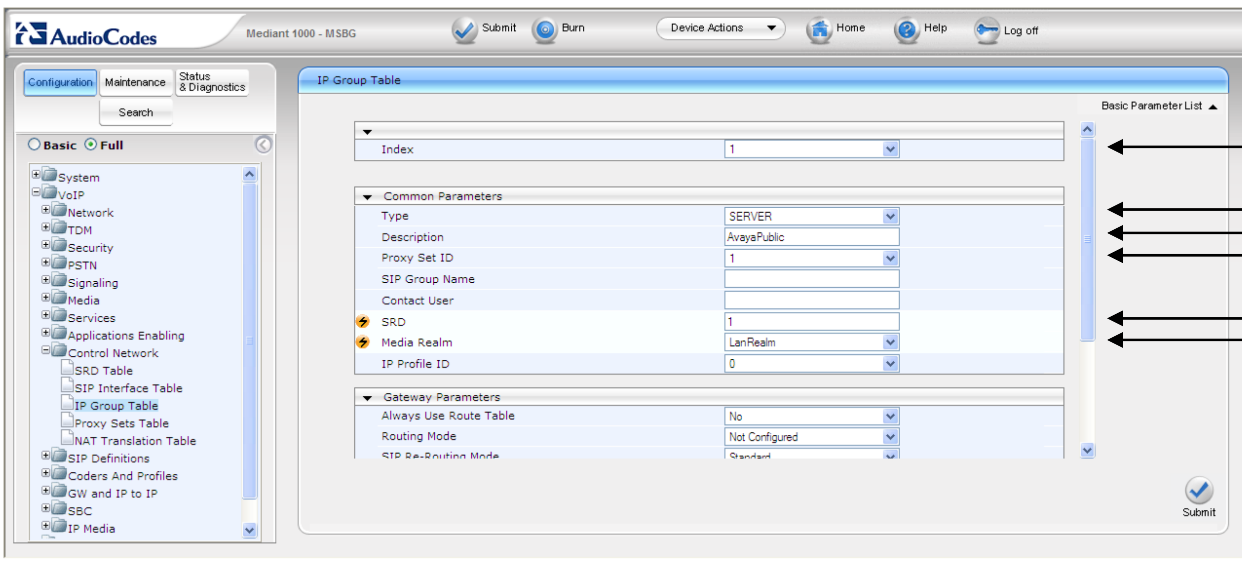
Create an interface in the **SIP Interface Table**. Ensure the **Network Interface** name used for the new index matches the name used in the initial settings for IP Settings, in this case **Voice**. This is the interface for the SBC Application. The SIP Interface table below states that the **Network Interface** known as **Voice** is being utilized by the SBC application. It also states that port **5060** should be used for both **UDP** and TCP, and port 5061 should be used for TLS. Note, port 5060 and UDP was utilized during compliance testing for communication between IP Office and the SBC, as defined in the SIP Line configuration in **Section 5**, **Step 4**. Finally, the table below states the **Voice** network interface is bound to **SRD 1**.

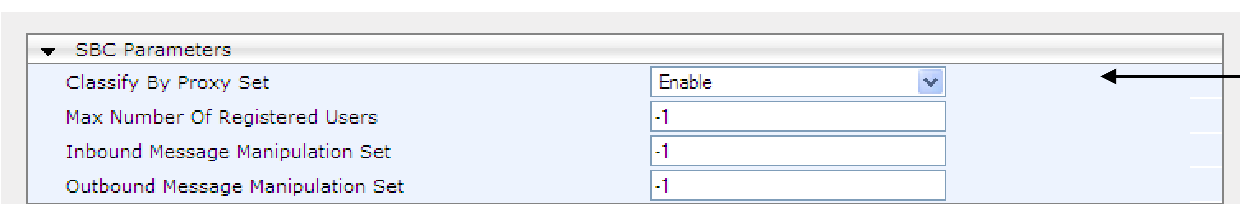


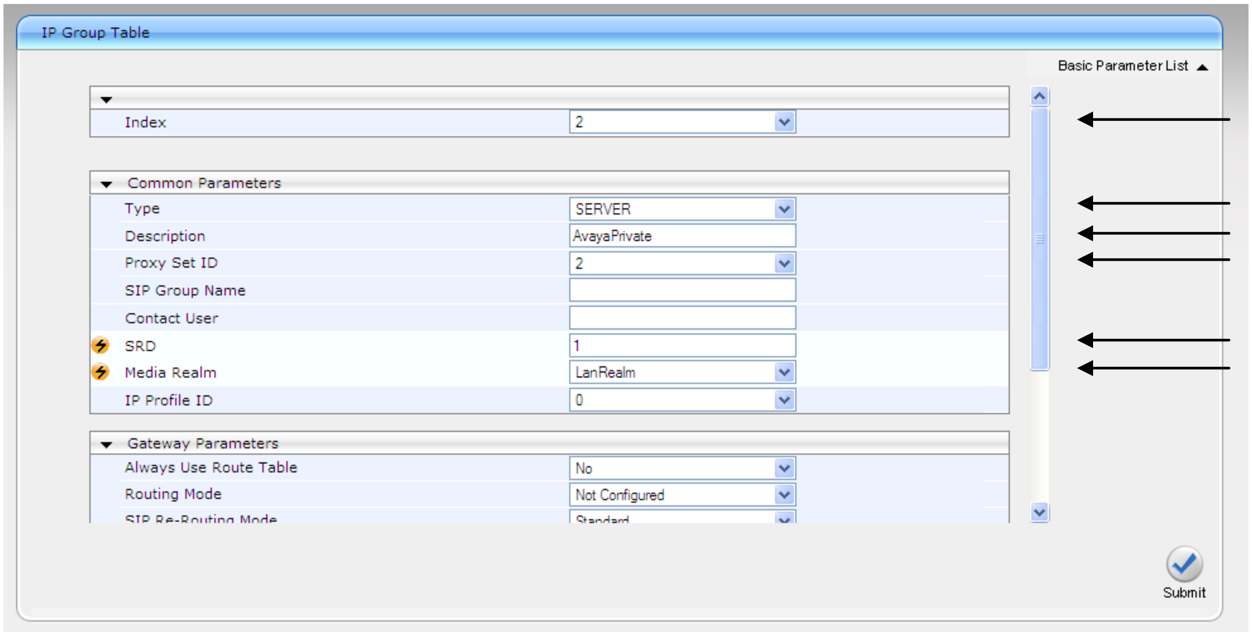
6.6. Configure the IP Group Table Settings

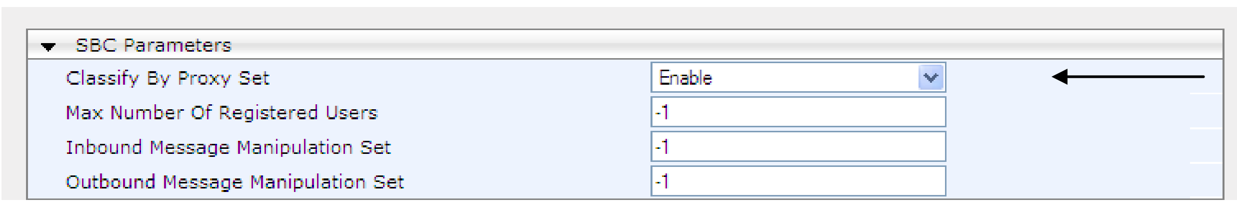
Open the **IP Group Table** page (**Configuration** tab > **VoIP** menu > **Control Network** submenu > **IP Group Table**) to configure the IP Group(s) and their respective parameters.

- Configure an unused IP Group index and assign appropriate parameters as required. During compliance testing, two indices were created representing the public interface (**AvayaPublic**) and the private interface (**AvayaPrivate**) on the SBC. Both indices used the **LanRealm** and **SRD 1** defined in **Sections 6.3** and **6.4**, respectively.
- Click the **Submit** button to save changes.
- Repeat previous two steps for the required amount of routes needed.
- To save the changes to flash memory, refer to "Saving Configuration" as shown in **Section 6.2**.






IP Group Table
Basic Parameter List
Index
2
Common Parameters
Type
SERVER
Description
AvayaPrivate
Proxy Set ID
2
SIP Group Name
Contact User
SRD
1
Media Realm
LanRealm
IP Profile ID
0
Gateway Parameters
Always Use Route Table
No
Routing Mode
Not Configured
Submit

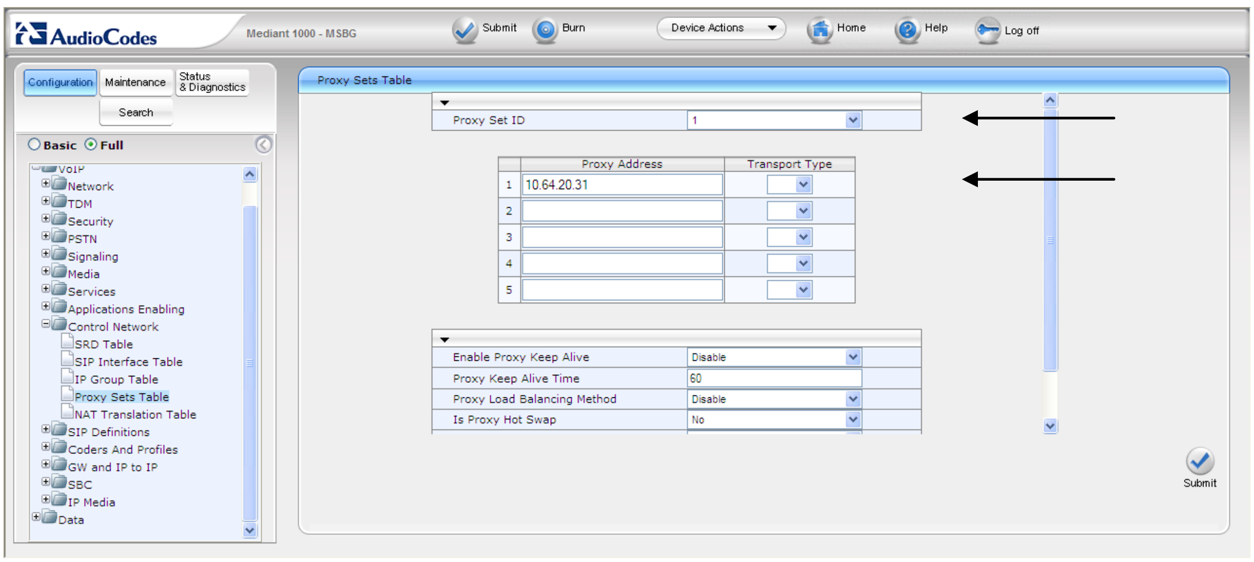

SBC Parameters
Classify By Proxy Set
Enable
Max Number Of Registered Users
-1
Inbound Message Manipulation Set
-1
Outbound Message Manipulation Set
-1

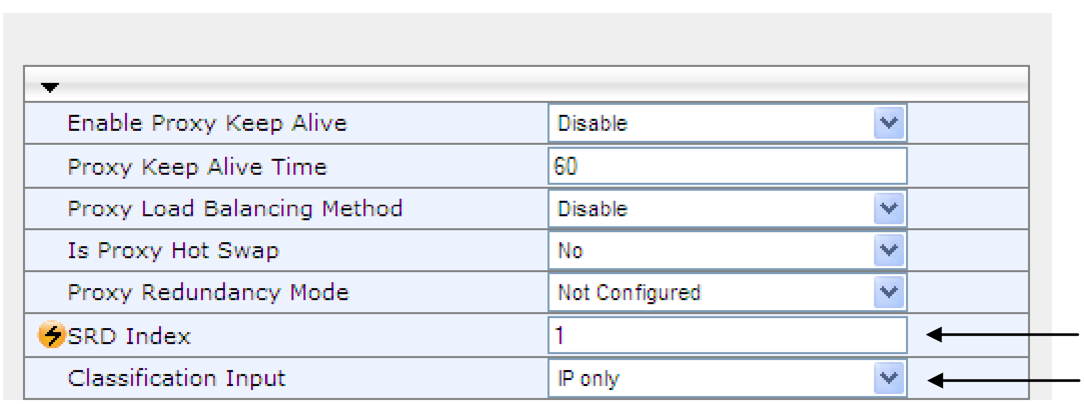
6.7. Configure Proxy Set Indices

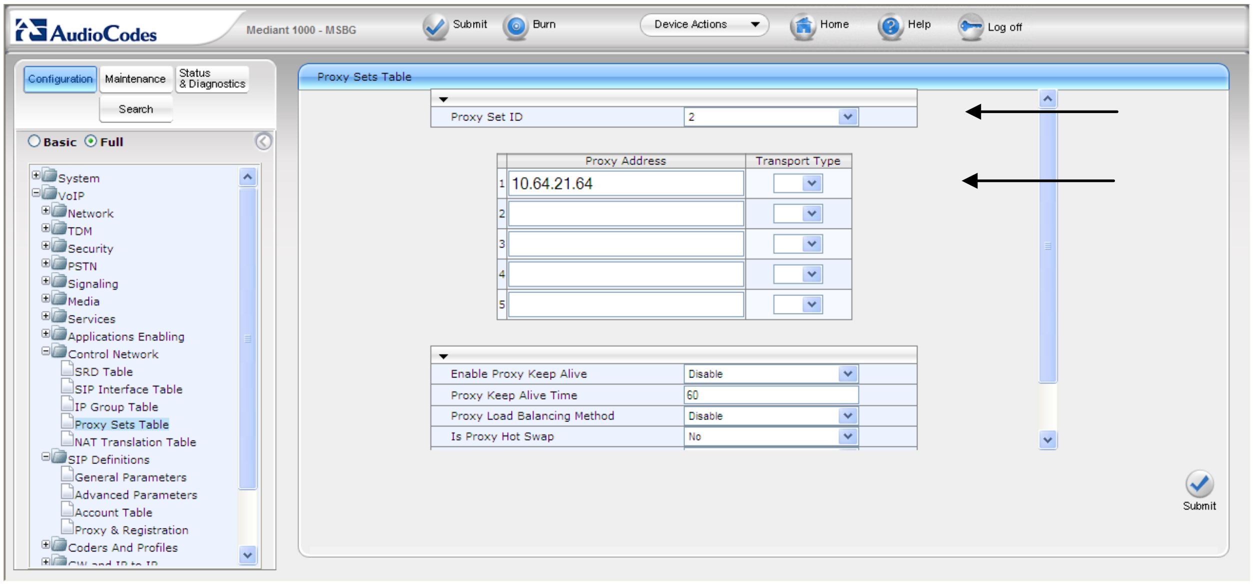
The use of Proxy Set index is utilized for identifying the specific Proxy (or set of proxy devices) for a respective IP Group Index (reference **Section 6.6** as an example: IP Group 1 is serviced by IP Proxy Set 1). Configure an unused Proxy Set Index and identify the IP address of the proxy for which calls will be routed. Do this for each unique IP group.

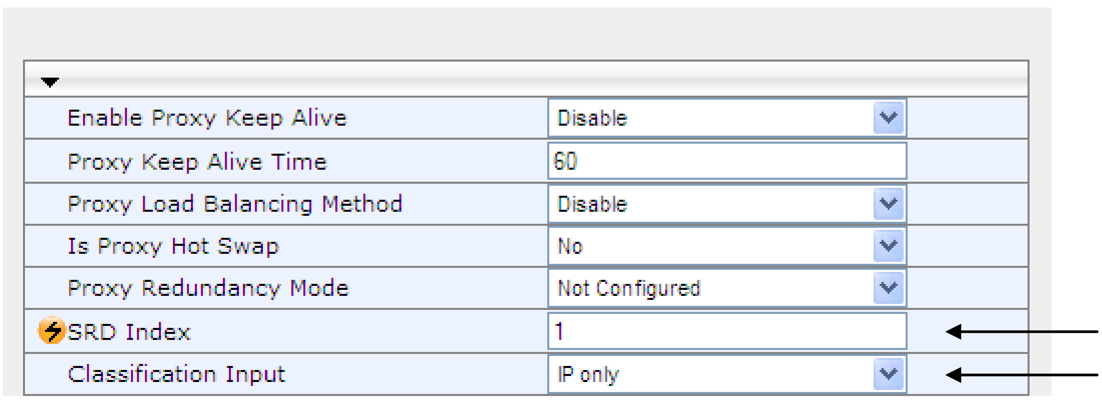
Open the **IP Group Table** page (**Configuration** tab > **VoIP** menu > **Control Network** submenu > **Proxy Sets Table**) to configure the Proxy Set(s) and their respective parameters:

- Configure an unused IP Group index and assign its appropriate parameters as required. (Note: 10.64.21.64 is the IP address of Avaya IP Office at the Enterprise site. 10.64.20.31 is the IP address of Session Manager at the simulated 2nd site.)
- Click the **Submit** button to save changes.
- Repeat previous two steps for the required amount of routes needed.
- To save the changes to flash memory, refer to "Saving Configuration" as shown in **Section 6.2**.





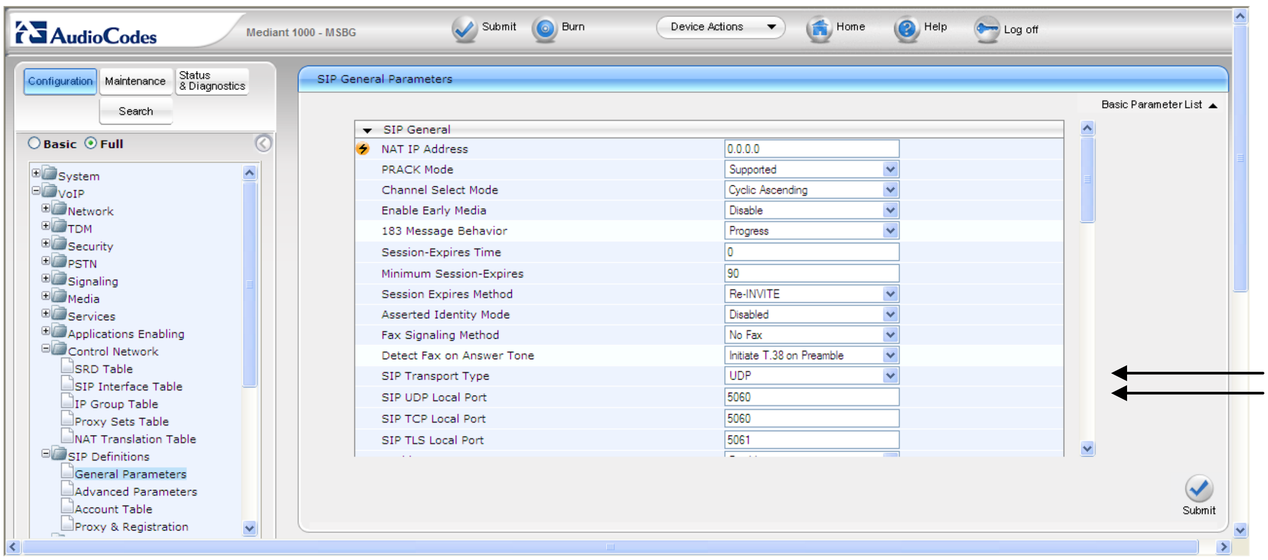

AudioCodes
Mediant 1000 - MSBG
Submit
Burn
Device Actions
Home
Help
Log off
Configuration
Maintenance
Status & Diagnostics
Search
Basic
Full
System
VoIP
Network
TDM
Security
PSTN
Signaling
Media
Services
Applications Enabling
Control Network
SRD Table
SIP Interface Table
IP Group Table
Proxy Sets Table
NAT Translation Table
SIP Definitions
General Parameters
Advanced Parameters
Account Table
Proxy & Registration
Coders And Profiles
Proxy Sets Table
Proxy Set ID
2
Proxy Address
Transport Type
10.64.21.64
Enable Proxy Keep Alive
Disable
Proxy Keep Alive Time
60
Proxy Load Balancing Method
Disable
Is Proxy Hot Swap
No
Submit

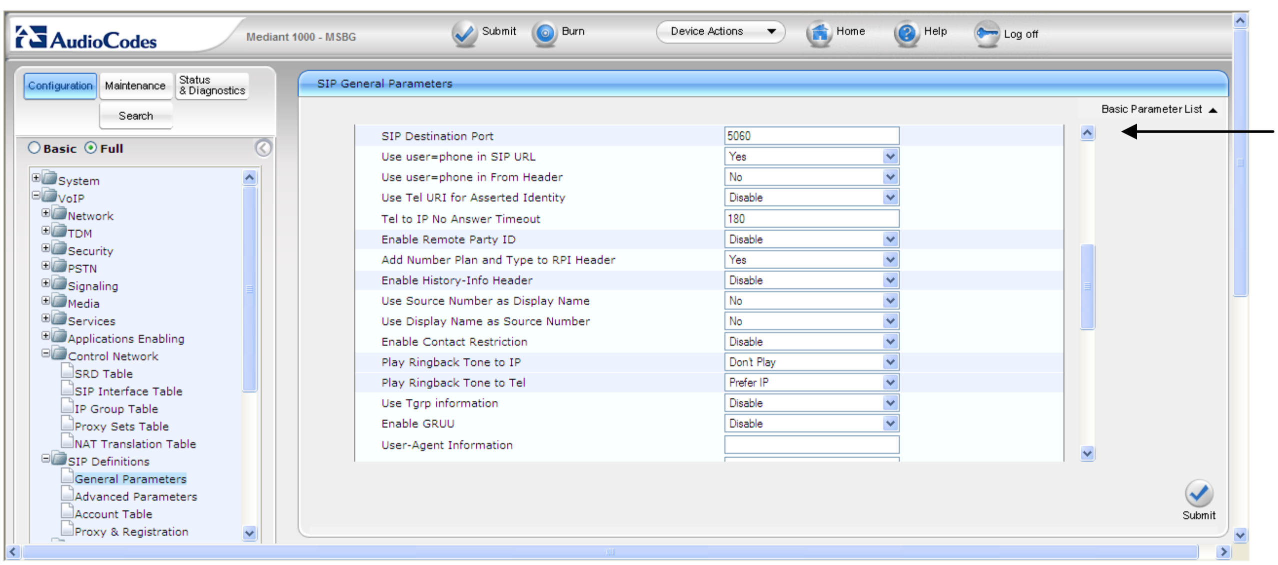

Enable Proxy Keep Alive
Disable
Proxy Keep Alive Time
60
Proxy Load Balancing Method
Disable
Is Proxy Hot Swap
No
Proxy Redundancy Mode
Not Configured
SRD Index
1
Classification Input
IP only

6.8. Configure SIP General Parameters

Open the **SIP General Parameters** page (**Configuration** tab > **VoIP** menu > **SIP Definitions** submenu > **General Parameters**) o configure the general SIP protocol parameters.

- Configure the parameters as required. (Note: Transport protocol UDP and Port 5060 were used for communication with IP Office. See the SIP Line configuration defined in **Section 5**, **Step 4**.)
- Click the **Submit** button to save changes.
- To save the changes to flash memory, refer to "Saving Configuration" as shown in **Section 6.2**.

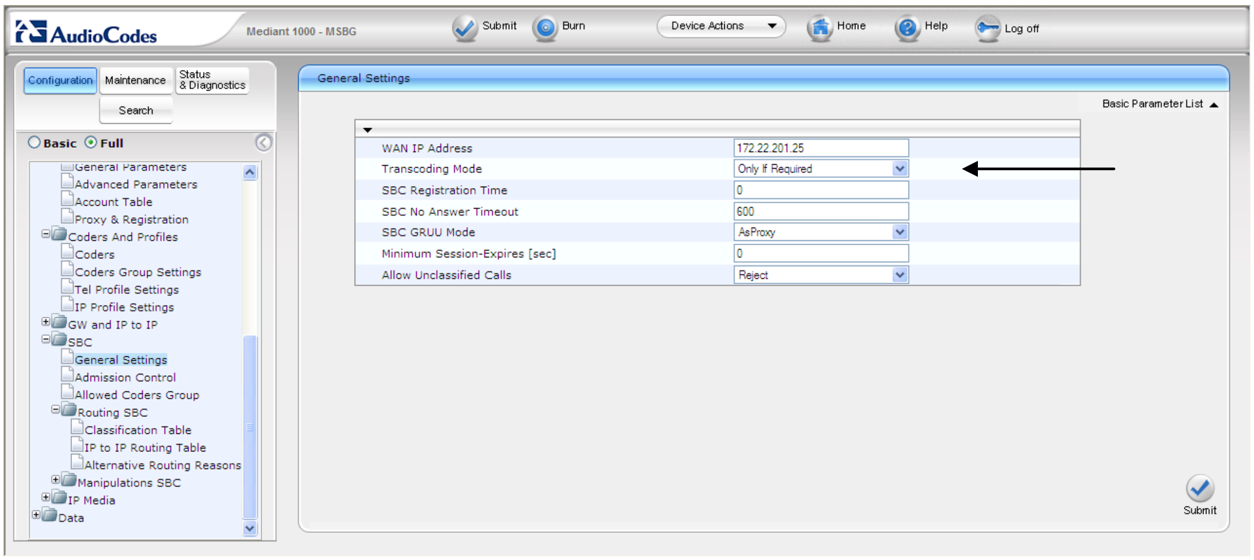




6.9. Configure General Settings

Open the **General Settings** page (**Configuration** tab > **VoIP** menu > **SBC** submenu > **General Settings**) to configure the general SBC parameters.

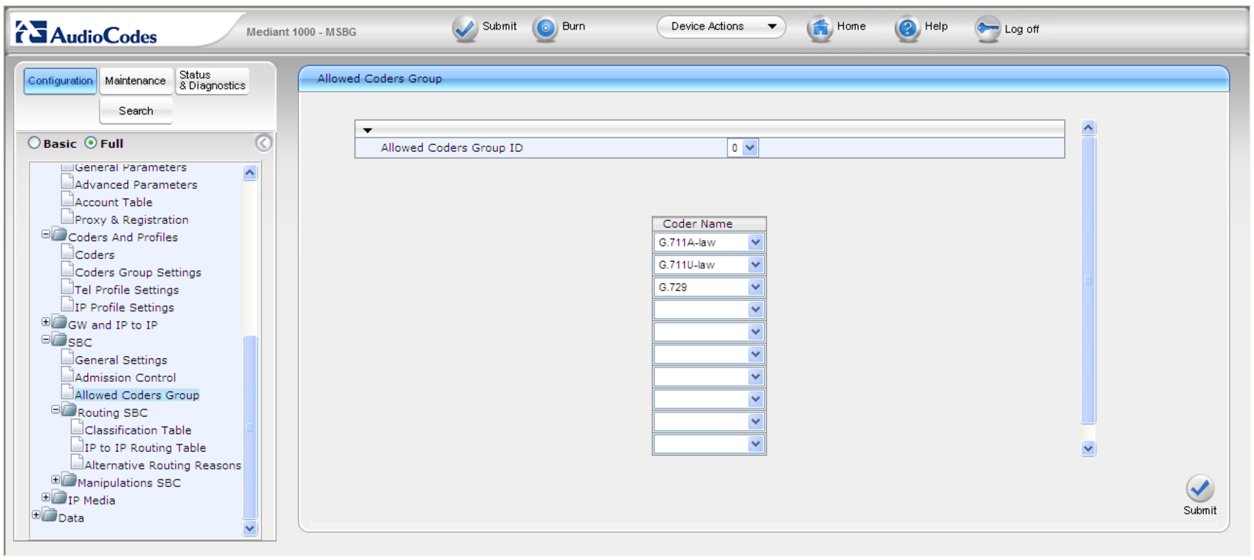
- Configure the parameters as required.
- Allowing of Unclassified calls is optional. All calls were classified by IP Group Index.
- Click the **Submit** button to save changes.
- To save the changes to flash memory, refer to "Saving Configuration" as shown in **Section 6.2**.



6.10. Configure Coders

Open the **Coders** page for the SBC application (**Configuration** tab > **VoIP** menu > **SBC** submenu > **Allowed Coders Group**) to configure the device's SBC Allowed coders.

- From the **Coder Name** drop-down list, select the required coder.
- Repeat steps for the next optional coders.
- Click the **Submit** button to save changes.
- To save the changes to flash memory, refer to "Saving Configuration" as shown in **Section 6.2**.



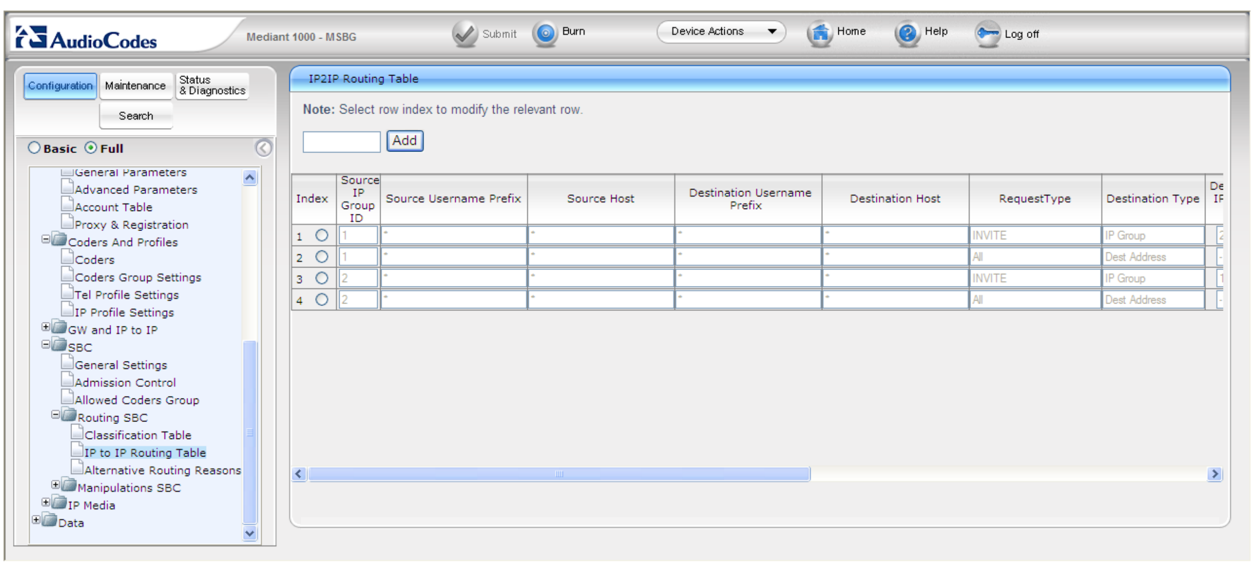
6.11. Configure IP to IP Routing Table

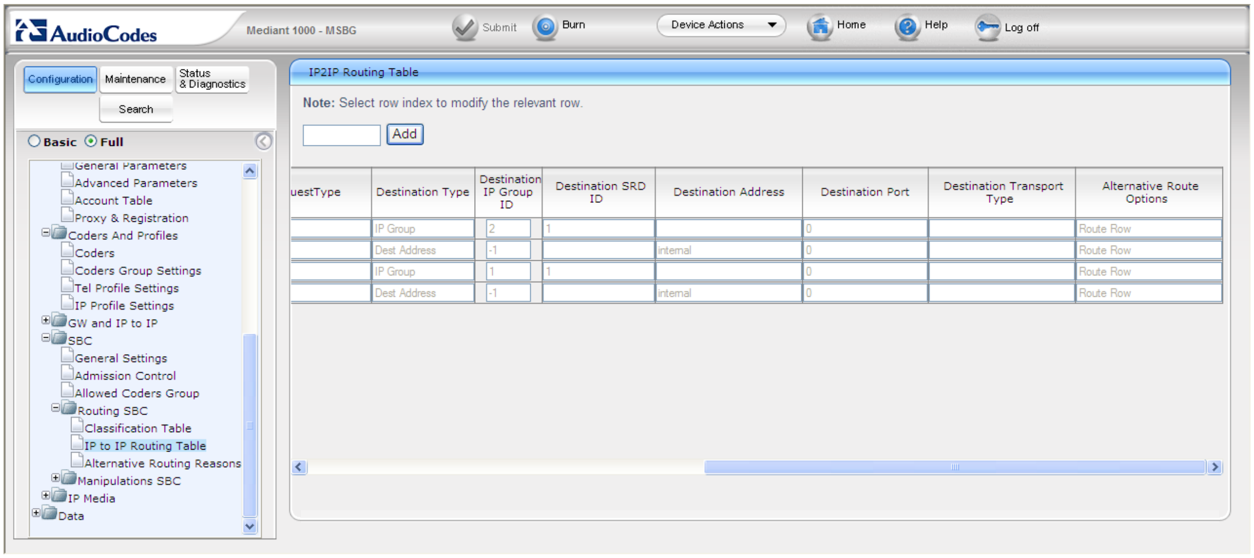
Open the **IP to IP Routing Table** page (**Configuration** tab > **VoIP** menu > **SBC** submenu > **Routing SBC** submenu > **IP to IP Routing Table**) to configure IP2IP routing rules.

The figures below shows the following configured outbound IP routing rules:

- **Rule 1:** If the incoming message originates from Source IP Group "1" and is associated with a call (Invite) then the call will be routed to a Destination IP Group of "2" and an SRD of "1".
- **Rule 2:** If the incoming message is not associated with a call, and originates from Source IP Group "1", then terminate the message to the internal device. This is set to enable the Avaya method of Heartbeat interworking for the product to return a 200 OK rather than send the received "Options" message to the terminating route.
- **Rule 3:** If the incoming message originates from Source IP Group "2" and is associated with a call (Invite) then the call will be routed to a Destination IP Group of "1" and an SRD of "1".
- **Rule 4:** If the incoming message is not associated with a call, and originates from Source IP Group "2", then terminate the message to the internal device. This is set to enable the Avaya method of Heartbeat interworking for the product to return a 200 OK rather than send the received "Options" message to the terminating route.

  - From the **Routing Index** drop-down list, select the range of entries that you want to add.
  - Configure the outbound IP routing rules according to the table below.
  - Click the **Submit** button to apply changes.
  - To save the changes to flash memory, refer to "Saving Configuration" as shown in **Section 6.2**.

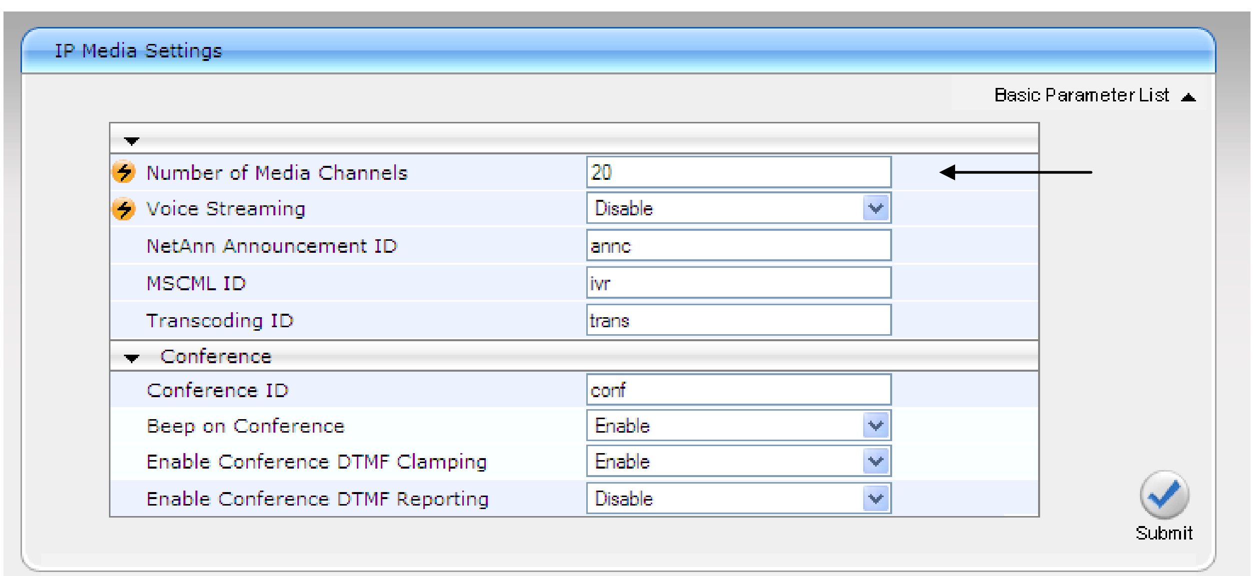




6.12. Configure IP Media Settings

Open the **IP Media Settings** page (**Configuration** tab > **VoIP** menu > **IP Media** submenu > **IP Media Settings**) to configure the IP Media Settings.

- Configure the IP Media Settings according to the required amount of supported sessions**.**
- Click the **Submit** button to save changes.
- To save the changes to the flash memory, refer to “Saving Configuration” as shown in **Section 6.2**.
- Reset the device to ensure the media resources are properly reserved.

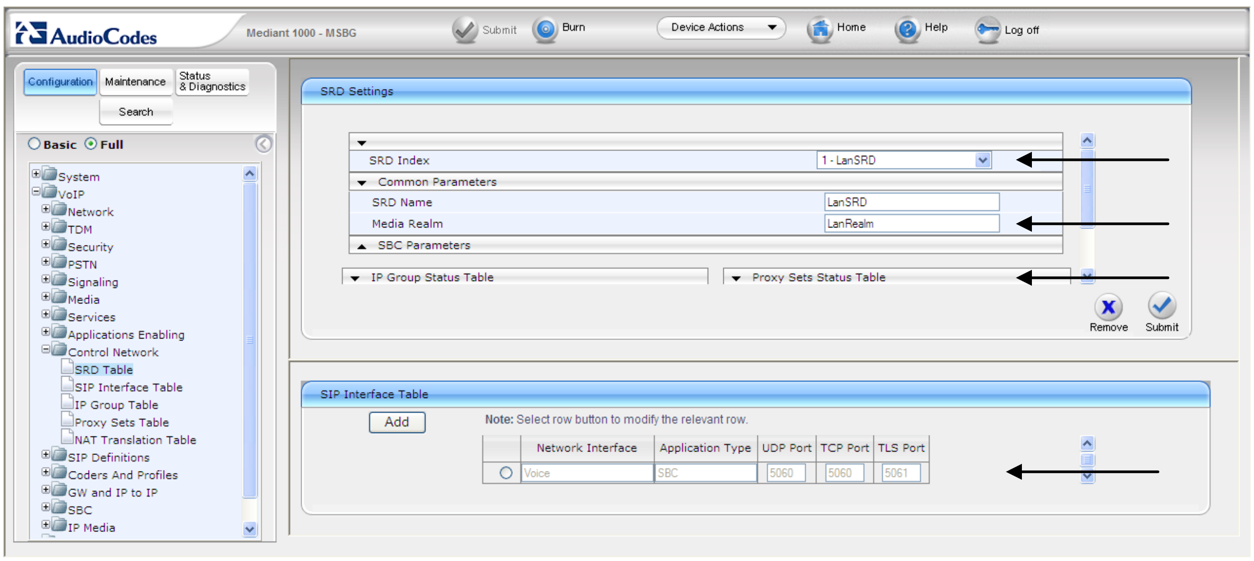


6.13. Configure SRD Table

Open the **SRD Table** page (**Configuration** tab > **VoIP** menu > **Control Network submenu > SRD Table** submenu) to view and confirm the device's intended SRD tables and respective routing interdependencies:

- Select the index that was created earlier.
- Insure the configured parameters are set as required.
- Click the IP Group Status and Proxy Sets Status sections to expand.
- Ensure the entries match that of the data previously entered.

Ensure the Network Interface name used for the new index matches the name used in the initial settings for IP Settings. This is the interface for the SBC Application.



IP Group Status Table

Index	Type	Description	Proxy set ID	SIP group name	IP profile ID
1	SERVER	AvayaPublic	1		0
2	SERVER	AvayaPrivate	2		0

- If Heart beating is required by the device, ensure that the value is set accordingly in the Proxy Set Indices.
- Ensure that there is a unique SRD name which is bound to a Media Realm created previously.

Proxy Sets Status Table

Index	Enable Proxy Keep Alive
1	Disable
2	Disable

7. Verification Steps

The proper installation/configuration of both the AudioCodes Mediant 1000 MSBG e-SBC and Avaya IP Office can be tested by placing inbound and outbound calls between two sites via the Mediant 1000 MSBG e-SBC. Verify that the calls are routed properly, the calls can be answered with 2-way talk paths, and that the calls can be terminated properly.

8. Conclusion

The AudioCodes Mediant 1000 MSBG e-SBC passed compliance testing. These Application Notes describe the procedures required to configure the AudioCodes Mediant 1000 MSBG e-SBC to interoperate with Avaya IP Office to support the network shown in **Figure 1**, where IP Office connects to the Mediant 1000 MSBG e-SBC using a SIP trunk.

9. Additional References

Product documentation for Avaya products may be found at http://support.avaya.com.

[1] *IP Office Installation*, Issue 22h, December 2010, Document Number 15-601042
[2] *IP Office Manager*, Issue 25k, January 2011, Document Number 15-601011

Additional IP Office documentation can be found at:

[3] *http://marketingtools.avaya.com/knowledgebase/*

Product documentation for the AudioCodes Mediant 1000 MSBG e-SBC e-SBC can be found at http://www.audiocodes.com/support.

[4] *LTRT-26901_SIP_CPE_Release_Notes_Ver._6.2.pdf*
[5] *LTRT-52306_SIP_CPE_Product_Reference_Manual_Ver_6.2.pdf*
[6] *LTRT-83508_Mediant_1000_SIP_Installation_Manual_Ver._6.2.pdf*
[7] *LTRT-83307_Mediant_600_and_Mediant_1000_SIP_User's_Manual_v6.2.pdf*

10. Appendix – AudioCodes .ini file

For completeness, the AudioCodes Mediant 1000 MSBG e-SBC ini configuration file (with its appropriate parameters) that was used during compliance testing is shown below:

```
;**************
;** Ini File **
;**************


[SYSTEM Params]

SyslogServerIP = 10.64.21.100
EnableSyslog = 1
PM_VEDSPUtil = '1,43,48,15'

[BSP Params]

PCMLawSelect = 3
RoutingTableDestinationsColumn = 128.0.0.0
RoutingTableDestinationPrefixLensColumn = 1
RoutingTableGatewaysColumn = 10.33.0.1
WANIPAddress = 172.22.201.25
WanInterfaceName = 'GigabitEthernet 0/0'

[Analog Params]

FXSLoopCharacteristicsFilename = 'M1K13-1-fxs16khz.dat'

[ControlProtocols Params]

AdminStateLockControl = 0

[MGCP Params]


[MEGACO Params]

EP_Num_0 = 0
EP_Num_1 = 1
EP_Num_2 = 1
EP_Num_3 = 0
EP_Num_4 = 0

[PSTN Params]


[SS7 Params]
```

[Voice Engine Params]

CNGDetectorMode = 1
CallProgressTonesFilename = 'usa_tones_13.dat'

[WEB Params]

LogoWidth = '145'
HTTPSCipherString = 'RC4:EXP'
WanMgmtHttpPort = 80

[SIP Params]

MEDIACHANNELS = 20
GWDEBUGLEVEL = 5
SIPGATEWAYNAME = 'avaya.com'
DISCONNECTONBROKENCONNECTION = 0
FAXCNGMODE = 1
ALLOWUNCLASSIFIEDCALLS = 1
ENABLESBCAPPLICATION = 1
SBCMAXFORWARDSLIMIT = 70

[SCTP Params]

[VXML Params]

[IPsec Params]

[Audio Staging Params]

[SNMP Params]

;
; *** TABLE InterfaceTable ***
;
;

[InterfaceTable]
FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes, InterfaceTable_InterfaceMode, InterfaceTable_IPAddress, InterfaceTable_PrefixLength, InterfaceTable_Gateway, InterfaceTable_VlanID, InterfaceTable_InterfaceName;

InterfaceTable 0 = 6, 10, 10.64.21.95, 24, 10.64.21.1, 1, Voice;
InterfaceTable 15 = 11, 10, 10.64.2.60, 16, 10.64.1.1, 1, Data;

[\InterfaceTable]

;
; *** TABLE DspTemplates ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts
;

;
; *** TABLE CpMediaRealm ***
;
;

[CpMediaRealm]
FORMAT CpMediaRealm_Index = CpMediaRealm_MediaRealmName, CpMediaRealm_IPv4IF, CpMediaRealm_IPv6IF, CpMediaRealm_PortRangeStart, CpMediaRealm_MediaSessionLeg, CpMediaRealm_PortRangeEnd;
CpMediaRealm 1 = LanRealm, Voice, , 6000, 10, 6090;
CpMediaRealm 2 = WanRealm, WAN, , 7000, 10, 7090;

[\CpMediaRealm]

;
; *** TABLE ProxyIp ***
;
;

[ProxyIp]
FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType, ProxyIp_ProxySetId;
ProxyIp 0 = 10.64.20.31, -1, 1;
ProxyIp 1 = 10.64.21.64, -1, 2;
ProxyIp 2 = 172.22.201.21, -1, 3;

[\ProxyIp]

;
; *** TABLE IpProfile ***
;
;

[IpProfile]
FORMAT IpProfile_Index = IpProfile_ProfileName, IpProfile_IpPreference, IpProfile_CodersGroupID, IpProfile_IsFaxUsed, IpProfile_JitterBufMinDelay, IpProfile_JitterBufOptFactor, IpProfile_IPDiffServ, IpProfile_SigIPDiffServ, IpProfile_SCE, IpProfile_RTPRedundancyDepth, IpProfile_RemoteBaseUDPPort, IpProfile_CNGmode,

IpProfile_VxxTransportType, IpProfile_NSEMode, IpProfile_IsDTMFUsed, IpProfile_PlayRBTone2IP, IpProfile_EnableEarlyMedia, IpProfile_ProgressIndicator2IP, IpProfile_EnableEchoCanceller, IpProfile_CopyDest2RedirectNumber, IpProfile_MediaSecurityBehaviour, IpProfile_CallLimit, IpProfile_DisconnectOnBrokenConnection, IpProfile_FirstTxDtmfOption, IpProfile_SecondTxDtmfOption, IpProfile_RxDTMFOption, IpProfile_EnableHold, IpProfile_InputGain, IpProfile_VoiceVolume, IpProfile_AddIEInSetup, IpProfile_SBCExtensionCodersGroupID, IpProfile_MediaIPVersionPreference, IpProfile_TranscodingMode, IpProfile_SBCAllowedCodersGroupID, IpProfile_SBCAllowedCodersMode, IpProfile_SBCMediaSecurityBehaviour, IpProfile_SBCRFC2833Behavior, IpProfile_SBCAlternativeDTMFMethod, IpProfile_SBCAssertIdentity, IpProfile_AMDSensitivityParameterSuit, IpProfile_AMDSensitivityLevel, IpProfile_AMDMaxGreetingTime, IpProfile_AMDMaxPostSilenceGreetingTime, IpProfile_SBCDiversionMode, IpProfile_SBCHistoryInfoMode;
IpProfile 1 = , 1, 0, 1, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0, 0, -1, 1, 0, 0, -1, 1, 4, -1, 1, 1, 0, 0, , -1, 0, 0, -1, 0, 0, 0, 0, -1, 0, 8, 300, 400, -1, -1;

[\IpProfile]

;
; *** TABLE ProxySet ***
;
;

[ProxySet]
FORMAT ProxySet_Index = ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime, ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap, ProxySet_SRD, ProxySet_ClassificationInput, ProxySet_ProxyRedundancyMode;
ProxySet 0 = 0, 60, 0, 0, 0, 0, -1;
ProxySet 1 = 0, 60, 0, 0, 1, 0, -1;
ProxySet 2 = 0, 60, 0, 0, 1, 0, -1;
ProxySet 3 = 0, 60, 0, 0, 2, 0, -1;

[\ProxySet]

;
; *** TABLE IPGroup ***
;
;

[IPGroup]
FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Description, IPGroup_ProxySetId, IPGroup_SIPGroupName, IPGroup_ContactUser, IPGroup_EnableSurvivability, IPGroup_ServingIPGroup, IPGroup_SipReRoutingMode, IPGroup_AlwaysUseRouteTable, IPGroup_RoutingMode, IPGroup_SRD, IPGroup_MediaRealm, IPGroup_ClassifyByProxySet, IPGroup_ProfileId, IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet, IPGroup_OutboundManSet, IPGroup_ContactName;
IPGroup 1 = 0, AvayaPublic, 1, avaya.com, avaya.com, 0, -1, 0, 0, -1, 1, LanRealm, 1, 0, -1, -1, -1, ;

IPGroup 2 = 0, AvayaPrivate, 2, avaya.com, avaya.com, 0, -1, 0, 0, -1, 1, LanRealm, 1, 0, -1, -1, -1, ;
IPGroup 3 = 0, WANMP21, 3, , , 0, -1, 0, 0, -1, 2, WanRealm, 1, 0, -1, -1, -1, ;

[\IPGroup]

;
; *** TABLE IP2IPRouting ***
;
;

[IP2IPRouting]
FORMAT IP2IPRouting_Index = IP2IPRouting_SrcIPGroupID, IP2IPRouting_SrcUsernamePrefix, IP2IPRouting_SrcHost, IP2IPRouting_DestUsernamePrefix, IP2IPRouting_DestHost, IP2IPRouting_RequestType, IP2IPRouting_DestType, IP2IPRouting_DestIPGroupID, IP2IPRouting_DestSRDID, IP2IPRouting_DestAddress, IP2IPRouting_DestPort, IP2IPRouting_DestTransportType, IP2IPRouting_AltRouteOptions;
IP2IPRouting 1 = 1, *, *, *, *, 1, 0, 2, 1, , 0, -1, 0;
IP2IPRouting 2 = 1, *, *, *, *, 0, 1, -1, -1, internal, 0, -1, 0;
IP2IPRouting 3 = 2, *, *, *, *, 1, 0, 1, 1, , 0, -1, 0;
IP2IPRouting 4 = 2, *, *, *, *, 0, 1, -1, -1, internal, 0, -1, 0;

[\IP2IPRouting]

;
; *** TABLE SIPInterface ***
;
;

[SIPInterface]
FORMAT SIPInterface_Index = SIPInterface_NetworkInterface, SIPInterface_ApplicationType, SIPInterface_UDPPort, SIPInterface_TCPPort, SIPInterface_TLSPort, SIPInterface_SRD;
SIPInterface 0 = Voice, 2, 5060, 5060, 5061, 1;
SIPInterface 1 = WAN, 2, 5070, 5070, 5071, 2;

[\SIPInterface]

;
; *** TABLE SRD ***
;
;

[SRD]
FORMAT SRD_Index = SRD_Name, SRD_MediaRealm, SRD_IntraSRDMediaAnchoring, SRD_BlockUnRegUsers, SRD_MaxNumOfRegUsers, SRD_EnableUnAuthenticatedRegistrations;
SRD 1 = LanSRD, LanRealm, 0, 0, -1, 1;
SRD 2 = WanSRD, WanRealm, 0, 0, -1, 1;

[\SRD]

```
;
;  *** TABLE CodersGroup0 ***
;
;

[ CodersGroup0 ]
FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime,
CodersGroup0_rate, CodersGroup0_PayloadType, CodersGroup0_Sce;
CodersGroup0 0 = g711Alaw64k, 20, 0, -1, 0;
CodersGroup0 1 = g711Ulaw64k, 20, 0, -1, 0;
CodersGroup0 2 = g729, 20, 0, -1, 0;

[ \CodersGroup0 ]

;
;  *** TABLE AllowedCodersGroup0 ***
;
;

[ AllowedCodersGroup0 ]
FORMAT AllowedCodersGroup0_Index = AllowedCodersGroup0_Name;
AllowedCodersGroup0 0 = g711Alaw64k;
AllowedCodersGroup0 1 = g711Ulaw64k;
AllowedCodersGroup0 2 = g729;

[ \AllowedCodersGroup0 ]
```

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.