Configuration Note
Connecting Microsoft® Lync™ and Skype SIP Trunk using AudioCodes® Mediant™ 1000 MSBG

Version 6.2
February 2012
Document #: LTRT-41301
# Table of Contents

1 Introduction ......................................................................................................... 9  
2 Components Information .................................................................................. 11  
   2.1 AudioCodes Gateway Version ................................................................ 11  
   2.2 Skype SIP Trunking Version ................................................................... 11  
   2.3 Microsoft Lync Version ........................................................................... 11  
   2.4 Topology .................................................................................................. 12  
3 Configuring Lync Server 2010 ......................................................................... 15  
   3.1 Configuring the Mediant 1000 MSBG as a ‘IP/PSTN Gateway’ ............ 15  
   3.2 Associating the ‘IP/PSTN Gateway’ with the Mediation Server ........... 20  
   3.3 Configuring the ‘Route’ on the Lync Server 2010................................. 26  
4 Configuring Mediant 1000 MSBG Device ........................................................ 37  
   4.1 Step 1: Configure IP Addresses ............................................................. 39  
      4.1.1 Configuring LAN IP Addresses ....................................................... 39  
      4.1.2 Configure WAN IP Addresses ....................................................... 41  
   4.2 Step 2: Configure Data Firewall .............................................................. 42  
   4.3 Step 3: Enable SIP SBC Application Mode ............................................ 43  
   4.4 Step 4: Configure Secure Real-Time Transport Protocol (SRTP) ........ 44  
   4.5 Step 5: Configure IP Media ..................................................................... 46  
   4.6 Step 6: Configure SIP General Parameters ........................................... 47  
   4.7 Step 7: Configure DTMF and Dialing ..................................................... 50  
   4.8 Step 8: Configure Coders ....................................................................... 51  
   4.9 Step 9: Configure Proxy and Registration ............................................. 52  
   4.10 Step 10: Configure Accounts ................................................................. 53  
   4.11 Step 11: Configure Proxy Sets ............................................................... 54  
   4.12 Step 12: Configure Coder Group ............................................................ 56  
   4.13 Step 13: Configure IP Profile ................................................................. 58  
   4.14 Step 14: Configure IP Group Tables ...................................................... 60  
   4.15 Step 15: Configure Routing Rules ......................................................... 62  
   4.16 Step 16 Configure Manipulation Rules .................................................. 64  
   4.17 Step 17: Secure Calls .............................................................................. 67  
   4.18 Step 18: Alternative Routing Reasons .................................................. 69  
   4.19 Step 19: Define SIP TLS Connection .................................................. 70  
      4.19.1 Step 19-1: Configure VoIP DNS Settings ....................................... 70  
      4.19.2 Step 19-2: Configure NTP Server ............................................... 70  
      4.19.3 Step 19-3: Configure a Certificate ............................................... 71  
   4.20 Step 20: Reset the Gateway .................................................................. 77  
5 Appendix A: AudioCodes INI file ..................................................................... 79  
6 Appendix B: Configuring Skype ...................................................................... 88  
   6.1 Skype Manager ......................................................................................... 88
List of Figures

Figure 2-1: Topology............................................................................................................................... 13
Figure 3-1: Starting the Lync Server Topology Builder ........................................................................ 15
Figure 3-2: Topology Builder Options ..................................................................................................... 16
Figure 3-3: Save Topology ...................................................................................................................... 17
Figure 3-4: Downloaded Topology .......................................................................................................... 18
Figure 3-5: New IP/PSTN Gateway .......................................................................................................... 18
Figure 3-6: Define New IP/PSTN Gateway ................................................................................................. 19
Figure 3-7: IP/PSTN Gateway .................................................................................................................. 19
Figure 3-8: Associating Mediation Server with IP/PSTN Gateway ........................................................... 20
Figure 3-9: Before Associating IP/PSTN Gateway to a Mediation Server Associations ......................... 21
Figure 3-10: After Associating IP/PSTN Gateway to Mediation Server ................................................. 21
Figure 3-11: Media Server PSTN Gateway Association Properties ..................................................... 22
Figure 3-12: Publishing Topology ......................................................................................................... 22
Figure 3-13: Publish Topology Confirmation ........................................................................................... 23
Figure 3-14: Publish Topology Confirmation screen ............................................................................... 24
Figure 3-15: Publish Topology Successfully Completed ......................................................................... 24
Figure 3-16: Lync Server Control Panel ................................................................................................. 26
Figure 3-17: Lync Server Credentials ................................................................................................... 27
Figure 3-18: CSCP Home page ............................................................................................................. 27
Figure 3-19: Voice Routing Option ......................................................................................................... 28
Figure 3-20: Route Option ...................................................................................................................... 28
Figure 3-21: Adding New Voice Route .................................................................................................. 29
Figure 3-22: List of Deployed Gateways ................................................................................................. 30
Figure 3-23: Selecting the Mediant 1000 MSBG Gateway ..................................................................... 31
Figure 3-24: Associating PSTN Usage to Mediant 1000 MSBG Gateway ............................................. 32
Figure 3-25: Confirmation of New Voice Route ....................................................................................... 33
Figure 3-26: Committing Voice Routes .................................................................................................. 33
Figure 3-27: Uncommitted Voice Configuration Settings ......................................................................... 34
Figure 3-28: Voice Routing Configuration Confirmation ......................................................................... 34
Figure 3-29: Voice Routing Screen Displaying Committed Routes ...................................................... 35
Figure 4-1: Web Interface Showing Basic/Full Navigation Tree Display ................................................ 38
Figure 4-2: IP Settings ............................................................................................................................ 39
Figure 4-3: Connections Page ................................................................................................................ 40
Figure 4-4: Defining LAN Data-Routing IP Address ............................................................................. 40
Figure 4-5: WAN Settings ....................................................................................................................... 41
Figure 4-6: General Security .................................................................................................................. 42
Figure 4-7: Applications Enabling ......................................................................................................... 43
Figure 4-8: Media Security Page ............................................................................................................. 44
Figure 4-9: IP Media Settings ................................................................................................................ 46
Figure 4-10: General Parameters ......................................................................................................... 47
Figure 4-11: General Parameters (Cont.) ............................................................................................... 48
Figure 4-12: INI file Output Window ..................................................................................................... 49
Figure 4-13: DTMF and Dialing ............................................................................................................. 50
Figure 4-14: Coders ............................................................................................................................... 51
<table>
<thead>
<tr>
<th>Figure</th>
<th>Description</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>4-15</td>
<td>Proxy and Registration</td>
<td>52</td>
</tr>
<tr>
<td>4-16</td>
<td>Account Table</td>
<td>53</td>
</tr>
<tr>
<td>4-17</td>
<td>Proxy Sets Table 1</td>
<td>54</td>
</tr>
<tr>
<td>4-18</td>
<td>Proxy Sets Table 2</td>
<td>55</td>
</tr>
<tr>
<td>4-19</td>
<td>Coders Group Settings</td>
<td>56</td>
</tr>
<tr>
<td>4-20</td>
<td>Coders Group Settings</td>
<td>57</td>
</tr>
<tr>
<td>4-21</td>
<td>IP Profile Settings</td>
<td>58</td>
</tr>
<tr>
<td>4-22</td>
<td>IP Profile Settings</td>
<td>59</td>
</tr>
<tr>
<td>4-23</td>
<td>IP Group Table 1</td>
<td>60</td>
</tr>
<tr>
<td>4-24</td>
<td>IP Group Table 2</td>
<td>61</td>
</tr>
<tr>
<td>4-25</td>
<td>Inbound IP Routing Table</td>
<td>62</td>
</tr>
<tr>
<td>4-26</td>
<td>Outbound IP Routing Table</td>
<td>63</td>
</tr>
<tr>
<td>4-27</td>
<td>Manipulation Tables</td>
<td>64</td>
</tr>
<tr>
<td>4-28</td>
<td>Advanced Parameters</td>
<td>67</td>
</tr>
<tr>
<td>4-29</td>
<td>Tel to IP Routing Table</td>
<td>68</td>
</tr>
<tr>
<td>4-30</td>
<td>Alternative Routing Reasons Table</td>
<td>69</td>
</tr>
<tr>
<td>4-31</td>
<td>VoIP DNS Settings</td>
<td>70</td>
</tr>
<tr>
<td>4-32</td>
<td>NTP Settings</td>
<td>70</td>
</tr>
<tr>
<td>4-33</td>
<td>Certificates Page</td>
<td>71</td>
</tr>
<tr>
<td>4-34</td>
<td>Microsoft Certificate Services Web Page</td>
<td>72</td>
</tr>
<tr>
<td>4-35</td>
<td>Request a Certificate Page</td>
<td>72</td>
</tr>
<tr>
<td>4-36</td>
<td>Advanced Certificate Request Page</td>
<td>73</td>
</tr>
<tr>
<td>4-37</td>
<td>Submit a Certificate Request or Renewal Request Page</td>
<td>74</td>
</tr>
<tr>
<td>4-38</td>
<td>Download a CA Certificate, Certificate Chain, or CRL Page</td>
<td>75</td>
</tr>
<tr>
<td>4-39</td>
<td>Certificates Page</td>
<td>76</td>
</tr>
<tr>
<td>4-40</td>
<td>Reset the GW</td>
<td>77</td>
</tr>
<tr>
<td>6-1</td>
<td>login Page</td>
<td>88</td>
</tr>
<tr>
<td>6-2</td>
<td>Main Page</td>
<td>88</td>
</tr>
<tr>
<td>6-3</td>
<td>Skype Connect page</td>
<td>89</td>
</tr>
<tr>
<td>6-4</td>
<td>Create Profile Page</td>
<td>89</td>
</tr>
<tr>
<td>6-5</td>
<td>Profile Settings Page</td>
<td>90</td>
</tr>
<tr>
<td>6-6</td>
<td>Channel Subscription Page</td>
<td>90</td>
</tr>
<tr>
<td>6-7</td>
<td>Outgoing Calls Page</td>
<td>91</td>
</tr>
<tr>
<td>6-8</td>
<td>Caller ID Page</td>
<td>91</td>
</tr>
<tr>
<td>6-9</td>
<td>Incoming Calls Page</td>
<td>92</td>
</tr>
<tr>
<td>6-10</td>
<td>Account page</td>
<td>92</td>
</tr>
</tbody>
</table>
List of Tables

Table 1-1: Acronyms ......................................................................................................................... 8
Table 2-1: AudioCodes Gateway Version .......................................................................................... 11
Table 2-2: Skype Version .................................................................................................................. 11
Table 2-3: Microsoft Lync Version .................................................................................................. 11
Notice

This document describes how to connect the Microsoft Lync server and Skype SIP Trunking using the Mediant 1000 MSBG SIP device.

Information contained in this document is believed to be accurate and reliable at the time of printing. However, due to ongoing product improvements and revisions, AudioCodes cannot guarantee accuracy of printed material after the Date Published, nor can it accept responsibility for errors or omissions. Updates to this document and other documents as well as software files can be viewed by registered customers at http://www.audiocodes.com/downloads.

© Copyright 2012 AudioCodes Ltd. All rights reserved.
This document is subject to change without notice.
Date Published: February-27-2012

Trademarks

AudioCodes, AC, AudioCoded, Ardito, CTI2, CTI², CTI Squared, HD VoIP, HD VoIP Sounds Better, InTouch, IPmedia, Mediant, MediaPack, NetCoder, Netrake, Nuer, Open Solutions Network, OSN, Stretto, TrunkPack, VMAS, VoicePacketizer, VoIPerfect, VoIPerfectHD, What's Inside Matters, Your Gateway To VoIP and 3GX are trademarks or registered trademarks of AudioCodes Limited. All other products or trademarks are property of their respective owners. Product specifications are subject to change without notice.

WEEE EU Directive

Pursuant to the WEEE EU Directive, electronic and electrical waste must not be disposed of with unsorted waste. Please contact your local recycling authority for disposal of this product.

Customer Support

Customer technical support and service are generally provided by AudioCodes’ Distributors, Partners, and Resellers from whom the product was purchased. For technical support for products purchased directly from AudioCodes, or for customers subscribed to AudioCodes Customer Technical Support (ACTS), contact support@audiocodes.com.
### Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used.

**Table 1-1: Acronyms**

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Meaning</th>
</tr>
</thead>
<tbody>
<tr>
<td>Transferee</td>
<td>The party being transferred to the transfer target</td>
</tr>
<tr>
<td>Transferor</td>
<td>The party initiating the transfer</td>
</tr>
<tr>
<td>Transfer target</td>
<td>The new party being introduced into a call with the transferee</td>
</tr>
<tr>
<td>Blind or semi-attended transfer</td>
<td>The transferor having a session in hold state with the transferee and initiating the transfer by a consultation call to the target performs the transfer while the target is in ringing state</td>
</tr>
<tr>
<td>Attended transfer or transfer on conversation</td>
<td>The transferor waits to be in conversation state with the target before completing the transfer</td>
</tr>
<tr>
<td>CLIP</td>
<td>Calling Line Identification Presentation</td>
</tr>
<tr>
<td>CNIP</td>
<td>Calling Name Identification Presentation</td>
</tr>
<tr>
<td>CLIR</td>
<td>Calling Line Identification Restriction</td>
</tr>
<tr>
<td>CNIR</td>
<td>Calling Name Identification Restriction</td>
</tr>
<tr>
<td>COLP</td>
<td>Connected Line Identification Presentation</td>
</tr>
<tr>
<td>CONP</td>
<td>Connected Name Identification Presentation</td>
</tr>
<tr>
<td>COLR</td>
<td>Connected Line Identification Restriction</td>
</tr>
<tr>
<td>CONR</td>
<td>Connected Name Identification Restriction</td>
</tr>
<tr>
<td>CRC</td>
<td>Customer Relationship Centre</td>
</tr>
<tr>
<td>PG</td>
<td>SIP GW XXX Peripheral Gateway</td>
</tr>
<tr>
<td>ICM</td>
<td>SIP GW XXX Intelligent Call Manager</td>
</tr>
<tr>
<td>CCM</td>
<td>SIP GW XXX Call Manager</td>
</tr>
<tr>
<td>CVP</td>
<td>Customer voice Portal</td>
</tr>
<tr>
<td>BC</td>
<td>ALU Business Contact</td>
</tr>
<tr>
<td>CTI</td>
<td>Computer Telephony Integration</td>
</tr>
</tbody>
</table>
1 Introduction

This document describes how to setup the MEDIANT 1000 MSBG device to work with the Skype SIP Trunking and Microsoft Lync Communication platform.

This configuration note is intended for Installation Engineers or AudioCodes and Skype Partners who are installing and configuring the Skype SIP Trunking and Microsoft Lync Communication platform to place VoIP calls using the AudioCodes gateway.

The AudioCodes Mediant 1000 MSBG device was used to implement this solution.

The Mediant 1000 MSBG is a networking device that combines multiple service functions, such as a Media Gateway, Session Border Controller (SBC), Data Router and Firewall, LAN switch, WAN access, Stand Alone Survivability (SAS) and an integrated general-purpose server. The MEDIANT MSBG device offers enhanced dialing plans and voice routing capabilities along with SIP-to-SIP mediation, allowing enterprises to implement SIP Trunking services (IP-to-IP call routing) and IP-based Unified Communications, as well as flexible PSTN and legacy PBX connectivity.

Note: The scope of this document does not cover security aspects for connecting the SIP Trunk to the Microsoft Lync environment. Security measures should be implemented in accordance with your organization's security policies. For basic security guidelines, see the ‘AudioCodes Security Guidelines’.
2 Components Information

2.1 AudioCodes Gateway Version

Table 2-1: AudioCodes Gateway Version

<table>
<thead>
<tr>
<th>Gateway Vendor</th>
<th>AudioCodes</th>
</tr>
</thead>
<tbody>
<tr>
<td>Model</td>
<td>Mediant 1000 MSBG</td>
</tr>
<tr>
<td>Software Version</td>
<td>SIP_6.20A.022.003</td>
</tr>
<tr>
<td>Interface Type</td>
<td>SIP/IP</td>
</tr>
<tr>
<td>VoIP Protocol</td>
<td>SIP</td>
</tr>
<tr>
<td>Additional Notes</td>
<td>None</td>
</tr>
</tbody>
</table>

2.2 Skype SIP Trunking Version

Table 2-2: Skype Version

<table>
<thead>
<tr>
<th>Gateway Vendor</th>
<th>Skype</th>
</tr>
</thead>
<tbody>
<tr>
<td>Models</td>
<td></td>
</tr>
<tr>
<td>Software Version</td>
<td></td>
</tr>
<tr>
<td>VoIP Protocol</td>
<td>SIP</td>
</tr>
<tr>
<td>Additional Notes</td>
<td>None</td>
</tr>
</tbody>
</table>

2.3 Microsoft Lync Version

Table 2-3: Microsoft Lync Version

<table>
<thead>
<tr>
<th>Gateway Vendor</th>
<th>Microsoft</th>
</tr>
</thead>
<tbody>
<tr>
<td>Models</td>
<td>Microsoft Lync</td>
</tr>
<tr>
<td>Software Version</td>
<td>RTM: Release 2010 4.0.7577.0</td>
</tr>
<tr>
<td>VoIP Protocol</td>
<td>SIP</td>
</tr>
<tr>
<td>Additional Notes</td>
<td>None</td>
</tr>
</tbody>
</table>
2.4 Topology

The procedures described in this document describe the following example scenario:

- An Enterprise has a deployed Microsoft® Lync server 2010 in its private network for enhanced communication within the company.
- The enterprise decides to offer its employees Enterprise voice and to connect the company to the PSTN network using the Skype SIP Trunking service.
- AudioCodes Session Border Controller (SBC) is used to manage the connection between the Enterprise LAN and the Skype SIP trunk.

The "session" refers to the real-time voice session using IP SIP signaling protocol. The "border" refers to the IP to IP network border between the Microsoft Lync network in the Enterprise LAN and the Skype SIP trunk in the public network.

Figure 2-1 below illustrates the interoperability topology between the Microsoft® Lync Server 2010 LAN and the Skype SIP Trunking site.

The setup requirements are characterized as follows:

- While the Microsoft® Lync Server 2010 environment is located on the Enterprise's Local Area Network (LAN), the Skype SIP Trunks are located on the WAN.
- The internal data routing capabilities of the Mediant 1000 MSBG device are used. Consequently, a separate WAN interface is configured in the LAN.
- Microsoft® Lync Server 2010 works with the TLS transport type, while the Skype SIP trunk works on the SIP over UDP transport type.
- Transcoding support: Microsoft® Lync Server 2010 supports G.711A-law and G.711U-law coders, while the Skype SIP Trunk also supports the G.729 coder type.
Support for early media handling

Figure 2-1: Topology

Skype Manager
Customer Site
Reader's Notes
3 Configuring Lync Server 2010

This section describes how to configure the Lync Server 2010 to operate with the Mediant 1000 MSBG. This section describes the following procedures:

2. Associating the ‘IP/PSTN Gateway’ with the Mediation Server. See Section 3.2 on page 20.
3. Configuring a ‘Route’ to utilize the SIP trunk connected to the Mediant 1000 MSBG. See Section 3.3 on page 26.

Note: Dial Plans, Voice Policies, and PSTN usages are also necessary for enterprise voice deployment; however, they are beyond the scope of this document.

3.1 Configuring the Mediant 1000 MSBG as a ‘IP/PSTN Gateway’

This section describes how to configure the Mediant 1000 MSBG as an IP/PSTN Gateway.

➢ To configure the Mediant 1000 MSBG as a IP/PSTN Gateway and associating it with the Mediation Server:

1. On the server where the Topology Builder is located, start the Lync Server 2010 Topology Builder: Click Start, select All Programs, then select Lync Server Topology Builder.

Figure 3-1: Starting the Lync Server Topology Builder
The following screen is displayed:

**Figure 3-2: Topology Builder Options**

2. Choose ‘Download Topology from the existing deployment and click **OK**. You are prompted to save the Topology which you have downloaded.
3. Enter new **File Name** and **Save** – this action enables you to rollback from any changes you make during the installation.

The Topology Builder screen with the topology downloaded is displayed.
4. Expand the Site; right-click on the IP/PSTN Gateway and choose ‘New IP/PSTN Gateway’.

Figure 3-5: New IP/PSTN Gateway
5. Enter the FQDN of the Mediant 1000 MSBG (i.e., 'Skype.lync.local') and click **OK**. Note that the listening port for the Gateway is '5067' and the transport type is 'TLS'. The Mediant 1000 MSBG is now added as a 'IP/PSTN Gateway'.
3.2 Associating the ‘IP/PSTN Gateway’ with the Mediation Server

This section describes how to associate the ‘IP/PSTN Gateway’ with the Mediation Server.

➢ To associate the IP/PSTN Gateway with the Mediation Server:

1. Right-click on the Mediation server to use with the Mediant 1000 MSBG (i.e. FE-Lync.Lync.local) and choose Edit Properties.
The following screen is displayed:

**Figure 3-9: Before Associating IP/PSTN Gateway to a Mediation Server Associations**

![Image of the screen showing the association process]

2. In the top-left corner, choose **PSTN gateway** and in the Mediation Server PSTN gateway pane, mark the Mediant 1000 MSBG gateway (i.e. 'Skype.lync.local') and click **Add** to associate it with this Mediation Server.

Note that there are two sub-panes, one including a list of gateways not associated with the Mediation server and one including a list of gateways associated with the Mediation server.

**Figure 3-10: After Associating IP/PSTN Gateway to Mediation Server**

![Image of the screen showing the associations after adding the gateway]

3. Click **OK**.
4. In the Lync Server main menu, choose **Action > Publish Topology**.

**Figure 3-12: Publishing Topology**
The Publish Topology screen is displayed.

**Figure 3-13: Publish Topology Confirmation**

### Publish the topology

In order for Lync Server 2010 to correctly route messages in your deployment, you must publish your topology. Before you publish the topology, ensure that the following tasks have been completed:

- A validation check on the root node did not return any errors.
- A file share has been created for all file stores that you have configured in this topology.
- All simple URLs have been defined.
- For Enterprise Edition Front End pools and for Monitoring Servers and Archiving Servers: All SQL stores are installed and accessible remotely; firewall exceptions for remote access to SQL Server are configured.
- For a single Standard Edition server: The task “Prepare first Standard Edition server” was run.
- You are currently logged on as a SQL administrator, for example, as a member of the SQL svadmin role.
- If you are removing a Front End pool, all users, common area phones, analog devices, application contact objects, and conference directories have been removed from the pool.

When you are ready to proceed, click Next.

5. Click **Next**.
   
The Topology Builder attempts to publish your topology.
Publishing in progress

Please wait while Topology Builder tries to publish your topology.

**Publishing topology ...**
**Downloading topology ...**
**Downloading topology ...**
Succeeded
**Downloading global simple URL settings.**

Wait until the publish topology process has ended successfully.

**Figure 3-15: Publish Topology Successfully Completed**

**Publishing wizard complete**

<table>
<thead>
<tr>
<th>Step</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Publishing topology ...</td>
<td>Success</td>
</tr>
<tr>
<td>Downloading topology ...</td>
<td>Success</td>
</tr>
<tr>
<td>Downloading global simple URL settings</td>
<td>Success</td>
</tr>
<tr>
<td>Enabling topology...</td>
<td>Success</td>
</tr>
</tbody>
</table>

To close the wizard, click Finish.

6. Click Finish.
3.3 Configuring the ‘Route’ on the Lync Server 2010

This section describes how to configure a ‘Route’ on the Lync server and associate it with the Mediant 1000 MSBG PSTN gateway.

➢ To configure the ‘route’ on the Lync server:

1. Open the Communication Server Control Panel (CSCP), click Start, select All Programs, and select Lync Server Control Panel.

Figure 3-16: Lync Server Control Panel

2. You are prompted for credentials; enter your domain username and password.
3. In the Navigation pane, select the ‘Voice Routing’ option.
4. In the Voice Routing menu at the top of the page, select the **Route** option.

5. In the content area toolbar, click **New**.

6. In the Build a Pattern to Match pane, fill in a Name for this route (i.e. SIP Trunk Route) and a Pattern to Match for the phone numbers you wish this route to handle. In this example, the pattern to match is "+", which means “to match all numbers”.

7. Click **Add**.
8. Associate the route with the Mediant 1000 MSBG IP/PSTN gateway you created above; scroll down to the Associated Gateways pane and click Add. A list of all the deployed Gateways is displayed.
9. Select the Mediant 1000 MSBG Gateway you created above and click **OK**.
10. Associate a PSTN Usage to this route. In the **Associated PSTN Usages** toolbar, click **Select** and add the associated PSTN Usage.
11. Click the OK button in the toolbar at the top of the New Voice Route pane.
12. In the Content area Toolbar, click on the arrow adjacent to the **Commit** button; a drop-down menu is displayed; select the ‘Commit All’ option.

13. In the Uncommitted Voice Configuration Settings window, click **Commit**.
The new committed Route is now displayed in the Voice Routing screen.

Figure 3-29: Voice Routing Screen Displaying Committed Routes
Reader's Notes
4 Configuring Mediant 1000 MSBG Device

This section describes the following steps for configuring the Mediant 1000 MSBG device in the Skype SIP Trunking environment:

- **Step 1**: Configure IP Addresses. See Section 4.1 on page 39.
- **Step 2**: Configure Data Firewall Settings. See Section 4.2 on page 42.
- **Step 3**: Configure Enable SIP SBC Applications. See Section 4.3 on page 43.
- **Step 4**: Configure Secure Real-Time Transport Protocol (SRTP). See Section 4.4 on page 44.
- **Step 5**: Configure IP Media. See Section 4.5 on page 46.
- **Step 6**: Configure SIP General Parameters. See Section 4.6 on page 47.
- **Step 7**: DTMF and Dialing. See Section 4.7 on page 50.
- **Step 8**: Configure Coders. See Section 4.8 on page 51.
- **Step 9**: Configure Proxy and Registration. See Section 4.9 on page 52.
- **Step 10**: Configure Account table. See Section 4.10 on page 53.
- **Step 11**: Configure Proxy Sets Tables. See Section 4.11 on page 54.
- **Step 12**: Configure Coder Group. See Section 4.12 on page 56.
- **Step 13**: Configure IP Profile. See Section 4.13 on page 58.
- **Step 14**: Configure IP Group Tables. See Section 4.14 on page 60.
- **Step 15**: Configure Routing Rules. See Section 4.15 on page 62.
- **Step 16**: Configure Manipulation Rules. See Section 4.16 on page 64.
- **Step 17**: Configure Secure Calls. See Section 4.17 on page 67.
- **Step 18**: Configure Alternative Routing Reasons. See Section 4.18 on page 69.
- **Step 19**: Define SIP TLS Connection. See Section 4.19 on page 70.
- **Step 19-1**: Configure VoIP DNS Settings. See Section 4.19.1 on page 70.
- **Step 19-2**: Configure NTP Server. See Section 4.19.2 on page 70.
- **Step 19-3**: Configure Certificates. See Section 4.19.3 on page 71.
- **Step 20**: Reset the Gateway. See Section 4.20 on page 77.
The procedures described in this section are performed using the Mediant 1000 MSBG devices' Web-based management tool (i.e., embedded Web server). Before you begin configuring the Mediant 1000 MSBG device, ensure that the Web interface's Navigation tree is in full menu display mode (i.e., the **Full** option on the Navigation bar is selected), as displayed below:

**Figure 4-1: Web Interface Showing Basic/Full Navigation Tree Display**
4.1 **Step 1: Configure IP Addresses**

This step describes how to configure LAN IP addresses when the internal data-routing capabilities of the Mediant 1000 MSBG device are used in order to connect to the Skype SIP Trunk. In this case, you must configure a separate WAN interface as described in this step.

**Notes:**
- The VoIP and Management interface must be in the same subnet as the data-routing interface as shown in the figure below.
- When operating with both VoIP and data-routing functionalities, it is recommended to define the Default Gateway IP address for the VoIP network interface in the same subnet and with the same VLAN ID as the IP address for the data-routing LAN interface as shown below.

### 4.1.1 Configuring LAN IP Addresses

This step describes how to configure the LAN addresses.

➢ To configure the VoIP IP settings:

1. Open the 'IP Settings' page (**Configuration** tab > **VoIP** menu > **Network** > **IP Settings**).

**Figure 4-2: IP Settings**

<table>
<thead>
<tr>
<th>Index</th>
<th>Application Type</th>
<th>IP Address</th>
<th>Prefix Length</th>
<th>Gateway</th>
<th>VLAN ID</th>
<th>Interface Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>OAMP + Media + Control</td>
<td>10.15.7.131</td>
<td>16</td>
<td>10.15.7.130</td>
<td>1</td>
<td>Voice</td>
</tr>
</tbody>
</table>

2. Select the 'Index' radio button corresponding to the Application Type "OAMP + Media + Control" (i.e., VoIP and management interface), and then click **Edit**.

3. Set the following parameters:
   - **IP-Address**: <Gateway IP-Address> (e.g., 10.15.7.131).
   - **Prefix Length**: The Subnet Mask in bits (e.g., 16 for 255.255.0.0).
   - **Gateway**: <Gateway Default Gateway> (e.g., 10.15.7.130).

4. Set the **WAN Interface Name**: "WAN Ethernet". This is the WAN interface (configured in Section 0 on page 41) on which your VoIP traffic interfaces with the public network.
To define the Mediant 1000 MSBG device's LAN data-routing IP address:

1. Access the Mediant 1000 MSBG device's Web interface with the IP address that you assigned to the VoIP and Management interface.

2. Access the 'Connections' page (Configuration tab > Data menu > Data System > Connections).

3. Click the Edit icon corresponding to the "LAN Switch VLAN 1" connection, and then click the Settings tab.

4. In the 'IP Address' and 'Subnet Mask' fields, enter the required IP address (e.g., 10.15.7.130) and subnet respectively, and then click OK.
4. Configuring Mediant 1000 MSBG Device

4.1.2 Configure WAN IP Addresses

This step describes how to configure the Mediant 1000 MSBG device firewall IP address used to connect to the WAN.

To configure the WAN IP address:

1. Cable the Mediant 1000 MSBG device to the WAN network (i.e., ADSL or Cable modem), using the WAN port.
2. Open the ‘Settings’ page (Configuration tab > Data menu > WAN Access > Settings).

3. Set the following parameters:
   - **IP Address**: <WAN IP-Address> (e.g., 195.189.192.154).
   - **Subnet Mask**: <Subnet Mask> (e.g., 255.255.255.128).
   - **Default Gateway**: <WAN Default GW IP-Address> (e.g., 195.189.192.129).
   - **Primary DNS Server**: <First DATA DNS IP-Address> (e.g., 80.179.52.100).
   - **Secondary DNS Server**: <Second Data DNS IP-Address> (e.g., 80.179.55.100).
4.2 Step 2: Configure Data Firewall

This step describes how to configure the data firewall settings for the Mediant 1000 MSBG device’s WAN interface. You must define firewall rules for the WAN interface to prevent unwanted access from the public network. The configuration shown below represents a typical WAN firewall implementation.

To configure data firewall settings:

1. Open the ‘General Security’ page (Configuration tab > Data menu > Firewall and ACL > General Security).

![Figure 4-6: General Security]


![Figure 4-6: General Security]
4.3 Step 3: Enable SIP SBC Application Mode

This step describes how to enable the SIP SBC application mode.

➢ To enable the IP2IP application mode:

1. Open the 'Applications Enabling' page (Configuration tab > VoIP menu > Applications Enabling > Applications Enabling).

   ![Applications Enabling](image)

2. Enable IP2IP Application.

Note:

1. To enable the SBC and IP2IP capabilities on the AudioCodes gateway, your gateway must be loaded with the feature key that includes the IP2IP feature.
2. The Mediant 1000 MSBG device must be running SIP version 6.2 or later.
3. Reset with BURN to FLASH is required.
4.4 Step 4: Configure Secure Real-Time Transport Protocol (SRTP)

If you configure TLS for the SIP transport link between the Mediant 1000 MSBG and the Mediation Server, you must specify Secure RTP (SRTP) encryption with one of the following options:

- **Required**: SRTP should be attempted, but do not use encryption if negotiation for SRTP is unsuccessful.
- **Optional**: Attempt to negotiate the use of SRTP to secure media packets. Use RTP if SRTP cannot be negotiated.
- **Not used**: Send media packets using RTP.

If you choose to configure the Mediation Server to use SRTP (Required or Optional), you need to configure the Media Gateway to operate in the same manner.

➢ To configure the media security:

1. Open the 'Media Security' page (Configuration tab > Media menu > Media Security).

   ![Figure 4-8: Media Security Page]

2. Set the **Media Security** to ‘Enable’.

3. Set the **Media Security Behavior**:
   - “Mandatory” if Mediation Server is configured to SRTP Required
   - “Preferable-Single media” if Mediation Server is configured to SRTP Optional.

4. Set the **Master Key Identifier (MKI) Size** to ‘1’.

5. Click **Submit**.

6. Save (burn) the Mediant 1000 MSBG configuration and reset the Gateway.
Note: In order to set the 'Media Security Behavior' to the IP Profile of the Mediation Server, see the IP Profile Settings (see Section 4.9 on page 52).
4.5 **Step 5: Configure IP Media**

This step describes how to configure the number of media channels for the IP media. In this configuration, 120 channels are configured.

➢ **To configure IP Media Settings:**

1. Open the 'IP Media Settings' page (Configuration tab > VoIP menu > IP Media > IP Media Settings).

![Figure 4-9: IP Media Settings](image)

2. Set **Number of Media Channels** to ‘120’.
4.6 Step 6: Configure SIP General Parameters

This step describes how to enable SIP General parameters.

To configure SIP General Parameters:

1. Open the 'Applications Enabling' page (Configuration tab > VoIP menu > SIP Definitions > General Parameters).

Figure 4-10: General Parameters

2. Set 'NAT IP Address', with the Global (public) IP address of the Mediant 1000 MSBG device.
3. Set Enable Early Media to 'Enable'.
4. Set Fax Signaling Method to 'G.711 Transport'.
5. Set SIP Transport Type to 'TLS'.
6. Set SIP TLS Local Port to '5067' (Lync server port).
7. Set SIP Destination Port to '5067' (Lync Server port)
8. Set **Play Ringback Tone to Tel** to ‘Play Local Until Remote Media Arriving’.

**Figure 4-11: General Parameters (Cont.)**

8

9. Set **Forking Handling Mode** to ‘Sequential handling’.

10. Set **Fake Retry After** to ‘60’.

11. Open the ‘Admin’ page, by appending the case-sensitive suffix ‘AdminPage’ to the Media Gateway’s IP address in your Web browser’s URL field (e.g., http://10.15.7.131/AdminPage).

12. On the left pane, click **ini Parameters**.

   In the ‘Parameter Name’ field, enter the following parameters
   - **IGNOREALERTAFTEREARLYMEDIA**: In the Enter Value field, enter ‘1’
   - **ENABLEEARLY183**: In the Enter Value field, enter ‘1’
   - **PLAYHELDTONEFORIP2IP**: In the Enter Value field, enter ‘1’

13. Click **Apply New Value**.
Figure 4-12: INI file Output Window
4.7 Step 7: Configure DTMF and Dialing

This step describes how to configure the DTMF and Dialing settings.

➢ To configure DTMF and Dialing:

1. Open the 'DTMF and Dialing' page (Configuration tab > VoIP menu > GW and IP to IP > DTMF and Supplementary > DTMF and Dialing).

![Figure 4-13: DTMF and Dialing]

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Max Digits In Phone Num</td>
<td>30</td>
</tr>
<tr>
<td>Inter Digit Timeout [sec]</td>
<td>4</td>
</tr>
<tr>
<td>Declare RFC 2833 in SDP</td>
<td>Yes</td>
</tr>
<tr>
<td>1st TX DTMF Option</td>
<td>RFC 2833</td>
</tr>
<tr>
<td>2nd TX DTMF Option</td>
<td></td>
</tr>
<tr>
<td>RFC 2833 Payload Type</td>
<td>96</td>
</tr>
<tr>
<td>Hook-Flash Option</td>
<td>Not Supported</td>
</tr>
<tr>
<td>Digit Mapping Rules</td>
<td></td>
</tr>
<tr>
<td>Dial Plan Index</td>
<td>-1</td>
</tr>
<tr>
<td>Dial Tone Duration [sec]</td>
<td>16</td>
</tr>
<tr>
<td>Hotline Dial Tone Duration [sec]</td>
<td>16</td>
</tr>
<tr>
<td>Enable Special Digits</td>
<td>Disable</td>
</tr>
</tbody>
</table>

2. Set Max Digits In Phone Num to ‘30’.
3. Set RFC 2833 Payload Type to ‘96’.
4.8 Step 8: Configure Coders

This step describes how to configure the SIP coders. Since the Mediation Server support only G.711A-law and G.711U-law voice coders, while the ITSP SIP trunk additionally supports the G.729 coder, you need to configure for the entries shown in the screen below.

➢ To configure Coders:

1. Open the 'Coders' page (Configuration tab > VoIP menu > Coders and Profiles > Coders).

![Figure 4-14: Coders](image)

2. Set the coders G.711A-law and G.711U-law.
4.9  **Step 9: Configure Proxy and Registration**

This step describes how to configure the SIP Proxy and Registration. This configuration includes setting a redundant route for the Microsoft Lync Proxy Set.

➢ **To configure Proxy and Registration:**

1. Open the 'Proxy and Registration' page (Configuration tab > VoIP menu > SIP Definitions > Proxy and Registration).

   ![Figure 4-15: Proxy and Registration](image)

2. Set **Use Default Proxy** to ‘No’.
3. Set **Redundancy Mode** to ‘Homing’.
4. Set **Redundant Routing Mode** to ‘Proxy’.
   
   This will allow entry back into the Proxy Set table for the next available route.
5. Set **Gateway Name** to Gateway FQDN Name (e.g., ‘Skype.Lync.local’) (note that you configured this name in Section 4.19.3 on page 71).
4.10  **Step 10: Configure Accounts**

This step describes how to configure and register Skype client extensions on the SIP Trunk. The Skype SIP trunk can register 10 Skype user client extensions for each account table entry. In this configuration, a single user client extension is registered. Note that in Section 4.14 on page 60, an IP Group is configured for the user entry in the table below.

➢ To configure Accounts:

1. Open the 'Account Table' page (Configuration tab > VoIP menu > SIP Definitions > Account Table).

   ![Figure 4-16: Account Table](image)

2. Enter an index table entry number, and then click **Add**.
3. Configure the account user entry according to the example table above.
4.11 Step 11: Configure Proxy Sets

This step describes how to configure the proxy set tables. You need to configure two proxy sets, one for the Microsoft Lync server and the other for the Skype SIP trunk.

➢ To configure Proxy Set 1 for Microsoft Lync:

1. Open the 'Proxy Sets Table' page (Configuration tab > VoIP menu > Control Network> Proxy Sets Table).

2. Set Proxy Set ID to ‘1’.
3. Configure Microsoft Lync Server SIP Trunking IP-Address or FQDN and Destination Port (e.g., FE-Lync.Lync.local).
4. Set Transport Type to ‘TLS’.
5. Set Enable Proxy Keep Alive to ‘Using Options’.
7. Set Is Proxy Hot Swap to ‘Yes’.
To configure Proxy Set 2 for Skype SIP Trunk:

1. Open the ‘Proxy Sets Table’ page (Configuration tab > VoIP menu > Control Network > Proxy Sets Table).

2. Set Proxy Set ID to ‘2’.

3. Configure Skype IP-Address or FQDN and Destination Port (e.g., ‘sip.skype.com:5060’).

4. Set Transport Type to ‘UDP’.

5. Set Enable Proxy Keep Alive to ‘Disable’.

Figure 4-18: Proxy Sets Table 2

<table>
<thead>
<tr>
<th>Proxy Set ID</th>
<th>Proxy Address</th>
<th>Transport Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>sip.skype.com:5060</td>
<td>UDP</td>
</tr>
<tr>
<td>2</td>
<td></td>
<td></td>
</tr>
<tr>
<td>3</td>
<td></td>
<td></td>
</tr>
<tr>
<td>4</td>
<td></td>
<td></td>
</tr>
<tr>
<td>5</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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: 0
Classification Input: IP only
4.12 Step 12: Configure Coder Group

This step describes how to configure the Coder Groups. Microsoft Lync supports only the G.711 coder, while the network connection to Skype may restrict you to work with lower bandwidth coders, such as G.729. The 'Coder Group Settings' allow you to define up to four different Coder Groups. These Coder Groups are then assigned to IP Profiles, where each IP profile is based on the respective supported coder (see Section 4.13 on page 58).

➢ To configure Coder Group for Microsoft Lync connection:

1. Open the 'Coder Group Settings' page (Configuration tab > VoIP menu > Coders and Profiles > Coder Group Settings).

   Figure 4-19: Coder Group Settings

2. Select Coder Group ID 1.
4. Click Submit.
To configure Coders Group for Skype SIP Trunk connection:

1. Open the 'Coders Group Settings' page (Configuration tab > VoIP menu > Coders And Profiles > Coders Group Settings).

   Figure 4-20: Coders Group Settings

<table>
<thead>
<tr>
<th>Coder Name</th>
<th>Packetization Time</th>
<th>Rate</th>
<th>Payload Type</th>
<th>Silence Suppression</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.729</td>
<td>20</td>
<td>8</td>
<td>8</td>
<td>Disabled</td>
</tr>
<tr>
<td>G.711A-law</td>
<td>20</td>
<td>64</td>
<td>8</td>
<td>Disabled</td>
</tr>
<tr>
<td>G.711U-law</td>
<td>20</td>
<td>64</td>
<td>0</td>
<td>Disabled</td>
</tr>
</tbody>
</table>

2. Select Coder Group ID 2.
4. Click Submit.
4.13 **Step 13: Configure IP Profile**

This step describes how to configure the IP Profile. In this configuration, the IP Profile is used to configure the SRTP/TLS mode and the Coder Group (see Section 4.12 on page 56).

You must configure Microsoft Lync to work in secure mode (SRTP/TLS); while, the Skype SIP trunk is configured in non-secure RTP/TLS mode.

➢ **To configure IP Profile for Microsoft Lync:**

1. Open the 'IP Profile Settings' page (Configuration tab > VoIP menu > Coders And Profiles > IP Profile Settings).

   ![Figure 4-21: IP Profile Settings](image)

   - **Profile ID**: 1
   - **Profile Name**: Lync

   ![Gateway Parameters]

   - **Gateway Parameters**
     - **Fox Signaling Method**: G.711 Transport
     - **Play Ringback Tone to IP**: Don't Play
     - **Enable Early Media**: Enable
     - **Copy Destination Number to Redirect Number**: Disable
     - **Media Security Behavior**: Preferable - Single Media
     - **CNG Detector Mode**: Disable
     - **Modems Transport Type**: Enable Sip Pass
     - **NSE Mode**: Disable
     - **Number of Calls Limit**: 1
     - **Progress Indicator to IP**: Not Configured
     - **Profile Preference**: 1
     - **Coder Group**: Coder Group 1
     - **Remote RTP Base UDP Port**: 0
     - **First Tx DTMF Option**: RFC 2833

2. Select **Profile ID 1**.
4. Set **Coder Group** to ‘Coder Group 1’.
5. Click **Submit**.
To configure IP Profile for Skype SIP Trunk:

1. Open the 'IP Profile Settings' page (Configuration tab > VoIP menu > Coders And Profiles > IP Profile Settings).

![Figure 4-22: IP Profile Settings]

2. Select Profile ID 2.
4. Set Coder Group to ‘Coder Group 2’.
5. Click Submit
4.14 Step 14: Configure IP Group Tables

This step describes how to create IP groups. Each IP group represents a SIP entity in the gateway's network. You need to create IP groups for the following entities:

1. Lync Server 2010 – Mediation Server
2. Skype SIP SIP Trunk

These IP groups are later used by the IP2IP application for routing calls.

➢ To configure IP Group 1:

1. Open the 'IP Group Table' page (Configuration tab > VoIP menu > Control Network > IP Group Table).

![Figure 4-23: IP Group Table 1](image)

2. Set Index to ‘1’.
3. Set Type to ‘SERVER’.
4. Set Proxy Set ID to ‘1’.
To configure IP Group 2:

1. Open the 'IP Group Table' page (Configuration tab > VoIP menu > Control Network> IP Group Table).

![Figure 4-24: IP Group Table 2]

2. Set Index to ‘2’.
3. Set Type to ‘Server’.
4. Set Proxy Set ID to ‘2’.
4.15 Step 15: Configure Routing Rules

This step describes how to configure the IP to IP routing table.

The device IP-to-IP routing rules are configured in the ‘IP to Trunk Group Routing’ and ‘Tel to IP Routing’ tables. Those tables provides enhanced IP-to-IP call routing capabilities for routing received SIP messages such as INVITE messages to a destination IP address. The routing rule must match one of the following input characteristics: Source IP Group, Source Phone Prefix, and/or Source Host Prefix.

It is crucial that you adhere to the following guidelines when configuring your IP-to-IP routing rules:

- Ensure that your routing rules are accurate and correctly defined.
- Ensure that your routing rules from source IP Group to destination IP Group are accurately defined to be eligible for the desired call routing outcome.
- Avoid (if possible) using the asterisk (*) symbol to indicate "any" for a specific parameter in your routing rules. This constitutes a weak routing rule. For strong routing rules, enter specific letter or numeric character values.

To configure inbound IP routing rules:

1. Open the 'IP to Trunk Group Routing Table' page (Configuration tab > VoIP menu > GW and IP to IP > Routing > IP to Trunk Group Routing Table).

2. Calls that arriving from the Microsoft Lync server will be send to the 'Tel to IP Routing Table' (-1) with ‘IP Profile ID’ = 1 and marks as ‘Source IPGroup ID’ = 1.

3. Calls the arriving from Skype will be send to the ‘Tel to IP Routing Table’ (-1) with ‘IP Profile ID’ = 2 and marks as ‘Source IPGroup ID’=2.
To configure outbound IP routing rules:

1. Open the 'Tel to IP Routing Table' page (Configuration tab > VoIP menu > GW and IP to IP > Routing > Tel to IP Routing Table).

Figure 4-26: Outbound IP Routing Table

2. Calls from Source IPGroup ID ‘1’ (e.g., from Microsoft Lync) will be send to ‘Dest. IPGroup ID’ 2 (e.g., To Skype).

3. Calls from Source IPGroup ID ‘2’ (e.g., from Skype) will be send to ‘Dest. IPGroup ID’ 1 (e.g., To Lync).

Note: The Routing configuration may change according to the local deployment topology.
4.16 Step 16 Configure Manipulation Rules

This step describes how to configure the manipulation tables.

The Manipulation Tables submenu allows you to configure number manipulation and mapping of NPI/TON to SIP messages.

Note: Adapt the manipulation table according to your environment dial plan.

➢ To configure Manipulation Tables:

1. Open the 'Manipulation Table' page (Configuration tab > VoIP menu > GW and IP to IP > Manipulations).

![Figure 4-27: Manipulation Tables](image_url)
The following includes examples for number manipulation on destination and source numbers in the Tel-to-IP tables:

➢ To configure destination phone number manipulation rules for Tel-to-IP calls:

Open the ‘Destination Phone Number Manipulation Table for Tel -> IP calls’ page (Configuration tab > VoIP menu > GW and IP to IP > Manipulations submenu > Dest Number Tel > IP).

![Figure 4-38: Destination Phone Number Manipulation Table for Tel-to-IP Calls Page](image)

- **Index #1** defines the destination number manipulation of IP calls from the Skype SIP Trunk. All calls received from Source IP Group 2 (i.e., from the Skype SIP Trunk) and the destination number prefix begins with ‘+’, do not perform any changes to the number.

- **Index #2** defines the destination number manipulation of IP calls from the Skype SIP Trunk. All calls received from Source IP Group 2 (i.e., from Skype SIP Trunk) and the destination number prefix begins with ‘1’, add the ‘+’ prefix to the number.
To configure source phone number manipulation rules for Tel-to-IP calls:

1. Open the ‘Source Phone Number Manipulation Table for Tel-to-IP calls’ page (Configuration tab > VoIP menu > GW and IP to IP > Manipulations submenu > Source Number Tel > IP).

Figure 4-40: Source Phone Number Manipulation Table for Tel -> IP Calls Page

- **Index #1** defines source number manipulation of IP calls from the MS Lync Server. All calls received from Source IP Group 1 (i.e., from the MS Lync Server) for any source number, replace the number with the Account Number (i.e., with 99051000136425 in the above example).
4.17 Step 17: Secure Calls

This step describes how to ensure that incoming calls to the LAN are sourced from valid IP addresses. This action prevents unwanted SIP calls, SIP messages, and/or VoIP spam. This feature is configured according to the device's policy on accepting or blocking SIP calls. You can configure the valid IP addresses by using one of the following methods:

- Using Proxies or Proxy Sets (see Section 4.11 on page 54).
- Using the Tel to IP routing table (see example below).

➢ To configure Secure Calls:

1. Open the 'Advanced Parameters' page (Configuration tab > VoIP menu > SIP Definitions > Advanced Parameters).

   ![Figure 4-28: Advanced Parameters](image)

   2. Set IP Security to ‘Secure All calls’.

➢ To configure allowed IP in Tel to IP Routing Table:

1. Open the ‘Tel to IP Routing Table’ page (Configuration tab > VoIP menu > GW and IP to IP > Routing > Tel to IP Routing Table).
2. Configure the allowed IP-Addresses (see example topology above).
4.18 Step 18: Alternative Routing Reasons

This step describes how to format the Alternative Routing Reasons. A 503 SIP response from the Mediation Server to an INVITE must cause the Mediant 1000 MSBG device to perform a failover. In other words, if the Lync Mediation Server primary proxy server is not responding, an attempt is made to establish communication with the secondary proxy server. For this event to occur, you need to perform the following actions:

- Configure the Reasons for Alternative Routing for Tel-to-IP calls to ‘503 SIP response’ (see below).
- Configure the Lync Mediation Proxy Set for redundancy purposes. See Section 4.11 on page 54.

➢ To configure Alternative Routing Reasons:


![Figure 4-30: Alternative Routing Reasons Table]

2. Set Tel to IP Reason 1 to ‘503’.
3. Click Submit.
4.19 Step 19: Define SIP TLS Connection

This step describes how to configure AudioCodes gateways for implementing a TLS connection with the Microsoft Lync Mediation server. The steps described in this section are essential elements for the configuration of a secure SIP TLS connection.

4.19.1 Step 19-1: Configure VoIP DNS Settings

This step describes how to define the VoIP LAN DNS server, which is a necessary action when a FQDN is configured (as in this scenario configuration, see Section 4.9 on page 9).

➢ To configure the VoIP DNS settings:

1. Open the 'DNS Settings' page (Configuration tab > VoIP menu > DNS > DNS Settings).

2. Set the following parameters:
   • DNS Primary Server IP: <Primary DNS IP-Address> (e.g., 10.15.9.10).

4.19.2 Step 19-2: Configure NTP Server

This step describes how to configure the NTP Server IP address. It is recommended to implement an NTP server (third-party) so that the Mediant 1000 MSBG device receives the accurate current date and time. This is necessary for validating remote parties' certificates.

➢ To configure NTP Settings:

1. Open the 'Application Settings' page (Configuration tab > System menu > Application Settings).

2. Set the NTP Server IP Address to <NTP Server IP-Address> (e.g., 10.15.9.10).
4.19.3 **Step 19-3: Configure a Certificate**

This step describes how to exchange a certificate with the Microsoft Certificate Authority. The certificate is used by the Mediant 1000 MSBG device to authenticate the connection with the management PC (the PC used to manage the Mediant 1000 MSBG using the embedded Web server).

➢ **To configure a certificate:**

1. Open the ‘Certificates’ page (Configuration tab > System menu > Certificates).

![Figure 4-33: Certificates Page](image)

2. In the ‘Subject Name’ field, enter the Media Gateway name i.e **Skype.Lync.local** and then click **Generate CSR**; a Certificate request is generated.

3. Copy the CSR (from the line “----BEGIN CERTIFICATE REQUEST----” to “END CERTIFICATE REQUEST----”) to a text file (such as Notepad), and then save it to a folder on your PC as **certreq.txt**.

**Figure 4-34: Microsoft Certificate Services Web Page**

Welcome

Use this Web site to request a certificate for your Web browser, e-mail client, or other program. By using a certificate, you can verify your identity to people you communicate with over the Web, sign and encrypt messages, and, depending upon the type of certificate you request, perform other security tasks.

You can also use this Web site to download a certificate authority (CA) certificate, certificate chain, or certificate revocation list (CRL), or to view the status of a pending request.

For more information about Certificate Services, see Certificate Services Documentation.

Select a task:
- Request a certificate
- View the status of a pending certificate request
- Download a CA certificate, certificate chain, or CRL

---

5. Click the link Request a Certificate.

**Figure 4-35: Request a Certificate Page**

Request a Certificate

Select the certificate type:
- Web Browser Certificate
- E-Mail Protection Certificate

Or, submit an advanced certificate request.
6. Click the link **Advanced Certificate Request**, and then click **Next**.

**Figure 4-36: Advanced Certificate Request Page**

7. Click the link **Submit a Certificate request by using base64 encoded...**, and then click **Next**.
8. Open the `certreq.txt` file that you created and saved (see Step 3), and then copy its contents to the ‘Base64 Encoded Certificate Request’ text box.

9. Select “Web Server” from the **Certificate Template** drop-down box.

10. Click **Submit**.

11. Choose the ‘Base 64’ encoding option, and then click the link **Download CA certificate**.

12. Save the file as ‘gateway.cer’ in a folder on your PC.


14. Click the link **Download a CA Certificate, Certificate Chain or CRL**.
15. Under the Encoding method group, perform the following:
   a. Select the ‘Base 64’ encoding method option.
   b. Click the link Download CA certificate.

16. Save the file as ‘certroot.cer’ in a folder on your PC.

17. Navigate back (in the Mediant 1000 MSBG device) to the ‘Certificates’ page.

18. In the ‘Certificates’ page, in the ‘Server Certificate’ field, click Browse and select the ‘Gateway.cer’ certificate file that you saved on your local disk (see Step 12), and then click Send File to upload the certificate.
19. In the ‘Certificates’ page, in the ‘Trusted Root Certificate Store’ field, click **Browse** and select the ‘Certroot.cer’ certificate file that you saved on your local disk (see Step 16), and then click **Send File** to upload the certificate.

**Figure 4-39: Certificates Page**

![Certificate Files](image)

20. Save (burn) the Media Gateway configuration and reset the Media Gateway, using the Web interface’s ‘Maintenance Actions’ page (On the Navigation bar, click the **Management** tab, and then in the Navigation tree, select the **Management Configuration** menu, and then choose the **Maintenance Actions**).
4.20 Step 20: Reset the Gateway

After you have completed the gateway configuration as described in the steps above, burn the configuration to the gateway’s flash memory and reset the gateway.

- Click the Reset button to burn the configuration to flash and reset the gateway (ensure that the ‘Burn to FLASH’ field is set to “Yes”).

**Figure 4-40: Reset the GW**

<table>
<thead>
<tr>
<th><img src="image" alt="AudioCodes" /></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Maintenance Actions</strong></td>
</tr>
<tr>
<td><strong>Reset Configuration</strong></td>
</tr>
<tr>
<td>Reset Board</td>
</tr>
<tr>
<td>Burn To FLASH</td>
</tr>
<tr>
<td>Graceful Option</td>
</tr>
<tr>
<td><strong>LOCK / UNLOCK</strong></td>
</tr>
<tr>
<td>Lock</td>
</tr>
<tr>
<td>Graceful Option</td>
</tr>
<tr>
<td>Gateway Operational State</td>
</tr>
<tr>
<td><strong>Save Configuration</strong></td>
</tr>
<tr>
<td>Burn To FLASH</td>
</tr>
</tbody>
</table>

**Note:** Reset with BURN to FLASH is required.
5 Appendix A: AudioCodes INI file

This step shows the Mediant 1000 MSBG device INI file. This file reflects the configuration described in Section 4 on page 37.

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

;Board: Mediant 1000 - MSBG
;Serial Number: 3589366
;Slot Number: 1
;Software Version: 6.20A.022.003
;DSP Software Version: 620AE3 => 620.08
;Board IP Address: 10.15.7.131
;Board Subnet Mask: 255.255.0.0
;Board Default Gateway: 10.15.7.130
;Ram size: 512M Flash size: 64M
;Num of DSP Cores: 12 Num DSP Channels: 48
;Profile: NONE

;Key features:
;Board Type: Mediant 1000 - MSBG
;PSTN Protocols: ISDN IUA=4 CAS ;Coders: G723 G729 GSM-FR G727 ILBC ;E1Trunks=4 ;T1Trunks=4 ;IP Media: Conf VXML VoicePromptAnnounc(H248.9) ;Channel Type: RTP PCI DspCh=240 IPMediaDspCh=240 ;DSP Voice features: EC128mSec AdditionTimeslotSummation FastSlowPlayback BargeIn PatternDetector IpMDetector ;DATA features: Routing Firewall&VPN WAN Advanced-Routing ;Security: IPSEC MediaEncryption StrongEncryption EncryptControlProtocol ;Control Protocols: MSFT MGCP MEGACO SIP SASurvivability SBC=120 ;Default features: ;Coders: G711 G726;

;--------- Mediant-1000 HW components ---------
;
; Slot # : Module type : # of ports : # of DSPs
;----------------------------------------------
; 1 : FALC56 : 2 : 3
; 2 : Empty
; 3 : Empty
; 4 : Empty
; 5 : Empty
; 6 : Empty
;----------------------------------------------

[SYSTEM Params]
DNSPriServerIP = 10.15.9.10
SyslogServerIP = 10.15.45.200
EnableSyslog = 1
NTPServerIP = 10.15.9.10
NTPServerUTCOffset = 7200
PM_VEDSPUtil = '1,43,48,15'
```
[BSP Params]
PCMLawSelect = 3
EnableLANWatchdog = 0

[Analog Params]

[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]
BrokenConnectionEventTimeout = 100
EnableAGC = 1
EnableDSPIPMDetectors = 1
ENABLEMEDIASECURITY = 1
SRTPTxPacketMKISize = 1
FarEndDisconnectSilenceMethod = 2
FarEndDisconnectSilencePeriod = 120
CallProgressTonesFilename = 'usa_tones_13.dat'

[WEB Params]
LogoWidth = '145'
HTTPSCipherString = 'RC4:EXP'
WanMgmtHttpPort = 80
WanMgmtHttpsPort = 443

[SIP Params]
MEDIACHANNELS = 120
ISPROXYUSED = 0
ISREGISTERNEEDED = 0
AUTHENTICATIONMODE = 1
SIPDESTINATIONPORT = 5067
PLAYRBTONE2TEL = 3
SECURECALLSFROMIP = 2
GWDEBUGLEVEL = 5
ENABLEEARLYMEDIA = 1
SIPGATEWAYNAME = 'GW-131.Lync.Local'
STATICNATIP = 195.189.192.151
PROGRESSINDICATOR2IP = -1
PROXYREDUNDANCYMODE = 1
CDRSYSLOGSERVERIP = 0.0.0.0
PSTNALERTTIMEOUT = 180
ISFAXUSED = 2
SIPTRANSPORTTYPE = 2
TLSLOCALSIPPORT = 5067
MEDIASECURITYBEHAVIOUR = 3
REDUNDANTROUTINGMODE = 2
ENABLECONTACTRESTRICTION = 1
FORKINGHANDLINGMODE = 1
ENABLEIP2IPAPPLICATION = 1
ENABLEEARLY183 = 1
FAKERETRYAFTER = 60

[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, 10.15.7.131, 16, 10.15.7.130, 1, Voice;

[ \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 PREFIX ***

;

[ PREFIX ]
FORMAT PREFIX_Index = PREFIX_DestinationPrefix, PREFIX_DestAddress,
PREFIX_SourcePrefix, PREFIX_ProfileId, PREFIX_MeteringCode,
PREFIX_DestPort, PREFIX_SrcIPGroupID, PREFIX_DestHostPrefix,
PREFIX_DestIPGroupID, PREFIX_SrcHostPrefix, PREFIX_TransportType,
PREFIX_SrcTrunkGroupID, PREFIX_DestSRD;
PREFIX 0 = *, , *, 2, 255, 0, 1, , 2, , -1, -1, -1;
PREFIX 1 = *, , *, 1, 255, 0, 2, , 1, , -1, -1, -1;
PREFIX 8 = *, 78.141.179.70, *, 0, 255, 0, -1, , -1, , -1, -1, -1;
PREFIX 9 = *, 10.15.9.11, *, 0, 255, 0, -1, , -1, , -1, -1, -1;

[ \PREFIX ]

;

; *** TABLE NumberMapIp2Tel ***

;

[ NumberMapIp2Tel ]
FORMAT NumberMapIp2Tel_Index = NumberMapIp2Tel_DestinationPrefix,
NumberMapIp2Tel_SourcePrefix, NumberMapIp2Tel_SourceAddress,
NumberMapIp2Tel_NumberType, NumberMapIp2Tel_NumberPlan,
NumberMapIp2Tel_RemoveFromLeft, NumberMapIp2Tel_RemoveFromRight,
NumberMapIp2Tel_LeaveFromRight, NumberMapIp2Tel_Prefix2Add,
NumberMapIp2Tel_Suffix2Add,
NumberMapIp2Tel_IsPresentationRestricted,
NumberMapIp2Tel_SrcTrunkGroupID, NumberMapIp2Tel_SrcIPGroupID;
NumberMapIp2Tel 1 = +999, *, *, 255, 255, 4, 0, 255, +972, , 255, -
1, -1;
NumberMapIp2Tel 2 = +, *, *, 255, 255, 0, 0, 255, , , 255, -1, -1;
NumberMapIp2Tel 3 = *, *, *, 255, 255, 0, 0, 255, +, , 255, -1, -1;

[ \NumberMapIp2Tel ]

;

; *** TABLE SourceNumberMapTel2Ip ***
[ SourceNumberMapTel2Ip ]
FORMAT SourceNumberMapTel2Ip_Index =
SourceNumberMapTel2Ip_DestinationPrefix,
SourceNumberMapTel2Ip_SourcePrefix,
SourceNumberMapTel2Ip_SourceAddress,
SourceNumberMapTel2Ip_NumberType, SourceNumberMapTel2Ip_NumberPlan,
SourceNumberMapTel2Ip_RemoveFromLeft,
SourceNumberMapTel2Ip_RemoveFromRight,
SourceNumberMapTel2Ip_LeaveFromRight,
SourceNumberMapTel2Ip_Prefix2Add, SourceNumberMapTel2Ip_Suffix2Add,
SourceNumberMapTel2Ip_IsPresentationRestricted,
SourceNumberMapTel2Ip_SrcTrunkGroupID,
SourceNumberMapTel2Ip_SrcIPGroupID;
SourceNumberMapTel2Ip 1 = *, , *, 255, 255, 0, 0, 0,
99051000136425, , 0, -1, 1;
SourceNumberMapTel2Ip 2 = *, +, *, 255, 255, 0, 0, 255, , , 255, -
1, -1;
SourceNumberMapTel2Ip 3 = *, *, *, 255, 255, 0, 0, 255, +, , 255, -
1, -1;

[ \SourceNumberMapTel2Ip ]

;  *** TABLE PstnPrefix ***
;
;
[PstnPrefix ]
FORMAT PstnPrefix_Index = PstnPrefix_DestPrefix,
PstnPrefix_TrunkGroupId, PstnPrefix_SourcePrefix,
PstnPrefix_SourceAddress, PstnPrefix_ProfileId,
PstnPrefix_SrcIPGroupID, PstnPrefix_DestHostPrefix,
PstnPrefix_SrcHostPrefix;
PstnPrefix 0 = *, -1, , 10.15.9.11, 1, 1, ;
PstnPrefix 1 = *, -1, , *, 2, 2, , ;

[ \PstnPrefix ]

;  *** TABLE AltRouteCauseTel2Ip ***
;
;
[ AltRouteCauseTel2Ip ]
FORMAT AltRouteCauseTel2Ip_Index =
AltRouteCauseTel2Ip_ReleaseCause;
AltRouteCauseTel2Ip 0 = 503;

[ \AltRouteCauseTel2Ip ]
; *** TABLE ProxyIp ***
;
;
[ ProxyIp ]
FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType,
ProxyIp_ProxySetId;
ProxyIp 0 = FE-Lync.Lync.local:5067, 2, 1;
ProxyIp 1 = sip.skype.com:5060, 0, 2;
[
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_SBCAlternativeDTMFPMethod, IpProfile_SBCAssertIdentity,
IpProfile_AMDSensitivityParameterSuit,
IpProfile_AMDSensitivityLevel, IpProfile_AMDMaxGreetingTime,
IpProfile_AMDMaxPostSilenceGreetingTime,
IpProfile_SBCDiversVersionMode, IpProfile_SBCHistoryInfoMode;
IpProfile 1 = Lync, 1, 0, 2, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0,
0, 1, -1, 1, 0, 3, -1, 1, 4, -1, 1, 1, 0, 0, -1, 0, 0, -1, 0, 0,
0, 0, -1, 0, 8, 300, 400, -1, -1;
IpProfile 2 = Skype, 1, 1, 2, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0,
0, 1, -1, 1, 0, 2, -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 ***
Configuration Note

5. Appendix A: AudioCodes INI file

[ProxySet]
ProxySet 0 = 0, 60, 0, 0, 0, -1;
ProxySet 1 = 1, 60, 1, 1, 0, -1;
ProxySet 2 = 0, 60, 0, 0, 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, Lync, 1, , , 0, -1, 0, -1, 0, 1, 0, -1, -1, -1, -1, -1, -1;
IPGroup 2 = 0, Skype, 2, , , 0, -1, 0, -1, 0, 1, 0, -1, -1, -1, -1, -1, -1;
[\IPGroup]

; *** TABLE Account ***
;
;
[Account]
FORMAT Account_Index = Account_ServedTrunkGroup, Account_ServedIPGroup, Account_ServingIPGroup, Account_Username, Account_Password, Account_HostName, Account_Register, Account_ContactUser, Account_ApplicationType;
Account 1 = -1, 1, 2, 99051000136425, *, sip.skype.com, 1, 99051000136425, 0;
[\Account]

; *** TABLE CodersGroup0 ***
;
FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime, CodersGroup0_rate, CodersGroup0_PayloadType, CodersGroup0_Sce;
CodersGroup0 0 = g711Alaw64k, 20, 0, -1, 0;
CodersGroup0 1 = g711Ulauw64k, 20, 0, -1, 0;

*** TABLE CodersGroup1 ***

FORMAT CodersGroup1_Index = CodersGroup1_Name, CodersGroup1_pTime, CodersGroup1_rate, CodersGroup1_PayloadType, CodersGroup1_Sce;
CodersGroup1 0 = g711Alaw64k, 20, 0, -1, 0;
CodersGroup1 1 = g711Ulauw64k, 20, 0, -1, 0;

*** TABLE CodersGroup2 ***

FORMAT CodersGroup2_Index = CodersGroup2_Name, CodersGroup2_pTime, CodersGroup2_rate, CodersGroup2_PayloadType, CodersGroup2_Sce;
CodersGroup2 0 = g729, 20, 0, -1, 0;
6 Appendix B: Configuring Skype

This step describes how to access and create a Skype account, in the Skype Manager web site.

6.1 Skype Manager

➢ To access the Skype Manager:

1. Open the browser and enter the following URL: https://login.skype.com/bcp/login?message=login_required.
2. Enter username and password.
3. Press Sign me in.

![Figure 6-1: login Page](image)

4. Go to Skype Connect:

![Figure 6-2: Main Page](image)
To create new account:

1. Press Create a new profile.

   Figure 6-3: Skype Connect page

2. Choose profile name and press Next.

   Figure 6-4: Create Profile Page
3. Enter **Profile Settings** page to proceed configuration.

4. Press **Buy a channel Subscription** to activate this profile.

   **Figure 6-5: Profile Settings Page**

   ![Profile Settings Page](image)

5. Enter the number of channels.

   **Figure 6-6: Channel Subscription Page**

   ![Channel Subscription Page](image)
6. Allow outgoing calls (purchase)

Figure 6-7: Outgoing Calls Page

7. Set up Caller ID.

Figure 6-8: Caller ID Page
8. Add PSTN number (for incoming calls).

**Figure 6-9: Incoming Calls Page**

9. View the Skype Account And settings

**Figure 6-10: Account page**

Figure 6-11 Profile Page
11. View the Authentication details.
12. Copy the authentication information into the account table on the Mediant 1000 MSBG (see Section 4.10 on page 53).

**Figure 6-12 Authentication page**