Avaya Solution & Interoperability Test Lab

Configuring ISDN-PRI connectivity between Avaya Communication Manager and the Avaya Meeting Exchange S6100 Conferencing Server - Issue 1.0

Abstract

These Application Notes present the procedures for configuring ISDN-PRI connectivity between Avaya Communication Manager and the Avaya Meeting Exchange S6100 Conferencing Server. ISDN-PRI connectivity is enabled via utilization of an AudioCodes TP-260/SIP Media Gateway. This configuration leverages the flexibility offered by Avaya Communication Manager to support a rich set of conferencing options provided by Avaya Meeting Exchange.

1. Introduction

These Application Notes present the procedures for configuring ISDN-PRI connectivity between Avaya Communication Manager and the Avaya Meeting Exchange S6100 Conferencing Server. ISDN-PRI connectivity is enabled via utilization of an AudioCodes TP-260/SIP Media Gateway. In this configuration, the AudioCodes Media Gateway is connected to Avaya Communication Manager via ISDN-PRI and Avaya Meeting Exchange via SIP.

This configuration leverages the inherent flexibility of protocols supported on Avaya Communication Manager by enabling any station or trunk type associated with Avaya Communication Manager to interoperate with Avaya Meeting Exchange. Thus, this configuration will allow access to a rich selection of conferencing features supported on Avaya Meeting Exchange.

The following call flows for accessing a conference on Avaya Meeting Exchange have been verified:

- DirectCallFlow; where conference participants Dial-In and enter a conference as moderator, without entering a passcode.
- BasicCallFlow; where conference participants Dial-In and enter a conference via passcode.

The following features have been verified for adding participants to an active conference:

- Blast Dial; where a moderator on a conference call can enter a feature access code (e.g., *9, see Section 6 Step 6.2) to Dial-Out to a pre-provisioned list of one or more participants. The participants have the option of joining the conference call.
- Originator Dial-Out; where a moderator on a conference call can Dial-Out and add a participant to the conference call.

Note the convention for Dial-In/Dial-Out assigns Avaya Meeting Exchange as the point of reference; e.g., *Dial-In to Avaya Meeting Exchange*, *Dial-Out from Avaya Meeting Exchange*.

These Application Notes will provide the administrative steps for configuring the following equipment in support of the configuration depicted in **Figure 1**:

- Avaya Communication Manager
- Avaya Meeting Exchange
- AudioCodes Media Gateway, which is plugged into the PCI slot on the Avaya Meeting Exchange server (for power only) and functions as a SIP/ISDN-PRI media gateway.

The connectivity from Avaya Communication Manager to Avaya Meeting Exchange is provided by AudioCodes Media Gateway. Avaya Communication Manager is directly connected to the AudioCodes Media Gateway via an ISDN-PRI trunk, and Avaya Meeting Exchange is directly connected to the AudioCodes Media Gateway via a SIP trunk.

Note: In this configuration, Avaya SIP Enablement Services is strictly utilized for registering SIP endpoints.

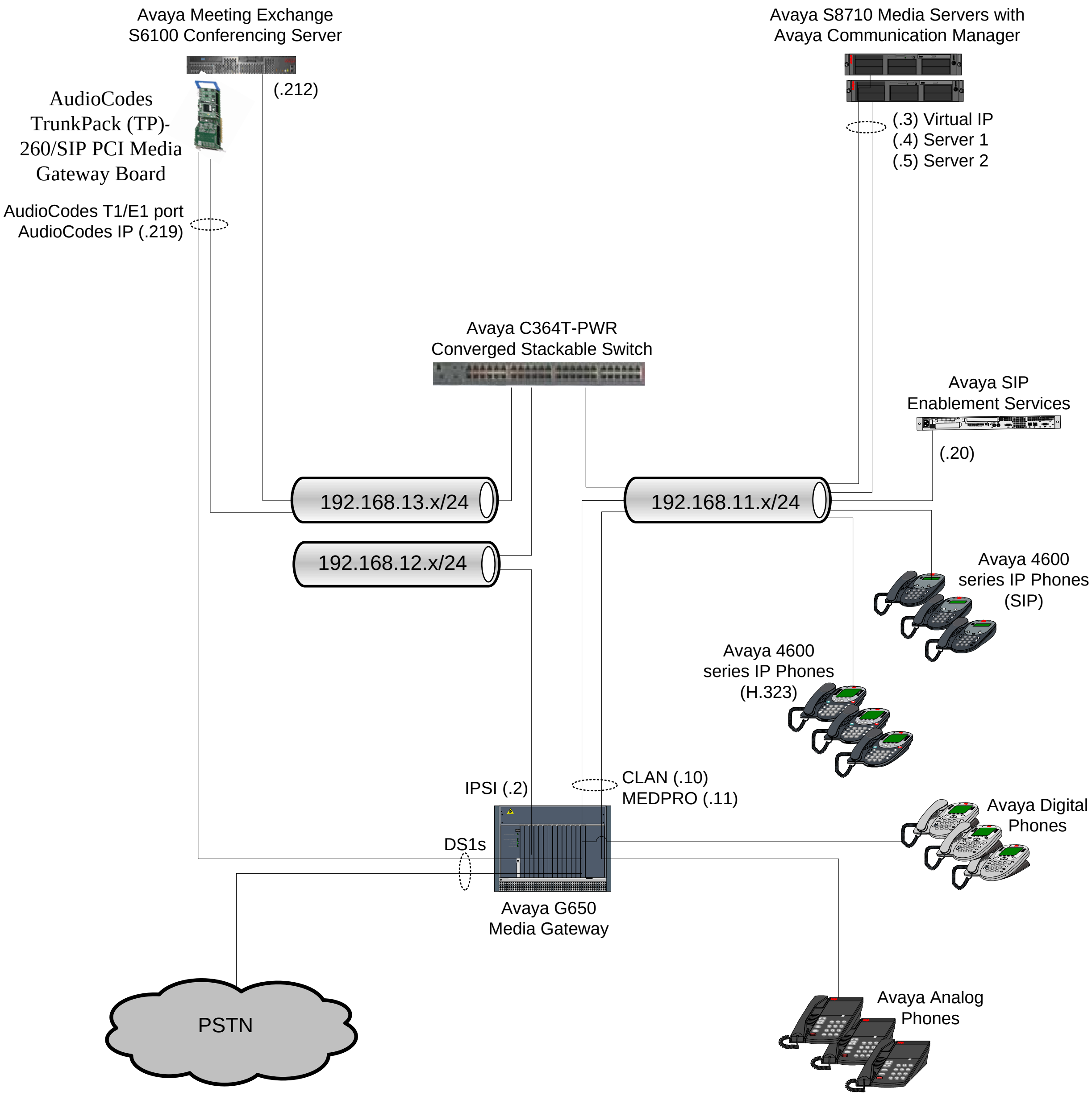


Figure 1: Network Configuration

2. Equipment and Software Validated

The following equipment and software versions were used for the configuration:

Equipment	Software
Avaya S8710 Media Servers	Avaya Communication Manager 3.1 (R013x.01.0.628.6)
Avaya G650 Media Gateway • Avaya TN2312BP (IPSI) • Avaya TN799DP (C-LAN) • Avaya TN2302AP (MEDPRO) • Avaya TN464F (DS1)	 HW12 FW031 HW01 FW017 HW20 FW112 000010
Avaya Meeting Exchange S6100 Conferencing Server • AudioCodes TrunkPack (TP)-260/SIP PCI Media Gateway Board	2.0.22.2 V4.40.240.454
Avaya SIP Enablement Services	3.1 (03.1-03.1.018.0)
Avaya C364T-PWR Converged Stackable Switch	V4.5.14
Avaya 4620 IP Telephones	2.3 (H.323)
Avaya 4602 IP Telephones	2.2 (SIP)
Avaya Analog Telephones	--
Avaya Digital Telephones	--

Table 1: Hardware and Software Versions

3. Avaya Communication Manager Configuration

This section describes the steps for configuring Avaya Communication Manager to interoperate with the AudioCodes Media Gateway via ISDN-PRI connectivity (see **Figure 1**).

The following configuration of Avaya Communication Manager was performed using the System Access Terminal (SAT). After completion of the configuration in this section, perform a **save translation** command to make the changes permanent.

Step	Description
3.1	Verify licensing for **OPTIONAL FEATURES** Issue the command "**display system-parameters customer-options**", and proceed to Page 3 and verify that the system is licensed to utilize Automatic Alternate Routing (**AAR**) without Feature Access Code (**FAC**). **Note**: **AAR** without **FAC** allows a direct access to the **AAR ANALYSIS TABLE** (see **Step 3.10**) upon matching a **Dialed String** in the **DIAL PLAN ANALYSIS TABLE** (see **Step 3.9**).

```
Page 3 of 10
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? n          Audible Message Waiting? n
        Access Security Gateway (ASG)? n              Authorization Codes? n
        Analog Trunk Incoming Call ID? n Backup Cluster Automatic Takeover? n
 A/D Grp/Sys List Dialing Start at 01? n                       CAS Branch? n
Answer Supervision by Call Classifier? n                         CAS Main? n
                                  ARS? y                Change COR by FAC? n
                 ARS/AAR Partitioning? y  Computer Telephony Adjunct Links? n
          ARS/AAR Dialing without FAC? y   Cvg Of Calls Redirected Off-net? n
          ASAI Link Core Capabilities? n                      DCS (Basic)? n
          ASAI Link Plus Capabilities? n                DCS Call Coverage? n
       Async. Transfer Mode (ATM) PNC? n               DCS with Rerouting? n
  Async. Transfer Mode (ATM) Trunking? n
            ATM WAN Spare Processor? n    Digital Loss Plan Modification? n
                                 ATMS? n                          DS1 MSP? n
                 Attendant Vectoring? n             DS1 Echo Cancellation? n



        (NOTE: You must logoff & login to effect the permission changes.)
```

Step	Description
3.2	Proceed to Page 4 on the **OPTIONAL FEATURES** form and verify: • The system is licensed for **ISDN-PRI**.

```
Page 4 of 10
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                                IP Stations? y
           Enable 'dadmin' Login? y                 Internet Protocol (IP) PNC? n
           Enhanced Conferencing? y                          ISDN Feature Plus? n
                  Enhanced EC500? y              ISDN Network Call Redirection? n
    Enterprise Survivable Server? n                            ISDN-BRI Trunks? n
       Enterprise Wide Licensing? n                                  ISDN-PRI? y
              ESS Administration? n                 Local Survivable Processor? n
          Extended Cvg/Fwd Admin? n                        Malicious Call Trace? n
     External Device Alarm Admin? n                   Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? n                   Multifrequency Signaling? y
      Global Call Classification? n Multimedia Appl. Server Interface (MASI)? n
             Hospitality (Basic)? y          Multimedia Call Handling (Basic)? y
 Hospitality (G3V3 Enhancements)? n       Multimedia Call Handling (Enhanced)? y
                       IP Trunks? y

           IP Attendant Consoles? n
         (NOTE: You must logoff & login to effect the permission changes.)
```

Step	Description
3.3	Proceed to Page 8 on the **OPTIONAL FEATURES** form and verify: • The system is licensed for **QSIG**.

```
Page 8 of 10
                             QSIG OPTIONAL FEATURES

                                          Basic Call Setup? y
                              Basic Supplementary Services? y
                                     Centralized Attendant? n
                                     Interworking with DCS? n
                          Supplementary Services with Rerouting? n
                                  Transfer into QSIG Voice Mail? n
                                        Value-Added (VALU)? n
```

Step	Description
3.4	Configure a **DS1 CIRCUIT PACK**. Place a DS1 Board in cabinet **Location**: **1A08**. Issue the command "**add ds1 1a08**"; and administer settings as per below. **Note**: The **Peer Protocol** is configured as **Q-SIG**. The QSIG protocol is based on the ISDN Q.931 standard and provides transparent support for supplementary PBX services between Avaya Communication Manager and AudioCodes Media Gateway.

```
Page 1 of 2
                                DS1 CIRCUIT PACK

            Location: 01A08                          Name: DS1 to S6100
            Bit Rate: 1.544                   Line Coding: b8zs
   Line Compensation: 1                      Framing Mode: esf
      Signaling Mode: isdn-pri
             Connect: pbx                       Interface: peer-master
   TN-C7 Long Timers? n                     Peer Protocol: Q-SIG
Interworking Message: PROGress                       Side: a
Interface Companding: mulaw                          CRC? n
           Idle Code: 11111111
                              DCP/Analog Bearer Capability: 3.1kHz

                                          T303 Timer(sec): 4


      Slip Detection? n                 Near-end CSU Type: other
```

Step	Description
3.5	Configure an ISDN-PRI **SIGNALING GROUP**. Issue the command "**add signaling-group <n>**", where **n** is an unallocated Signaling Group number; and administer settings as per below. • The **Group Type** is configured as **isdn-pri**, and will utilize the DSI Circuit Pack configured in **Step 3.4**. • The **Primary D-Channel** is set to **01A0824**, which is channel **24** on Board **01A08**.

```
Page 1 of 1
                                SIGNALING GROUP

 Group Number: 13             Group Type: isdn-pri
                   Associated Signaling? y          Max number of NCA TSC: 0
                      Primary D-Channel: 01A0824     Max number of CA TSC: 0
                                                   Trunk Group for NCA TSC:
       Trunk Group for Channel Selection:
          Supplementary Service Protocol: b
```

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.6</td><td>Configure an ISDN TRUNK GROUP.

Issue the command “add trunk-group <n>”, where n is an unallocated Trunk Group number; and administer settings as per below.
The settings for the Group Type and Carrier Medium are consistent with the Signaling Group provisioned in Step 3.5.
The setting for the Trunk Access Code (TAC) is a number that is consistent with the existing dial plan (see Step 3.9).
<pre>
Page 1 of 21
                                TRUNK GROUP

Group Number: 13   Group Type: isdn   CDR Reports: y
  Group Name: S6100 ISDN PRI   COR: 1   TN: 1   TAC: 113
   Direction: two-way   Outgoing Display? n   Carrier Medium: PRI/BRI
 Dial Access? n   Busy Threshold: 255   Night Service:
Queue Length: 0
Service Type: tie   Auth Code? n   TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
</pre>
</td></tr>
</table>

Step	Description
3.7	Proceed to Page 2 of the **TRUNK GROUP** form and administer hunting as per below. When ISDN-PRI interfaces are used, it is acceptable for both ends to have the **Trunk Hunt** fields administered as **cyclical**, but if one end is administered as **ascend**, the other end must be administered as **descend**. This helps avoid the possibility of glare conditions, where glare is defined as follows: Glare occurs when both sides of an ISDN interface select the same B-channel for call initiation. For example, a user side of an interface selects the B-channel for an outgoing call and, before Avaya Communication Manager receives and processes the SETUP message, the server also selects the same B-channel for call origination. To reduce glare probability, the network needs to be administered so both sides of the interface select channels from opposite ends of facilities. This is called linear hunting, ascending or descending. For example, on a 23B+D trunk group, the user side could be administered to select B-channels starting at channel 23 while the network side would be administered to start selecting at channel 1. Using the same example, if channel 22 is active but channel 23 is idle, the user side should select channel 23 for re-use. For these Application Notes, Avaya Communication Manager is administered as **ascend**; while the other end (the AudioCodes Media Gateway, see **Section 5, Step 5.7**) is configured to hunt as **Descending**.

```
Page 2 of 21
      Group Type: isdn

TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: a     Digit Handling (in/out): enbloc/enbloc

            Trunk Hunt: ascend
                                                   Digital Loss Group: 13
Incoming Calling Number - Delete:     Insert:                Format:
              Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0
```

Step	Description
3.8	Proceed to Page 5 of the **TRUNK GROUP** form and administer the members for the **TRUNK GROUP** as per below. • The values for the **Port** field are resources allocated from **DS1 CIRCUIT PACK 01A08**, provisioned in **Step 3.4**. • Enter the number of the Signaling Group provisioned in **Step 3.5** in the **Sig Grp** field.

```
Page 5 of 21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/23
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  23

       Port    Code Sfx Name        Night           Sig Grp
  1: 01A0801  TN464  F                                13
  2: 01A0802  TN464  F                                13
  3: 01A0803  TN464  F                                13
  4: 01A0804  TN464  F                                13
  5: 01A0805  TN464  F                                13
  6: 01A0806  TN464  F                                13
  7: 01A0807  TN464  F                                13
  8: 01A0808  TN464  F                                13
  9: 01A0809  TN464  F                                13
 10: 01A0810  TN464  F                                13
 11: 01A0811  TN464  F                                13
 12: 01A0812  TN464  F                                13
 13: 01A0813  TN464  F                                13
 14: 01A0814  TN464  F                                13
 15: 01A0815  TN464  F                                13
 16: 01A0816  TN464  F                                13
 17: 01A0817  TN464  F                                13
 18: 01A0819  TN464  F                                13
 19: 01A0818  TN464  F                                13
 20: 01A0820  TN464  F                                13
 21: 01A0821  TN464  F                                13
 22: 01A0822  TN464  F                                13
 23: 01A0823  TN464  F                                13
```

Step	Description
3.9	Configure the **DIAL PLAN ANALYSIS TABLE** Issue the command "**change dialplan analysis**". Add an entry in the table to treat any digit string of **3** digits in **Total Length** with a leading **Dialed String** of **4** as a **Call Type** of **aar**.

```
Page 1 of 12
                             DIAL PLAN ANALYSIS TABLE
                                                            Percent Full:    1

      Dialed  Total  Call      Dialed  Total  Call      Dialed  Total  Call
      String  Length Type      String  Length Type      String  Length Type
        0       1    attd
        1       3    dac
        2       5    ext
        3       5    ext
        4       3    aar
        5       3    aar
        6       3    ext
        7       4    ext
        7       5    ext
        8       1    fac
        9       1    fac
        *       3    fac
        #       3    fac
```

Step	Description
3.10	Configure the **AAR ANALYSIS TABLE** Issue the command "**change aar analysis**". Add entries in the table to send the following **Dialed Strings** to **Route Pattern 13**. • Dialed String **413** will be used by Avaya Meeting Exchange for **BasicCallFlow** (see **Section 4, Step 4.4**). • Dialed String **444** will be used by Avaya Meeting Exchange for **DirectCallFlow** (see **Section 4, Steps 4.5 and 4.11**).

```
Page 1 of 2
                             AAR DIGIT ANALYSIS TABLE
                                                            Percent Full:    1

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    401                   3    3     1        aar          n
    412                   3    3     12       aar          n
    413                   3    3     13       aar          n
    444                   3    3     13       aar          n
```

Step	Description
3.11	Configure a **ROUTE PATTERN** Issue the command "**change route-pattern <n>**", where **n** is the number of the route pattern administered in **Step 3.10**. Add an entry in the table to utilize the Trunk Group provisioned in **Step 3.6**.

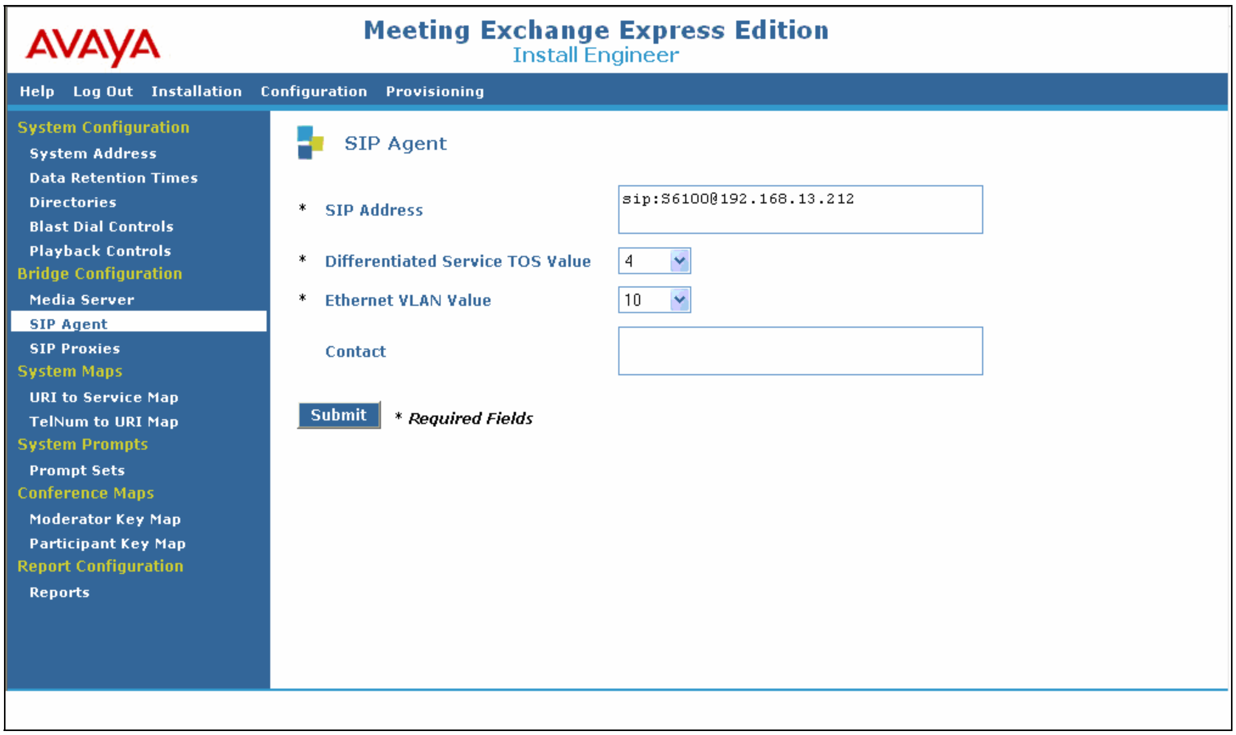
```
Page 1 of 3
                    Pattern Number: 13  Pattern Name: S6100 PRI
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 13   0                    0                                        n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 3 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                        none
 2: y y y y y n  n            rest                                        none
 3: y y y y y n  n            rest                                        none
 4: y y y y y n  n            rest                                        none
 5: y y y y y n  n            rest                                        none
 6: y y y y y n  n            rest                                        none
```

4. Avaya Meeting Exchange Configuration

This section describes the steps for configuring Avaya Meeting Exchange to interoperate with the AudioCodes Media Gateway via SIP connectivity (see **Figure 1**).

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.1</td><td>Verify Licensing as follows:
Avaya Meeting Exchange uses Avaya Web License Manager (WebLM) to support licensing. Open a web browser and enter the following URL:
http://<IP Address of Avaya Meeting Exchange>/WebLM.
Log in to the WebLM server with the appropriate credentials, and verify Avaya Meeting Exchange is licensed for appropriate Number of Meeting Exchange Groupware Edition Ports.
Note: Each conference participant on Avaya Meeting Exchange requires one port for the duration they are on a conference call. The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

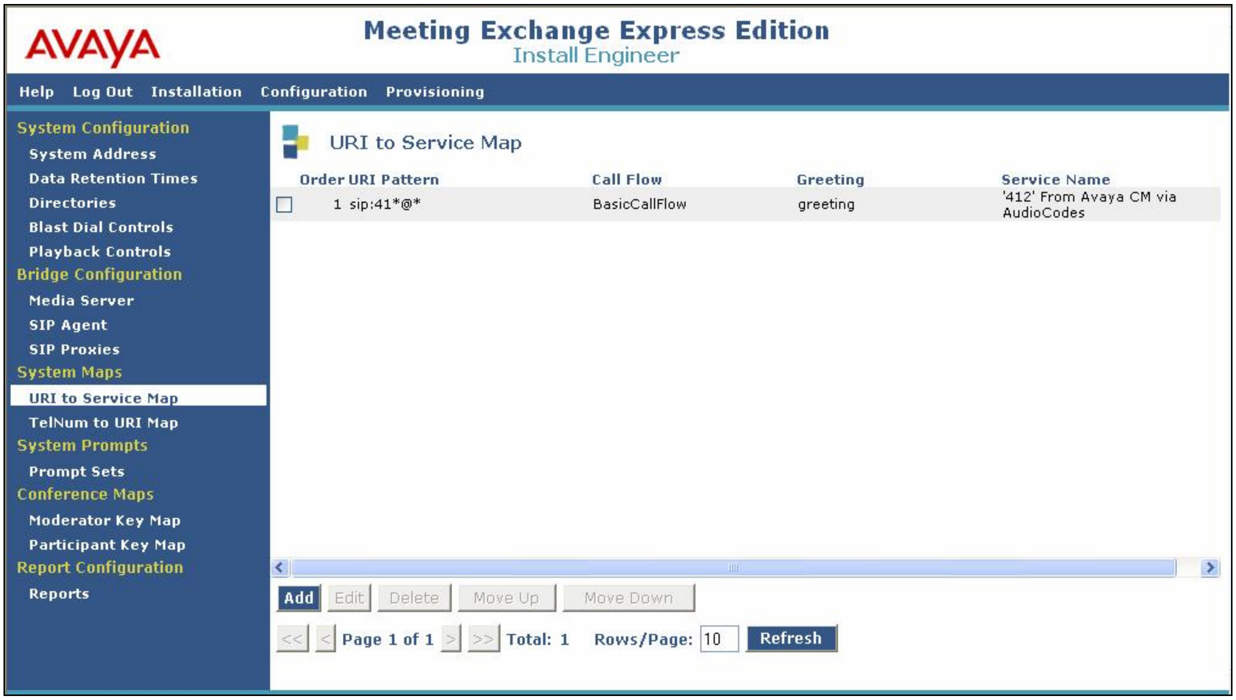

</td></tr>
</table>

Step	Description
4.2	Administer settings for Avaya Meeting Exchange as follows: • Open a web browser and enter the following URL: **http://<IP Address of Avaya Meeting Exchange>** • Log in to Avaya Meeting Exchange with the appropriate credentials.
4.3	Configure settings that relate to the existence of Avaya Meeting Exchange within the SIP network by administering **SIP Agent** parameters as follows: • Click **Configuration** from the S6100 web interface toolbar. • Click **SIP Agent** from the **Configuration** menu. • Add a **SIP Address** for Avaya Meeting Exchange. • When finished, click the **Submit** button. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.4</td><td>To associate incoming calls to Avaya Meeting Exchange with a corresponding Call Flow, add a URI to Service Map entry as follows:

Click URI to Service Map from the Configuration menu.
Click the Add button.

Note: There is an entry for a BasicCallFlow already provisioned. Step 4.5 describes how to provision a new call flow (e.g., DirectCallFlow).

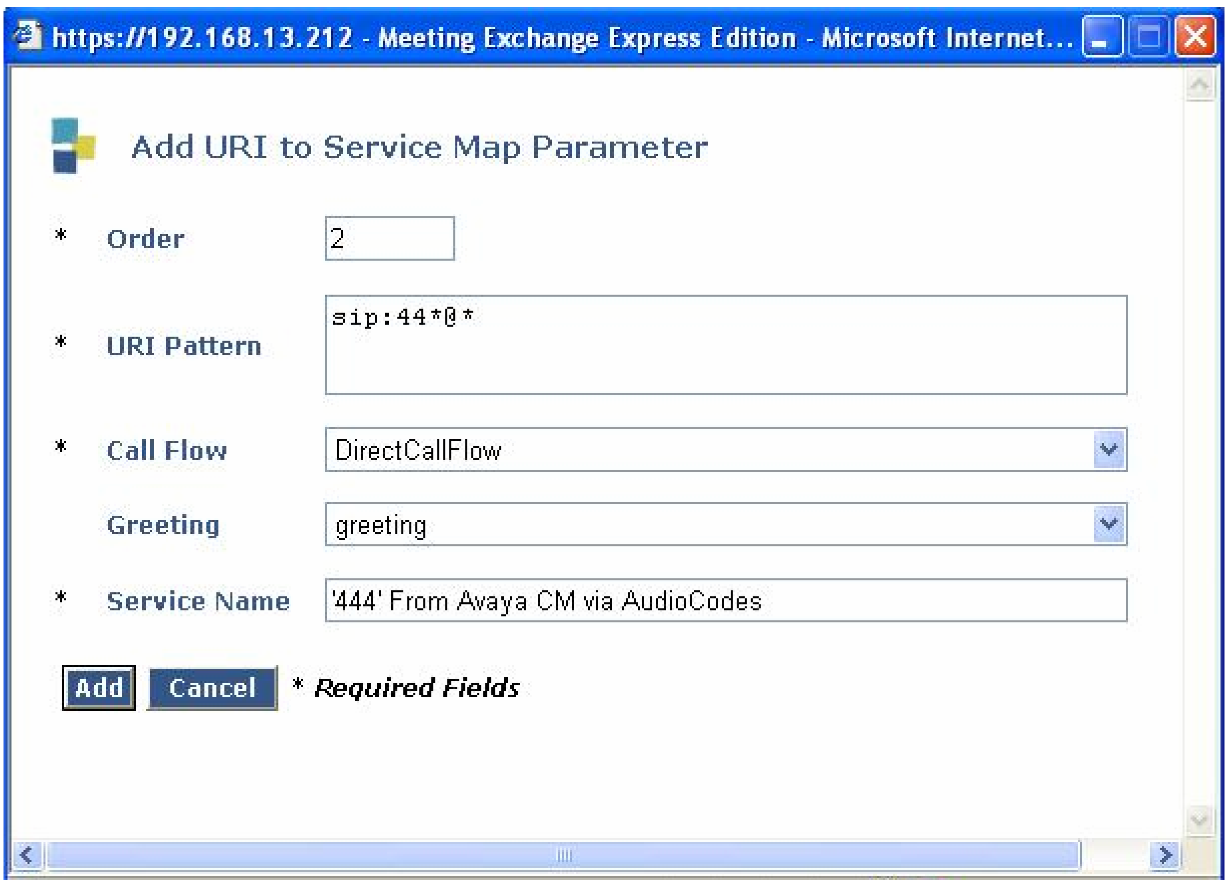


</td></tr>
</table>

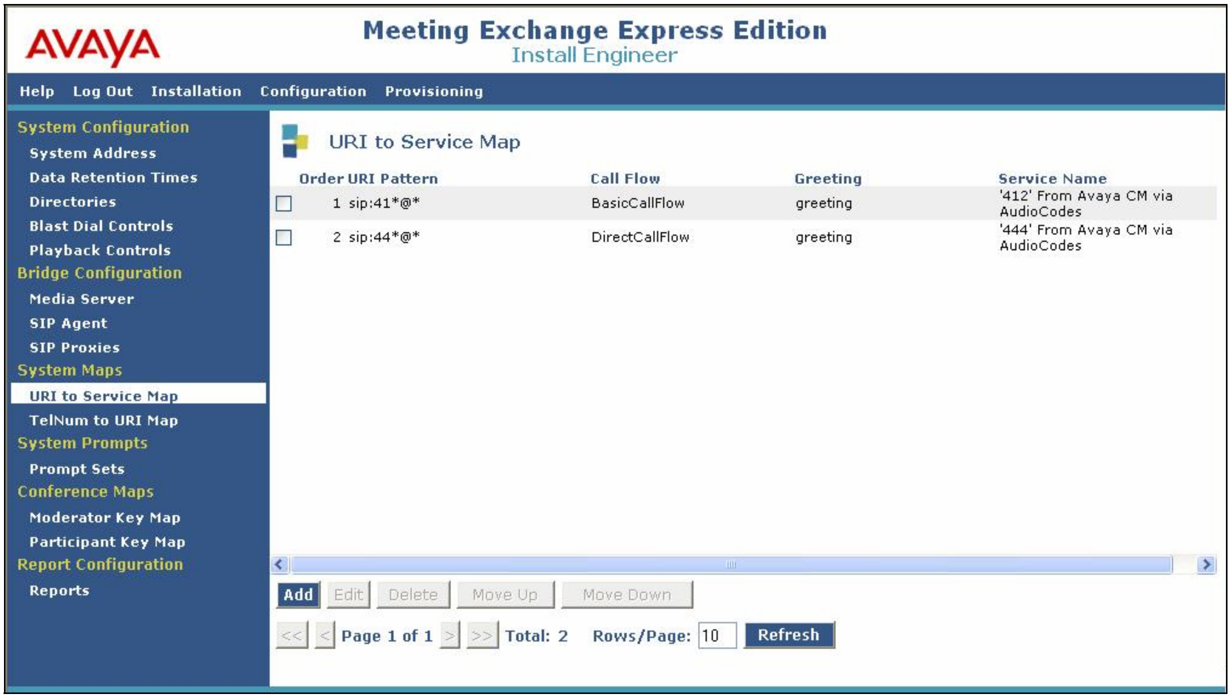
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.5</td><td>

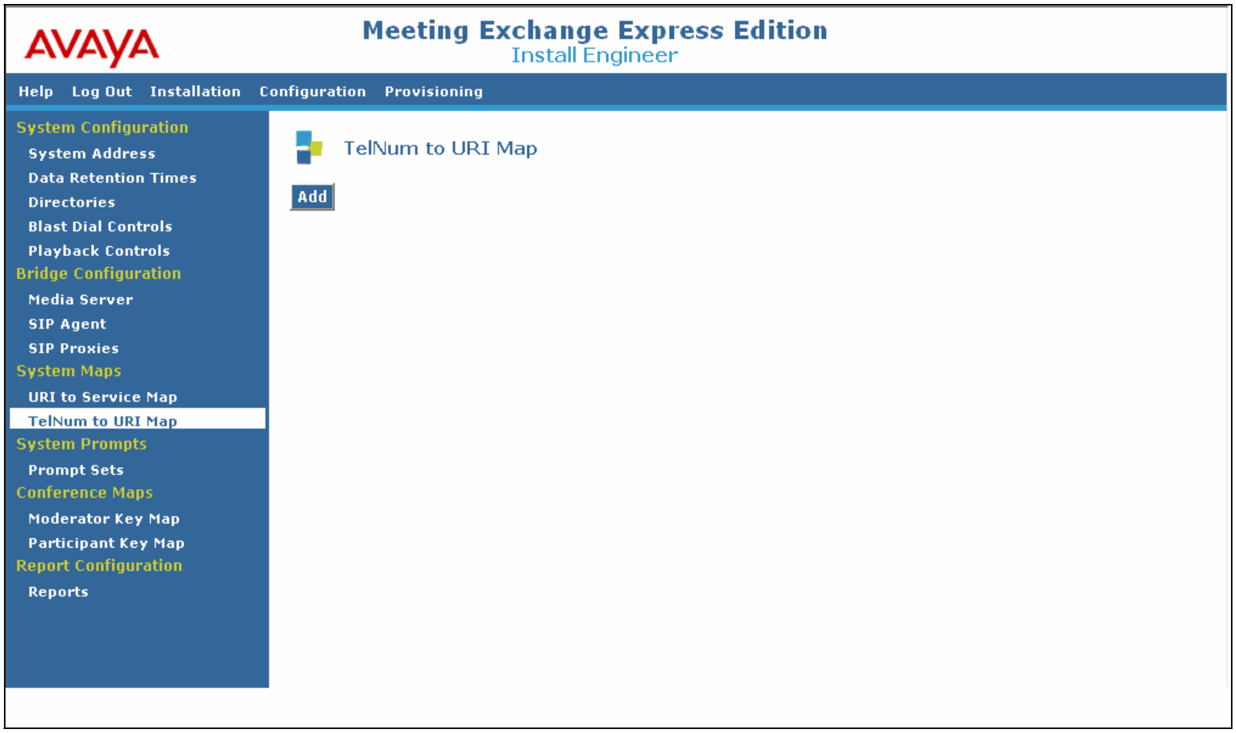
Configure a **URI to Service Map Parameter** for a **DirectCallFlow** as follows:

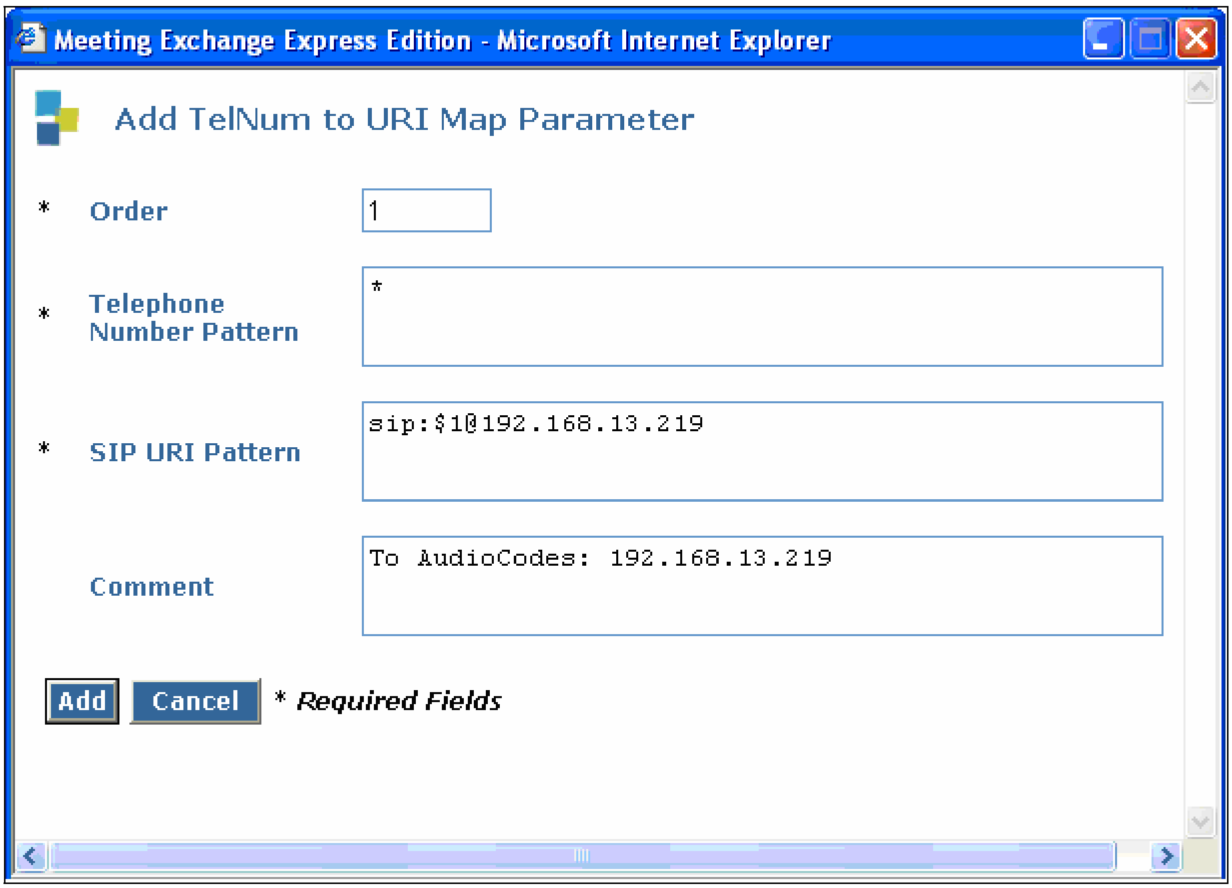
- The **Order** field is left at the default setting. It is defaulted to **2** due to the existing **BasicCallFlow** entry in the table (see **Step 4.4**).
  Note: Avaya Meeting Exchange parses **System Maps** searching for pattern matches in descending order; terminating the search once a pattern is matched. For these Application Notes, **Order** is irrelevant as the patterns for **DirectCallFlow** and **BasicCallFlow** (see **Step 4.4**) are mutually exclusive.
- Add a **URI Pattern** to allow Dial-In to Avaya Meeting Exchange from Avaya Communication Manager via the AudioCodes Media Gateway by matching the pattern of incoming SIP URIs in SIP INVITE messages from the AudioCodes Media Gateway. For example, the AudioCodes Media Gateway sends the following URI: **sip:444@192.168.13.211;user=phone**. The URI Pattern is configured to match **sip:44*@***, which will match **sip:44** and any string until the **@** is reached, then any string following the **@**.
- The **Service Name** field is a descriptive label.
- When finished, click the **Add** button.

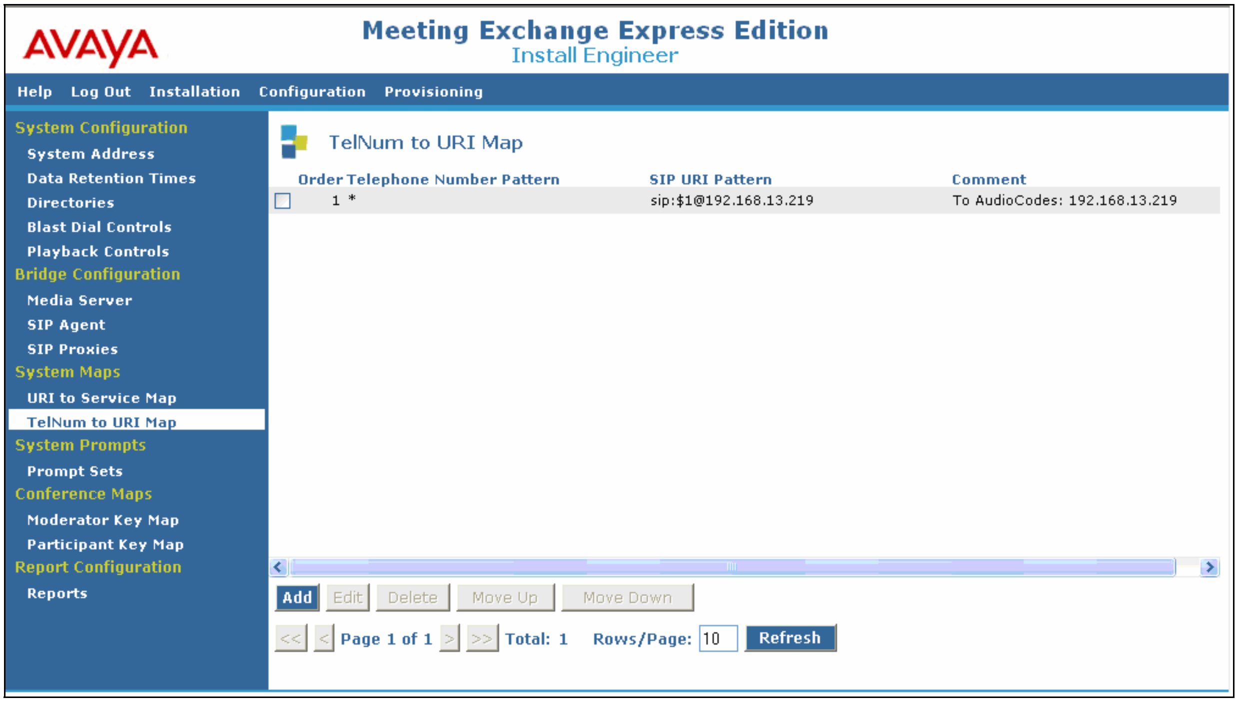


</td></tr>
</table>

Step	Description
4.6	The **URI to Service Map** entry is successfully added. 

Step	Description
4.7	To configure routing of outbound call from Avaya Meeting Exchange, add a **TelNum to URI Map** entry as follows: • Click **TelNum to URI Map** from the **Configuration** menu. • When finished, click the **Add** button. 

Step	Description
4.8	Configure a **TelNum to URI Map Parameter** as follows: • Add a **Telephone Number Pattern** to allow for Dial-Out from Avaya Meeting Exchange. **Note**: The configuration for these Application Notes sends all Dial-Out traffic (***** = match all) to the AudioCodes Media Gateway (**192.168.13.219**). • The **Comment** field is a descriptive label. • When finished, click the **Add** button. 

Step	Description
4.9	The **TelNum to URI Map** entry is successfully added. 
4.10	Following all updates to Avaya Meeting Exchange via the web browser, reboot Avaya Meeting Exchange as follows: • Log in to the Avaya Meeting Exchange Server console to access the command line interface with the appropriate credentials. • At the command prompt, enter the command: **init 6**. `[S6100]> init 6`

Step	Description
4.11	To utilize the **DirectCallFlow** provisioned in **Step 4.5**, administer an Account CSV file as follows: • If not already logged on, log in to the Avaya Meeting Exchange Server console to access the command line interface with the appropriate credentials. • Create an Account CSV file with the format of the **myAccount.csv** shown below. The **myAccount.csv** file is correlated to the **URI Pattern** provisioned in **Step 4.5** via the **def_modpass_code** entry.
	[S6100]> cat /usr/tmp/csvFiles/**myAccount.csv** account_note,def_confpass_code,**def_modpass_code**,mx_conf_size,mx_confdur_mins,import_tag,disabled_ind,logon_password,contact_name,contact_phone,contact_email,import_tag,conf_profile_id,message_profile_id "DirectDial_444","1444","**444**","250","30","444_Tag","f","444","CSV Account 444","1234551444","csv@account444.com","CSV_Company_5","5",""
	• Write the **myAccount.csv** file to the database by running the **bulk-loader.sh** utility as follows: o cd to **/usr/crystal/bulkloader** o At the command prompt, enter the command: **sh bulk-loader.sh -A/usr/tmp/csvFiles/myAccount.csv**
	[S6100]> **sh bulk-loader.sh -A/usr/tmp/csvFiles/myAccount.csv** com.avaya.crystal.common.Logger.LogDir not set, setting log location to default ... com.avaya.crystal.common.Logger.LogDir set to: /usr/crystal/config/../logs Log configuration file [/usr/crystal/config/CrystalLog.xml] loaDING. Log configuration file [/usr/crystal/config/CrystalLog.xml] was loaded. Write Account File :**All 1 row(s) were successfull**

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.12</td><td>To enable the Blast Dial feature, administer a Blast Dial CSV file as follows:
Create a Blast Dial CSV file with the format of the myBlastDial.csv shown below.
The myBlastDial.csv file is correlated to the myAccount.csv file provisioned in Step 4.11 via the reservation_import_tag entry.
The contact_phone variable is the number dialed when the Blast Dial feature is invoked.</td></tr>
<tr><td></td><td><pre>[S6100]> cat /usr/tmp/csvFiles/myBlastDial.csv
reservation_import_tag,contact_name,contact_phone,contact_email,person_import_tag
"444_Tag","BlastDialContact4","31001","csv@blastdialcontact4.com","PersonImportTag4"
"444_Tag","BlastDialContact5","32001","csv@blastdialcontact5.com","PersonImportTag5"
"444_Tag","BlastDialContact6","32002","csv@blastdialcontact6.com","PersonImportTag6"
"444_Tag","BlastDialContact7","33002","csv@blastdialcontact7.com","PersonImportTag7"</pre></td></tr>
<tr><td></td><td>Write the myBlastDial.csv file to the database by running the bulk-loader.sh utility as follows:
cd to /usr/crystal/bulkloader
At the command prompt, enter the command:
sh bulk-loader.sh -B/usr/tmp/csvFiles/ myBlastDial.csv</td></tr>
<tr><td></td><td><pre>[S6100]> sh bulk-loader.sh -B/usr/tmp/csvFiles/myBlastDial.csv
com.avaya.crystal.common.Logger.LogDir not set, setting log location to default ...
com.avaya.crystal.common.Logger.LogDir set to: /usr/crystal/config/../logs
Log configuration file [/usr/crystal/config/CrystalLog.xml] loaDING.
Log configuration file [/usr/crystal/config/CrystalLog.xml] was loaded.
Write BlastDial File :All 4 row(s) were successfull</pre></td></tr>
<tr><td></td><td></td></tr>
</table>

5. Configure the AudioCodes Media Gateway

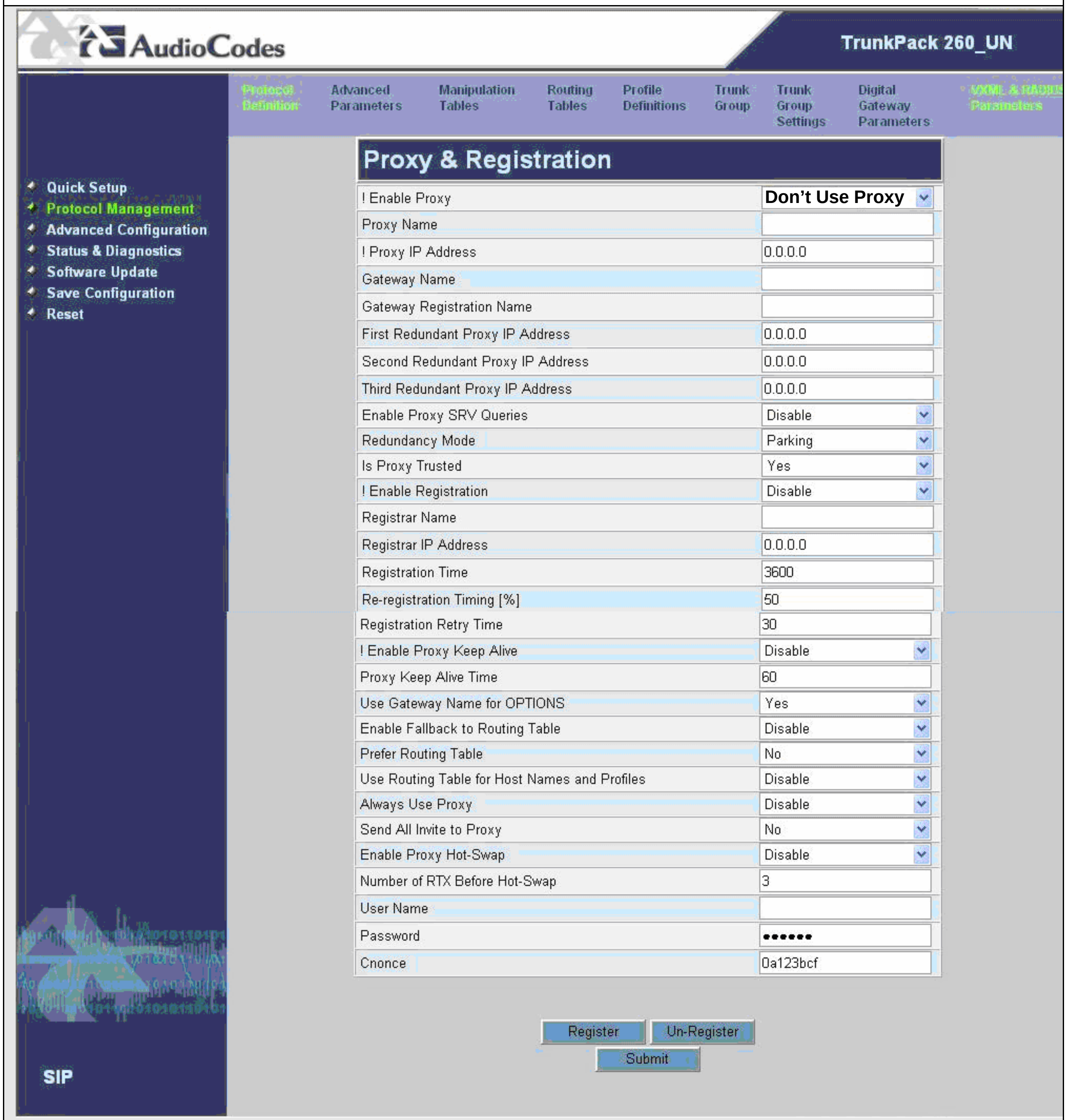
This section describes the steps for configuring the AudioCodes Media Gateway to interoperate with:

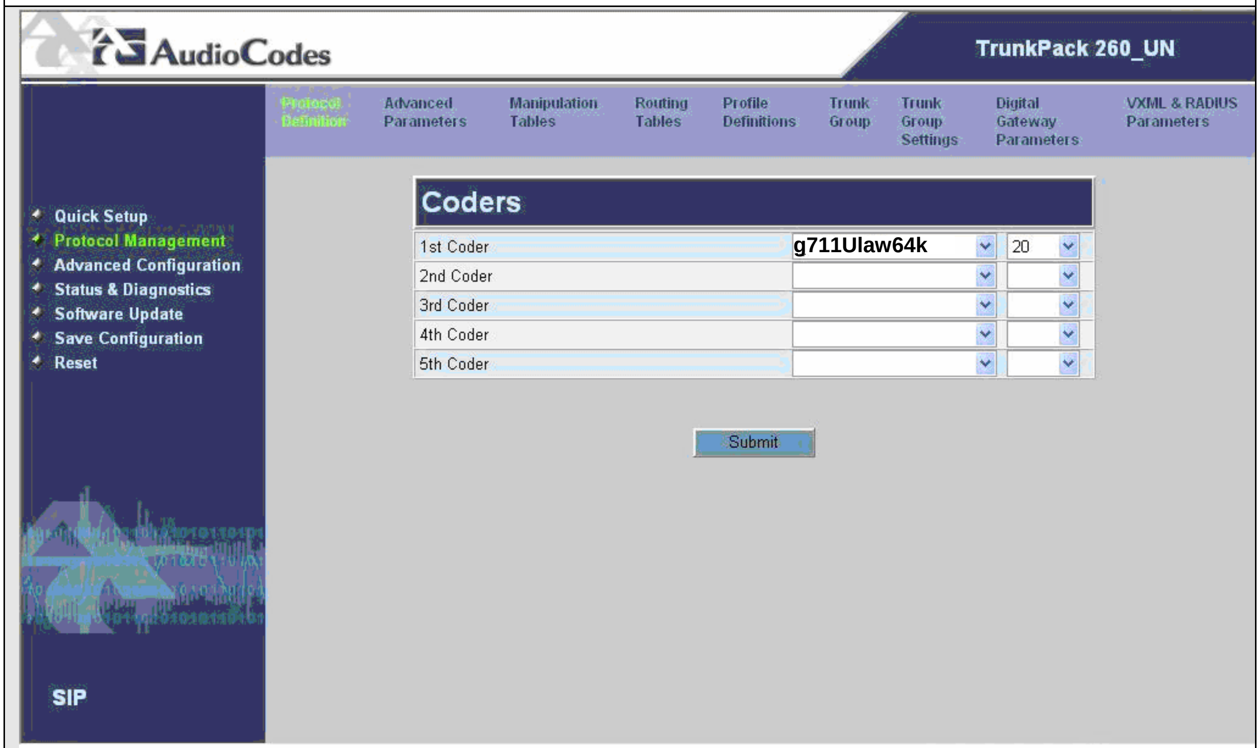
- Avaya Meeting Exchange via SIP connectivity (see **Figure 1**).
- Avaya Communication Manager via ISDN-PRI connectivity (see **Figure 1**).

Step	Description
5.1	Administer settings for AudioCodes Media Gateway as follows: • Open a web browser and enter the following URL: **http://<default IP Address>** • Log in to the AudioCodes Media Gateway with the appropriate credentials. **Note**: To obtain default IP Address, login and password information, refer to Section 8 reference [3].

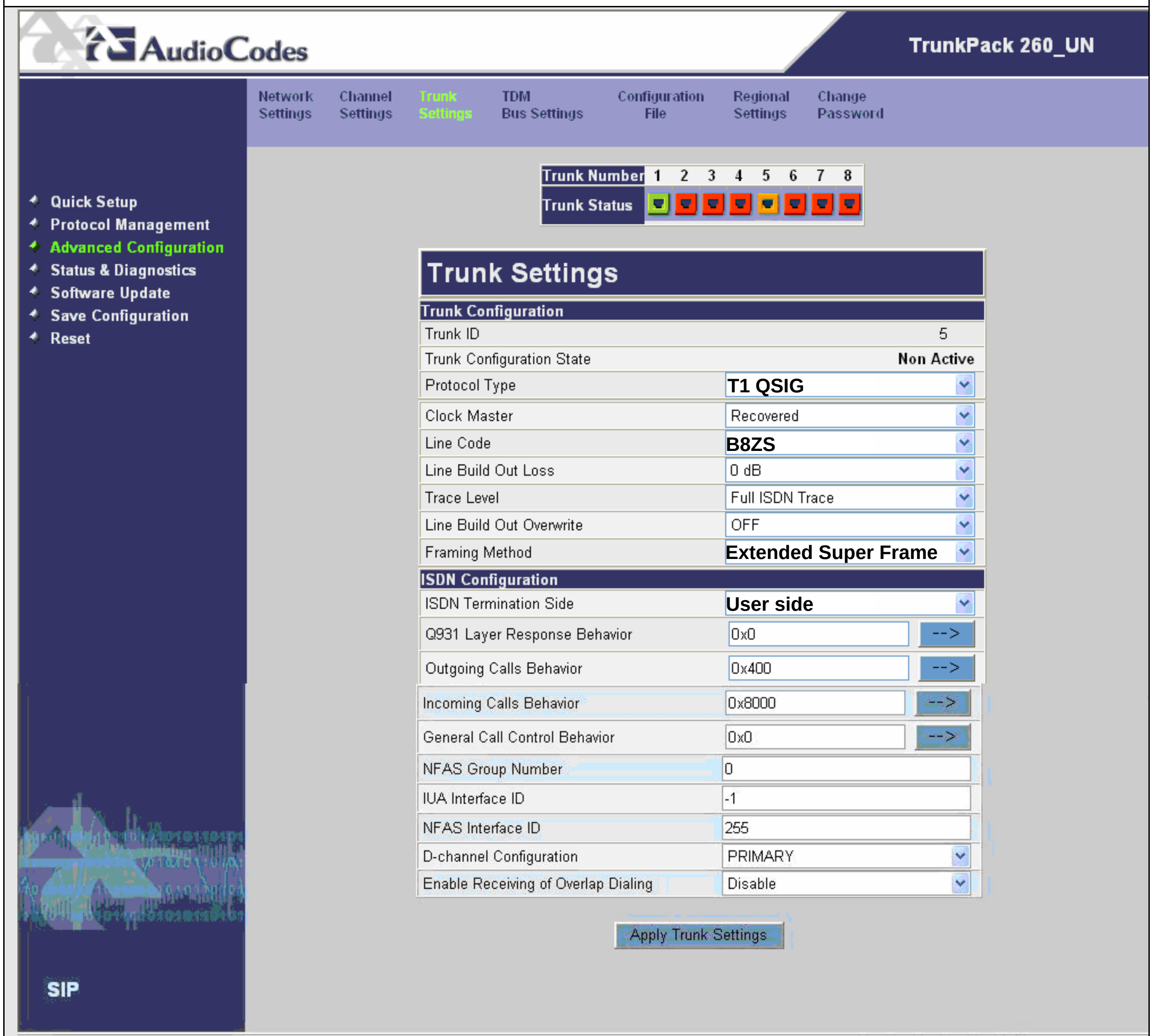
Step	Description
5.2	Configure **Network Settings** as follows: • Click **Advanced Configuration**. • Click **Network Settings**. • Administer settings as per below. The **IP Settings** configured for the AudioCodes Media Gateway must enable layer 3 connectivity with Avaya Meeting Exchange. • When finished, click the **Submit** button on the bottom of the screen.

Step	Description
5.3	Configure **Proxy & Registration** parameters as follows: • Click **Protocol Management**. • Click **Protocol Definition ➔ Proxy & Registration**. • Administer settings as per below. • When finished, click the **Submit** button on the bottom of the screen.

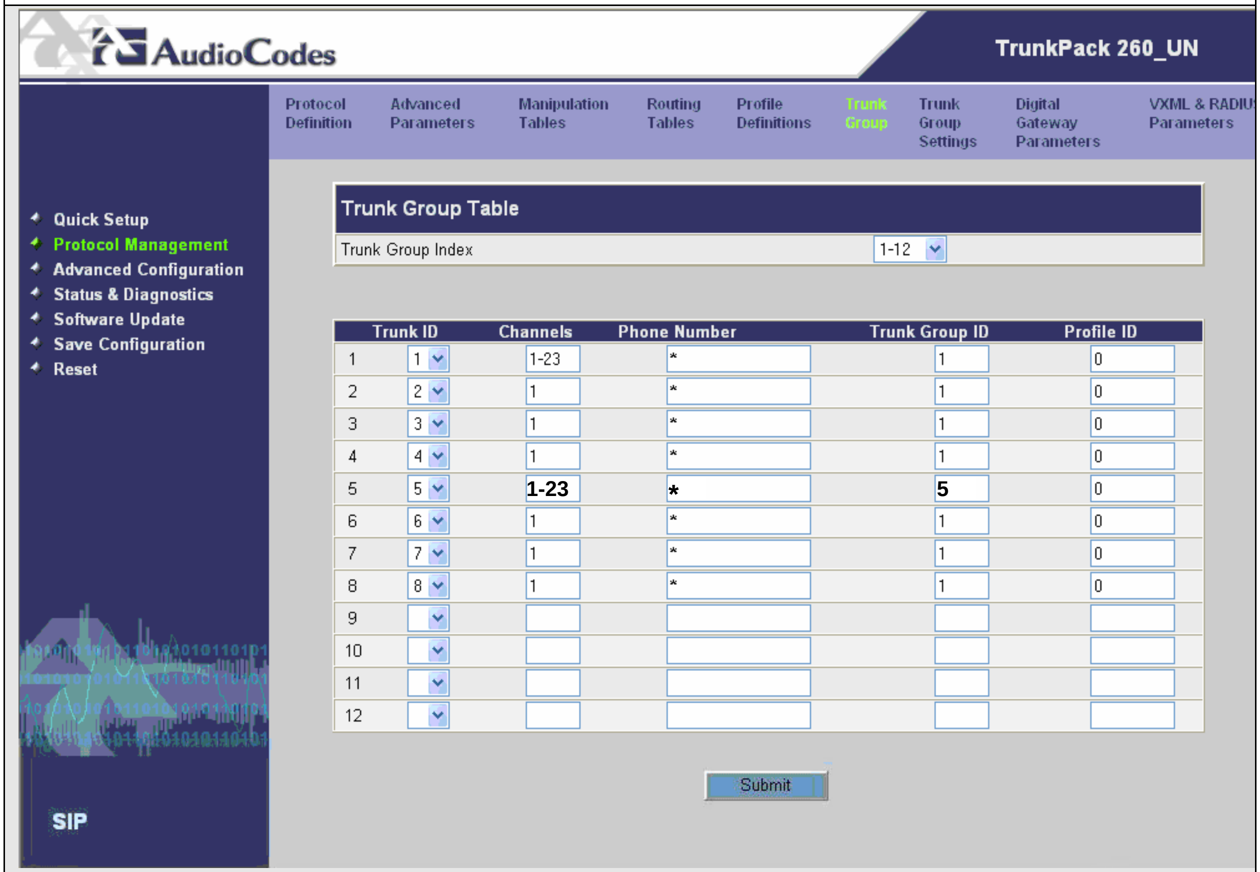


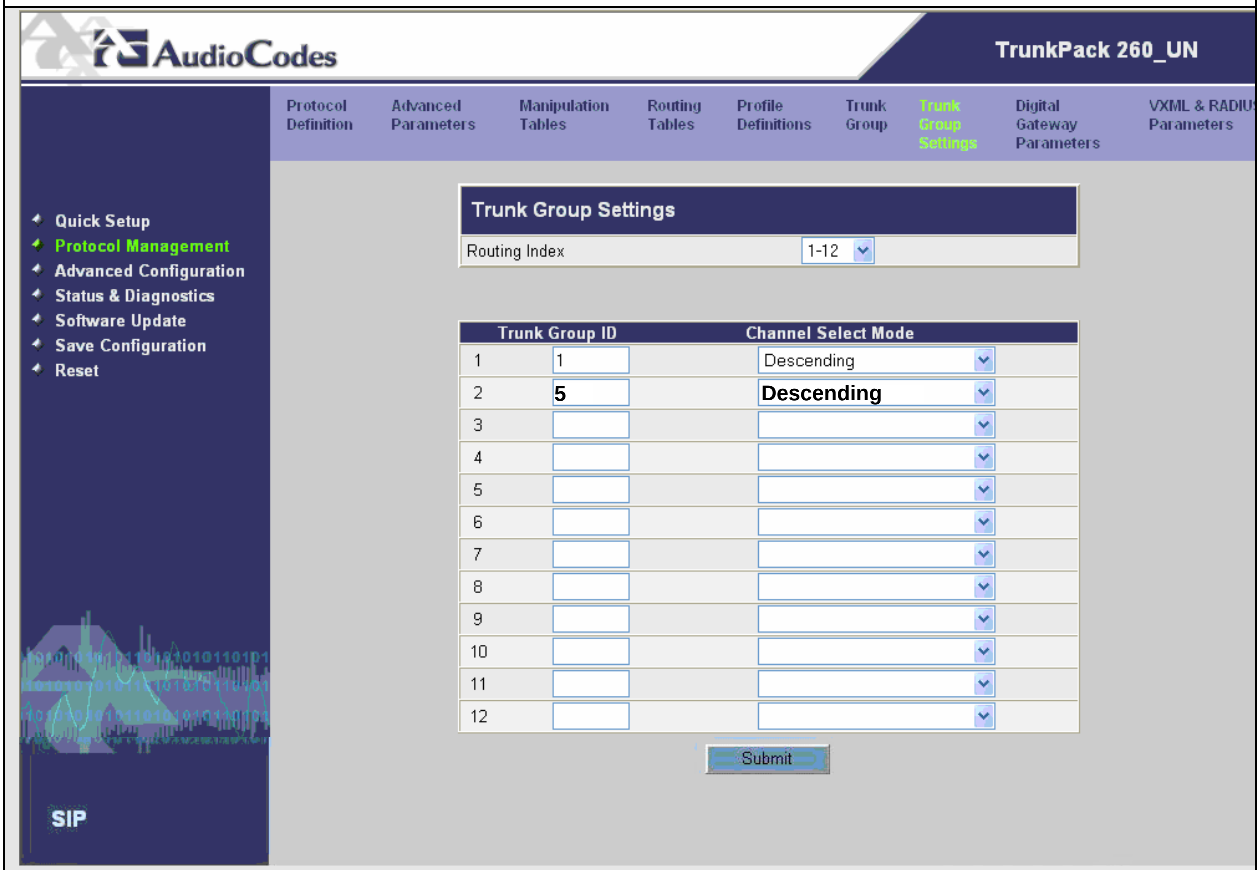
Step	Description
5.4	Configure **Coders** as follows: • Click **Protocol Management**. • Click **Protocol Definition →Coders**. • Administer settings as per below. Configure a **Coder** that is supported on Avaya Meeting Exchange; either **g711Ulaw64k**, or **g711Alaw64k**. • When finished, click the **Submit** button on the bottom of the screen.
	

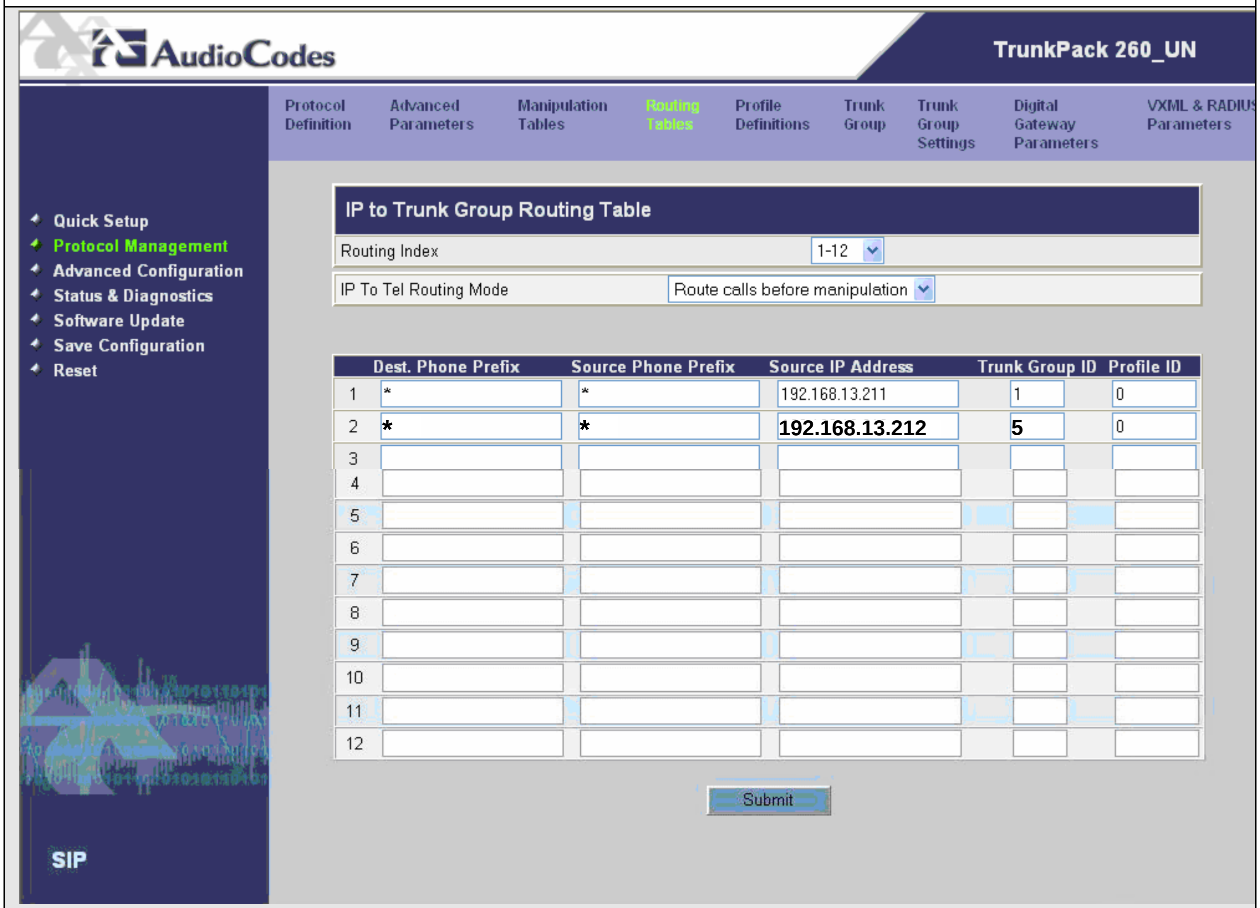
Step	Description
5.5	Configure **Trunk Settings** to interoperate with the **DS1 CIRCUIT PACK** configuration on Avaya Communication Manager (see **Section 3**, **Step 3.4**) as follows: • Click **Advanced Configuration**. • Click **Trunk Settings**. • Select **Trunk Number 5**. • Administer settings as per below. • When finished, click the **Apply Trunk Settings** button on the bottom of the screen.



Step	Description
5.6	Configure **Trunk Group Table** as follows: • Click **Protocol Management**. • Click **Trunk Group**. • Administer settings as per below. **1-23** channels for **Trunk ID 5** are provisioned due to Channel 24 being utilized as a signaling channel for ISDN-PRI QSIG. • When finished, click the **Submit** button on the bottom of the screen.



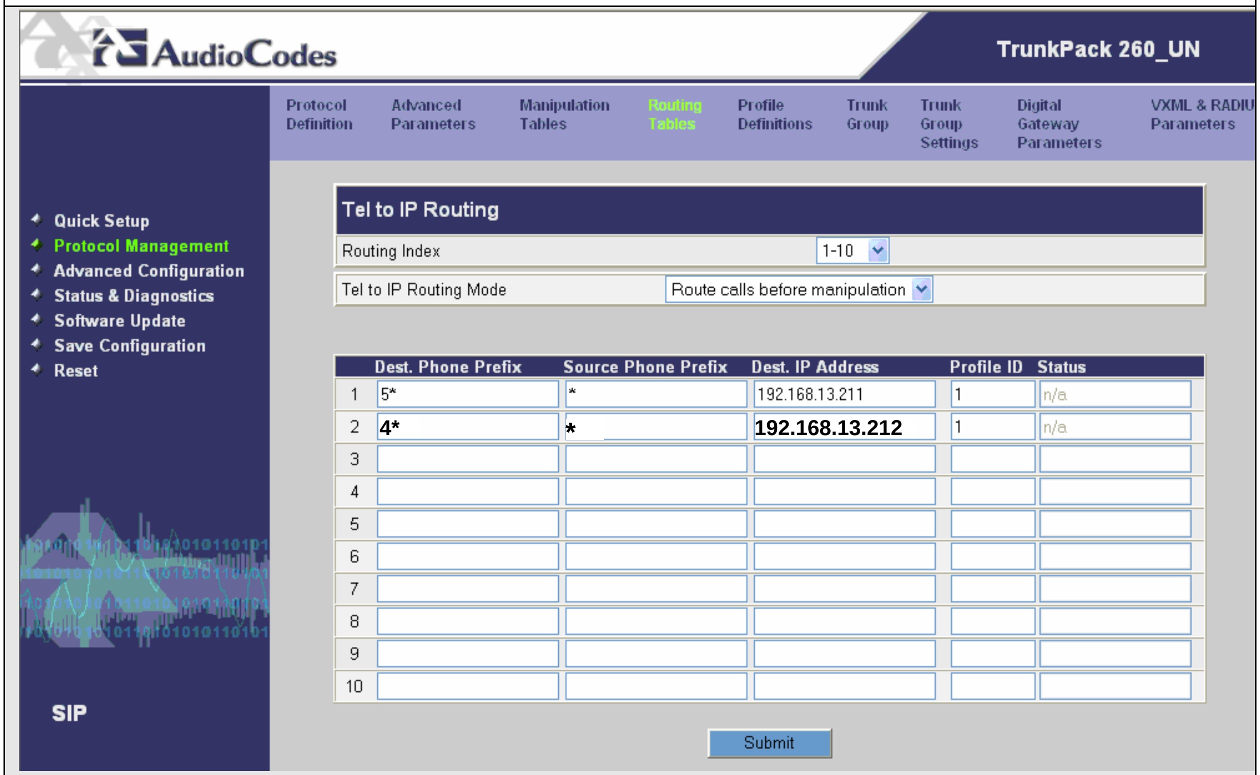
Step	Description
5.7	Configure **Trunk Group Settings** as follows: • Click **Protocol Management**. • Click **Trunk Group Settings**. • Administer settings as per below. The **Channel Select Mode** is provisioned to hunt in the opposite direction as Avaya Communication Manager (see **Section 3**, **Step 3.7**). • When finished, click the **Submit** button on the bottom of the screen. 

Step	Description
5.8	To enable Dial-Out from Avaya Meeting Exchange, configure the **IP to Trunk Group Routing Table** as follows: • Click **Protocol Management**. • Click **Routing Tables, IP to Trunk Group Routing**. • Administer settings as per below. All calls originating from **Source IP Address** = **192.168.13.212** (Avaya Meeting Exchange) are routed to **Trunk ID** = **5** (the trunk that connects to Avaya Communication Manager). • When finished, click the **Submit** button on the bottom of the screen. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.9</td><td>To allow Dial-In to Avaya Meeting Exchange, configure Tel to IP Routing as follows:

Click Protocol Management.
Click Routing Tables, Tel to IP Routing.
Administer settings as per below. All calls with Dest. Phone Prefix = 4* are ‘routed’ to Dest. IP Address = 192.168.13.212 (Avaya Meeting Exchange).
When finished, click the Submit button on the bottom of the screen.

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



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

6. Verification Steps

The following steps can be used to verify the configuration described in these Application Notes.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.1</td><td>Verify all members for the ISDN trunk group provisioned in Section 3, Step 3.8 are in-service/idle.

From a SAT session:
Issue the command “status trunk 13”.
Verify that all members in Trunk Group 13 are in-service/idle.</td></tr>
<tr><td>6.2</td><td>Verify the ISDN trunk group is utilized when a call from a SIP station Dials-In to Avaya Meeting Exchange.

From a SAT session:
Issue the command “list trace tac 113”, where 113 is the TAC defined for the trunk group provisioned in Section3, Step 3.6.
From a SIP station, dial 444 to enter a conference as moderator via a DirectCallFlow scenario.
Enter *9 to initiate a Blast Dial.
Note: This trace shows the SIP Station Dialing-In via a DirectCallFlow. Dial-Out, (e.g., Blast Dial) is not shown. Also, a SIP station was arbitrarily selected for these Verification Steps; as any station type (e.g., SIP, H.323, Digital or Analog) is capable of Dialing-In to Avaya Meeting Exchange from Avaya Communication Manager via the AudioCodes Media Gateway.

<pre>list trace tac 113   Page   1

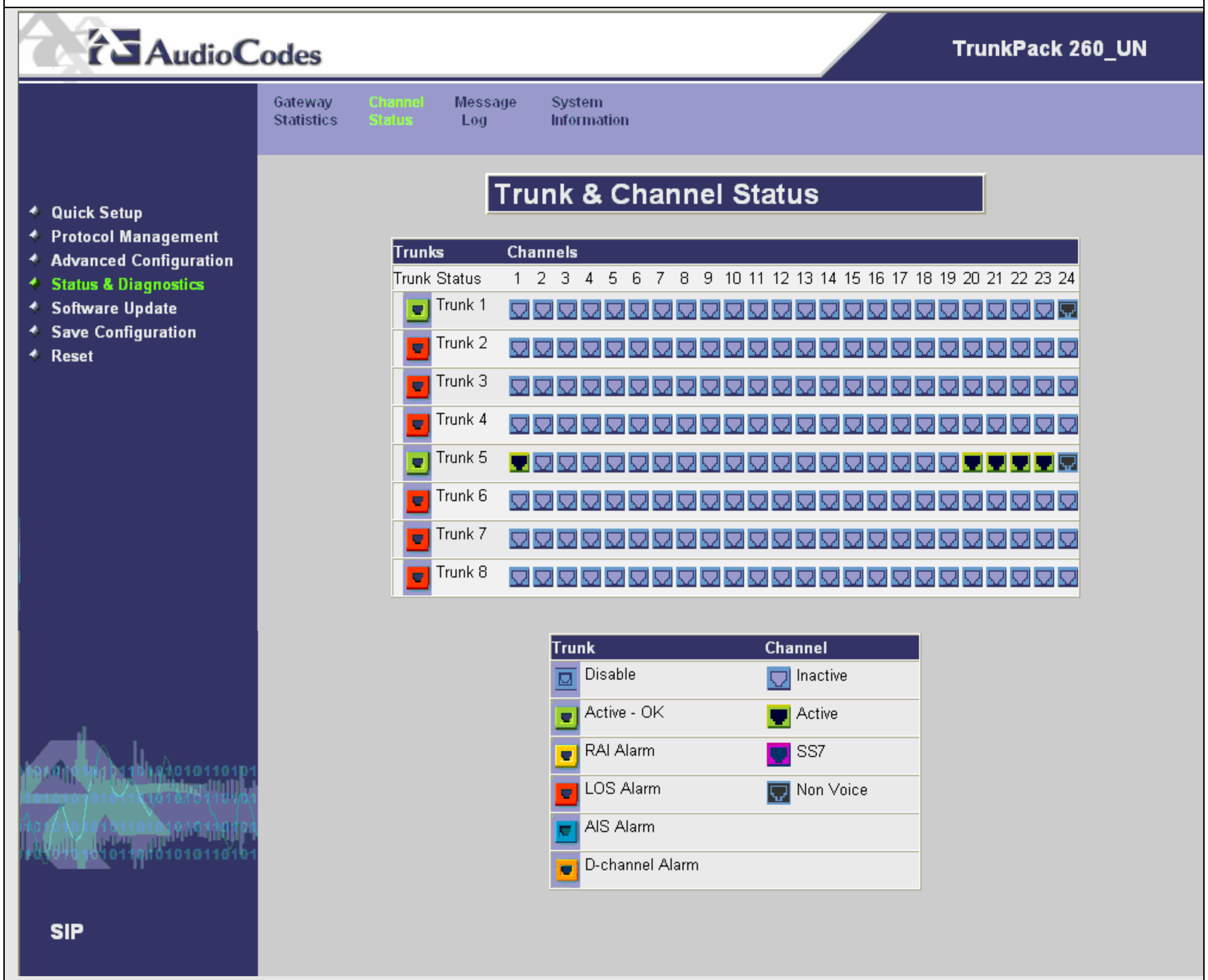
                                LIST TRACE

time   data

16:18:10   dial 444 route:AAR
16:18:10   term trunk-group 13   cid 0x27a
16:18:10   dial 444 route:AAR
16:18:10   route-pattern 13 preference 1 cid 0x27a
16:18:10   seize trunk-group 13 member 1 cid 0x27a
16:18:10   Calling Number & Name NO-CPNumber NO-CPName
16:18:10   Proceed trunk-group 13 member 1 cid 0x27a
16:18:10   active trunk-group 13 member 1 cid 0x27a</pre></td></tr>
</table>

Step	Description

6.3	Verify ISDN **Trunk & Channel Status** on AudioCodes as follows:
	• Click **Status & Diagnostics**.
	• Click **Channel Status**.
	• This screen capture depicts the scenario initiated in **Step 6.2**. Also, note the following:
	    o Status for **Trunk 5** is **Active - OK**.
	    o The Hunt pattern for 5 **Active** channels on **Trunk 1**. **Channel 1** is the SIP Station Dialing-In via a DirectCallFlow. **Channels 20-23** are the conferees that were added to the conference via Blast Dial.

Step	Description
6.4	Verify that calls to and from Avaya Meeting Exchange are managed correctly, e.g., callers are added/removed from conferences. This is verified by the following procedures: • Log in to the Avaya Meeting Exchange Server console to access the command line interface with the appropriate credentials. • At the command prompt, enter the command: **watch -t -n 5 -d "ipinfo -l |egrep -ci active".** This command will provide a real time, continuous update of port utilization on Avaya Meeting Exchange.

7. Conclusion

These Application Notes provide administrators with the procedures to configure Avaya Communication Manager and the Avaya Meeting Exchange S6100 Conferencing Server, utilizing ISDN-PRI connectivity via AudioCodes Media Gateway. With appropriate configuration, Dial-In and Dial-Out conferencing is successfully established between Avaya Meeting Exchange and Avaya Communication Manager.

8. Additional References

Avaya references, available at http://support.avaya.com

1. *Administrator Guide for Avaya Communication Manager*, Doc ID: 03-300509
2. *Administration for Network Connectivity for Avaya Communication Manager*, Doc ID: 555-233-504

AudioCodes reference

3. *TP-260 UN SIP User's Manual Version 4.4*, Document #: LTRT-68002

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 Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com