Avaya Solution & Interoperability Test Lab

Front-Ending Avaya Partner Advanced Communication System with an AudioCodes MP-118 SIP Media Gateway to Interoperate with Avaya Aura™ Session Manger – Issue 1.0

Abstract

These Application Notes present a sample configuration that uses an AudioCodes MP-118 SIP Media Gateway to connect the Avaya Partner Advanced Communication System (ACS) 8.0 with Avaya Aura™ Session Manager 5.2, which in turn provides call routing support to other Avaya SIP products such as Avaya Aura™ Communication Manager and Avaya Modular Messaging.

This solution addresses the need for the Avaya Aura™ telephony infrastructure located at a central site to interoperate with telephone key systems like the Avaya Partner Advanced Communication System in branch locations. The sample configuration involves 2 sites connected through a WAN. The Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager, and Avaya Modular Messaging reside at the Headquarters; the AudioCodes MP-118 and the Avaya Partner Advanced Communication System reside at the Branch site. SIP trunks link the AudioCodes MP-118 front-ending the Partner Advanced Communication System at the Branch site to the Avaya Aura™ Session Manager at the Headquarters site enabling calls from the PSTN to reach branch Partner phones as well as inter-branch calling and calls between the branch and the Headquarters.

The validation test of the sample configuration was conducted at the Avaya Solution and Interoperability Test Lab at the request of the Avaya Solutions and Marketing Team.

1. Introduction

These Application Notes present a sample configuration that uses an AudioCodes MP-118 SIP Media Gateway to connect the Avaya Partner Advanced Communication System (ACS) 8.0 with Avaya Aura™ Session Manager 5.2, which in turn provides call routing support to other Avaya SIP products such as Avaya Aura™ Communication Manager 5.2.1, and Avaya Modular Messaging 5.2.

This solution addresses the need for the Avaya Aura™ telephony infrastructure located at a central site to interoperate with telephone key systems like the Avaya Partner Advanced Communication System in branch locations. The sample configuration involves 2 sites connected through a WAN. The Session Manager, Communication Manager, and Modular Messaging reside at the Headquarters; the AudioCodes MP-118 and the Avaya Partner Advanced Communication System reside at the Branch site. SIP trunks link the AudioCodes MP-118 front-ending the Avaya Partner Advanced Communication System at the Branch site to the Session Manager at the Headquarters site enabling calls from PSTN to reach branch Partner phones as well as inter-branch calling and calls between the branch and the Headquarters.

For the sample configuration, the Avaya Aura™ Session Manager 5.2 runs on an Avaya S8510 Server, the Avaya Aura™ Communication Manager 5.2.1 runs on an Avaya S8720 Server with Avaya G650 Media Gateway. The results in these Application Notes should be applicable to other Avaya servers and media gateways that support the Avaya Aura™ architecture.

The sample configuration utilizes the AudioCodes SIP Media Gateway model MP-118 at the branch location. The configuration steps presented in these Application Notes should also be applicable to the AudioCodes SIP Media Gateway model MP-114 (which is similar to MP-118 but with less port capacities) using the AudioCodes firmware version as specified in **Section 3**.

1.1. Interoperability Testing

The interoperability testing focused on voice and fax calls between PSTN and branch Partner phones as well as calling between the Headquarters and the branch.

1.1.1. Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager

Avaya Aura™ Session Manger is a routing hub for SIP calls among connected SIP telephony system components. The Avaya Aura™ System Manager provides management functions for the Avaya Aura™ Session Manager. In the sample configuration, SIP trunks link the Avaya Aura™ Session Manager to the Avaya Aura™ Communication Manager and Avaya Modular Messaging at the Headquarters and the AudioCodes MP-118 SIP Media Gateway at the Branch site. Note that the sample configuration uses the Avaya Aura™ Communication Manager as an Access Element which supports natively configured H.323, Digital and analog (fax) endpoints. In order to add SIP phones at the Headquarters that interwork with the other types of endpoints, a separate Avaya Aura™ Communication Manager configured as a Feature Server needs to be added.

1.1.2. AudioCodes SIP Media Gateway

An AudioCodes SIP Media Gateway, referred to as AudioCodes MP-118 throughout the remainder of this document, is located at each branch site. The AudioCodes MP-118 front-ends the Avaya Partner Advanced Communication System key system located in the branch allowing the branch Partner phones to interwork with the Headquarters phones as well as to access the PSTN through the T1/E1 facilities at the central site via the Avaya G650 Media Gateway. For the sample configuration, the AudioCodes MP-118 connects to the Avaya Aura™ Session Manager by SIP trunks through the WAN. Of the 4 FXS ports on the AudioCodes MP-118, 3 are connected to the line ports on the Avaya Partner Advanced Communication System key system and the 4th FXS port is directly connected to a fax machine in the branch. The FXO ports on the AudioCodes MP-118 are not used.

1.1.3. Avaya Partner Advanced Communication System

The Avaya Partner Advanced Communication System is a key telephone system located at each branch site front-ended by an AudioCodes MP-118. In the sample configuration, 3 line ports of this Avaya key telephone system are connected to the FXS ports of the AudioCodes MP-118. A separate line port on the Avaya Partner Advanced Communication System is directly connected to an analog line from the Service Provider for 911 calls to the local Emergency Response Center as well as DID (Direct Inward Dialing) calls from the PSTN when the branch WAN connection to the Headquarters is out of service. The validation testing of the sample configuration focused on the interoperability of the Avaya Partner Advanced Communication System with the Avaya telephony infrastructure at the Headquarters through the branch AudioCodes MP-118.

1.1.4. PSTN Access

The sample configuration provides two PSTN access methods for the branch:

1. via the central Avaya telephony infrastructure
2. via the Service Provider analog line directly connected to the branch Avaya Partner Advanced Communication System

The first access listed above is for normal calls between PSTN and the branch including faxing. The second access is for local 911 calls and DID calls to the branch from the PSTN if access to the branch through the centralized Avaya telephony infrastructure is blocked due to network problems.

It should be noted that the inter-site calling and calls between Headquarters and the branch are all on-net calls in the sample configuration. The direct PSTN access method provides fail-over access for the branch to Headquarters and other branches when connectivity to the central site is lost.

1.1.5. Dialing Numbers

To help with understanding the routing configurations contained in the ensuing text, the dialing numbers are listed below. Note that these dialing numbers are made-up for validating the sample configuration. In certain situations, a real number is mapped to the made-up number where appropriate (e.g., a real Direct Inward Dialing number from the PSTN is mapped in Avaya Aura™ Communication Manager to the made-up Branch access number for testing). Also note that calls between Headquarters and the branch as well as inter-branch calls are on-net calls in the normal situation where network connectivity from the Headquarters to the branch sites is intact.

From	To	Call Type	Dialed Number
PSTN	Branch	Voice	1+10 digit branch access number *
PSTN	Branch	Fax	1+10 digit branch fax number **
Headquarters	Branch	Voice	8+1 908 543 4000
Headquarters	Branch	Fax	8+1 908 543 4009
Branch	PSTN	Voice	1+10 digit PSTN number
Branch	PSTN	Fax	1+10 digit PSTN fax number
Branch	Headquarters	Voice	1+908 123 xxxx
Branch	Headquarters	Fax	1+908 123 xxxx
Branch A	Branch B	Voice	1+10 digit Branch access number
Branch A	Branch B	Fax	1+10 digit Branch fax number

* The DID number is mapped at the Communication Manger to the branch access number 19085434000

** The DID number is mapped at the Communication Manger to the branch fax number 19085434009

At the branch, users of the Avaya Partner Advanced Communication System phones press an available Line button on the phone, hears a line dial tone, then dials the 11-digit number as listed above to call the PSTN, Headquarters, or another branch. To make an intra-branch station-to-station call, users of the Avaya Partner Advanced Communication System phones press an available Intercom button on the phone, hears an intercom dial tone, then dials a 2-digit extension number to make the call. The incoming call from the PSTN, Headquarters, or another branch will ring and flash one of the available Line buttons on the Avaya Partner Advanced Communication System phones. The user presses the ringing/flashing Line button to answer the call. The incoming call from another phone in the same branch will ring and flash one of the available Intercom button.

1.2. Support

For technical support on the AudioCodes MP-118 SIP Media Gateway, contact AudioCodes via the support link at http://www.audiocodes.com/support. In case of an existing support agreement please use iSupport system at https://crm.audiocodes.com/OA_HTML/jtflogin.jsp.

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support. Customers may also use specific numbers provided on http://support.avaya.com to directly access specific support and consultation services based upon their Avaya support agreements.

2. Reference Configuration

The network implemented for the sample configuration is shown in **Figure 1**. The network is modeled after an Enterprise consisting of a main Headquarters/Datacenter location and multiple branch locations all inter-connected over a corporate WAN. While configuration for one branch location is presented in these Application Notes, the same configuration procedures can be followed to configure additional branches with the appropriate site-specific changes (IP for the local branch AudioCodes MP-118, the local branch access phone and fax numbers, etc.).

In the sample configuration the Session Manager serves as the routing hub linking the Communications Manager at the Headquarters to the branch-located AudioCodes MP-118 through SIP trunks. All calls between the two sites are carried over these SIP trunks. At the branch site, 3 of the Partner line ports are connected directly to 3 FXS interface ports on the AudioCodes MP-118 for connectivity to the Headquarters through the AudioCodes SIP Media Gateway. The Partner Advanced Communication System is also directly connected to the PSTN through one of its available line ports to a service provider CO line. This connection provides for direct PSTN access for 911 calls from the branch, as well as fail-over connectivity to the PSTN, Headquarters, and other branches if the corporate WAN is temporarily out of service.

Fax machines are set up at both the branch (connected to a MP-118 FXS port) and the Headquarters (connected to a port on the analog circuit pack in the G650 Media Gateway) to enable faxing between the two locations as well as faxing between PSTN and the branch fax machine.

The Headquarters phones use Modular Messaging for voice mail access and coverage. If the calls to a branch are unanswered after a configured time period, the unanswered calls are redirected to the Modular Messaging system at the Headquarters. If all lines to the branch Partner key system are busy, incoming calls are similarly redirected to the Modular Messaging system at the Headquarters. In real deployment, the Modular Messaging can be replaced by a Call Center or an automatic voice response system like Avaya Voice Portal. The branch phones use the Partner Voice Messaging capabilities for voice mail access and coverage.

Note that the Communication Manger serves as an Access Element in the sample configuration. As such, it supports non-SIP phones (H.323 and other Avaya digital and/or analog telephone sets) natively configured on the same Communication Manager. In order to add SIP phones at the Headquarters and enable inter-working between the SIP phones and non-SIP phones, a Feature Server must be added to the sample network[1]. This restriction will be lifted in future releases of Session Manager and Communication Manager. In the sample configuration, no SIP phones are used at the Headquarters location.

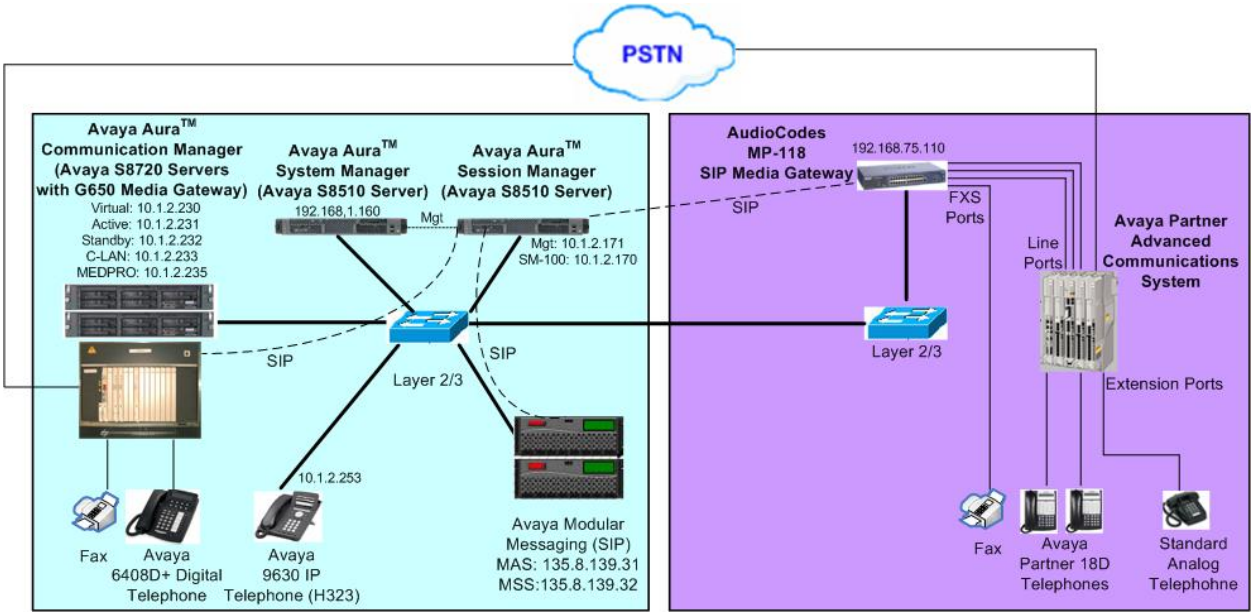


Figure 1 – Network Diagram

[1] See **[4]** for configuring Communication Manager as a Feature Server to support SIP endpoints.

3. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8510 Server	Avaya Aura™ Session Manager 5.2, Service Pack 1, Load 5.2.1.0.52010 (GA)
	Avaya Aura™ System Manager 5.2, Service Pack 1, Load 5.2.1.0.521001 (GA)
Avaya S8720 Server	Avaya Aura™ Communication Manager 5.2.1 (R015x.02.1.016.4, Patch 17959)
Avaya G650 Media Gateway with following circuit packs: – Control-LAN – IP Media Processor – Digital Line – Analog Line – DS1 Interface	 TN799DP – HW01 FW032 TN2602AP – HW02 FW047 TN2224B – 000012 TN793CP – HW05 FW006 TN464F – 000004
Avaya Modular Messaging – Application Server (MAS) on Avaya S8800 1U Server – Storage Server (MSS) on Avaya S8800 1U Server	 5.2, Build 9.2.150.13 (Patch 520008) 5.2, Build 5.2-11.0
Avaya 9620 IP Telephones (H.323)	3.1
Avaya 6480+ Digital Telephone	-
Avaya 6210 Analog Telephone	-
Avaya Partner ACS	8.0
Avaya Partner 18D Telephone	-
AudioCodes MP-118	5.80A.035.004

Table 1 – Software/Hardware Version Information

4. Configure Avaya Aura™ Communication Manager

This section shows the necessary steps to configure Communication Manager to support the sample configuration. It is assumed that the basic configuration on Communication Manager, the required licensing, the configuration for connection to the PSTN through the T1/E1 interface, the configuration required for accessing Communication Manager Messaging, as well as station configurations for natively connected Avaya H.323, Digital and analog (fax) phones have already been administered. See listed documents in **Section 11** for additional information.

All commands discussed in this section are executed on Communication Manager using the System Access Terminal (SAT). The administration procedures in this section include the following areas. Some administration screens have been abbreviated for clarity.

- Verify Avaya Aura™ Communication Manager license
- Configure System Parameters Features
- Configure IP Node Names
- Configure IP Interface
- Configure IP Codec Set
- Configure IP Network Region
- Configure SIP Signaling Group and Trunk Group
- Configure Route Pattern
- Configure Location and Public Unknown Numbering
- Configure Automatic Alternate Routing (AAR)
- Configure Automatic Route Selection (ARS)
- Administer Incoming Call Handling Treatment
- Configure Voice Messaging Hunt Group
- Configure Voice Messaging Coverage Path

4.1. Verify Avaya Aura™ Communication Manger License

Log in to the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column. The difference between the two values needs to be greater than or equal to the desired number of simultaneous SIP trunk connections.

The license file installed on the system controls the maximum capacities permitted. If there is insufficient capacity or a required feature is not enabled, contact an authorized Avaya sales representative to make the appropriate changes.

```
display system-parameters customer-options                      Page   2 of  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                           USED
                     Maximum Administered H.323 Trunks: 800   200
           Maximum Concurrently Registered IP Stations: 18000 4
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 800   57
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 10    1
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   2
   Maximum Number of Expanded Meet-me Conference Ports: 0     0

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

4.2. Configure System Parameters Features

Use the "change system-parameters features" command to allow for trunk-to-trunk transfers. This feature is needed to be able to transfer an incoming/outgoing call from/to the remote switch back out to the same or another switch. For simplicity, the **Trunk-to-Trunk Transfer** field was set to "all" to enable all trunk-to-trunk transfers on a system-wide basis.

Note that this feature poses significant security risk, and must be used with caution. As an alternative, the trunk-to-trunk feature can be implemented using Class Of Restriction or Class Of Service levels. Refer to the appropriate documentation in **Section 11** for more details.

```
change system-parameters features                                Page   1 of  18
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? y
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                  AAR/ARS Dial Tone Required? y
                             Music/Tone on Hold: music Type: ext   65021
             Music (or Silence) on Transferred Trunk Calls? no
                       DID/Tie/ISDN/SIP Intercept Treatment: attd
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n
```

4.3. Configure IP Node Names

Use the "change node-names ip" command to add entries for the C-LAN that will be used for connectivity, its default gateway, and Session Manager. In this case, "clan1" and "10.1.2.233" are entered as **Name** and **IP Address** for the C-LAN, "asm" and "10.1.2.170" are entered for the Session Manager Security Module (SM-100) interface, and "Gateway001" and "10.1.2.1" are entered for the default gateway. Note that "Gateway001" will be used to configure the IP interface for the C-LAN (see **Section 4.4**) and the configured node-name "asm" and "clan1" will be used later on in the SIP Signaling Group administration (**Section 4.7.1**). The actual node names and IP addresses may vary. Submit these changes.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
DenverASM           10.80.100.23
DenverCS1000        10.80.50.50
Gateway001          10.1.2.1
HDTG1               10.1.2.63
HDTG2               10.1.2.64
Home1               10.3.3.50
Home2               10.3.3.41
SESBr1              10.1.2.12
SurvCM              10.32.2.80
asm                 10.1.2.170
callrtg1            10.1.2.217
callrtg2            10.1.2.218
calltraff1          10.1.2.193
calltraff2          10.1.2.194
clan1               10.1.2.233
clan2               10.1.2.234
( 16 of 29   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

4.4. Configure IP Interface

Add the C-LAN to the system configuration using the “add ip-interface 1a02” command. The actual slot number may vary. In this case, “1a02” is used as the slot number in the G650 Media Gateway. Enter the C-LAN node name assigned from **Section 4.3** into the **Node Name** field. Enter proper values for the **Subnet Mask** and **Gateway Node Name** fields. In this case, “/24” and “Gateway001” are used to correspond to the network configuration in these Application Notes. Set the **Enable Interface** and **Allow H.323 Endpoints** fields to “y”. Default values may be used in the remaining fields. Submit these changes.

```
add ip-interface 1a02                                          Page   1 of   3
                                   IP INTERFACES


                  Type: C-LAN
                  Slot: 01A02        Target socket load and Warning level: 400
           Code/Suffix: TN799  D           Receive Buffer TCP Window Size: 8320
      Enable Interface? y                           Allow H.323 Endpoints? y
                  VLAN: n                            Allow H.248 Gateways? y
        Network Region: 1                             Gatekeeper Priority: 5



                                  IPV4 PARAMETERS
             Node Name: clan1
           Subnet Mask: /24
     Gateway Node Name: Gateway001

         Ethernet Link: 1
         Network uses 1's for Broadcast Addresses? y
```

4.5. Configure IP Codec Set

Configure the IP codec set to use for IP calls. Use the “change ip-codec-set n” command, where “n” is the codec set number to be used for interoperability. Enter the desired audio codec type in the **Audio Codec** field. Retain the default values for the remaining fields. The “G.711MU” codec was used in the test configuration.

```
change ip-codec-set 1                                          Page   1 of   2

                           IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2:
 3:
 4:
 5:
 6:
 7:
```

On **Page 2** of the IP Codes Set form, change the **Mode** field for **FAX** to "t.38-standard" for allowing faxing to and from branch locations. Retain the default values for the remaining fields, and submit these changes.

```
change ip-codec-set 1                                          Page   2 of   2

                           IP Codec Set

                               Allow Direct-IP Multimedia? n



                     Mode                Redundancy
    FAX              t.38-standard        0
    Modem            off                  0
    TDD/TTY          pass-through         0
    Clear-channel    n                    0
```

4.6. Configure IP Network Region

In the sample configuration, network region "1" was used for calls to the AudioCodes MP-118 via Session Manager. Use the "change ip-network-region 1" command to configure this network region. For the **Authoritative Domain** field, enter the SIP domain name configured for this enterprise network. This value is used to populate the SIP domain in the From header of SIP INVITE messages for outbound calls. It also must match the SIP domain in the request URI of incoming INVITEs from other systems. Enter a descriptive **Name**. For the **Codec Set** field, enter the corresponding audio codec set configured above in **Section 4.5**. Enable the **Intra-region IP-IP Direct Audio**, and **Inter-region IP-IP Direct Audio**. These settings will enable direct media between Avaya IP telephones and the MP-118. Retain the default values for the remaining fields, and submit these changes.

```
display ip-network-region 1                                     Page   1 of  19
                              IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: avaya.com
    Name: ASM to MP-118
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 10001
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                     RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

4.7. Configure SIP Signaling Group and Trunk Group

In the sample configuration, trunk group 32 and signaling group 32 were used to reach Session Manager. Use the "add signaling-group n" command, where "n" is an available signaling group number. Enter the following values for the specified fields in the next sub-sections and retain the default values for all remaining fields. Submit these changes.

4.7.1. SIP Signaling Group

Use the command "add signaling-group n" to add the SIP signaling group, where "n" is an available signaling group number. Enter the following values for the specified fields, and retain the default values for all remaining fields.

- **Group Type**: "sip"
- **Transport Method**: "tcp"
- **Near-end Node Name**: "clan1" node name from **Section 4.3**
- **Far-end Node Name**: "asm" Session Manager node name from **Section 4.3**
- **Far-end Network Region**: Network region number "1" from **Section 4.6**
- **Far-end Domain**: SIP domain name from **Section 4.6**
- **DTMF over IP**: "rtp-payload"

```
add signaling-group 32
                                SIGNALING GROUP

 Group Number: 32             Group Type: sip
                        Transport Method: tcp
  IMS Enabled? n





   Near-end Node Name: clan1                 Far-end Node Name: asm
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                       Far-end Network Region: 1
Far-end Domain: avaya.com

                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                      IP Audio Hairpinning? n
         Enable Layer 3 Test? n                     Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

For the validation test of the sample configuration, the **Enable Layer 3 Test** field was set to "n". With this setting, the Communication Manager will attempt to ping the far-end node to verify the SIP connection. If the **Enable Layer 3 Test** field is set to "y", the Communication Manger will use the SIP OPTIONS message to verify that the SIP connection is available.

4.7.2. SIP Trunk Group

Use the "add trunk-group n" command, where "n" is an available trunk group number, to add SIP trunk groups. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type**: "sip"
- **Group Name**: Descriptive text
- **TAC**: An available trunk access code as per the dialplan
- **Service Type**: "tie"
- **Signaling Group**: The signaling group number as configured in **Section 4.7.1**
- **Number of Members**: Equal to maximum number of concurrent calls supported

```
add trunk-group 32                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 32                  Group Type: sip           CDR Reports: y
  Group Name: To ASM                     COR: 1        TN: 1        TAC: 132
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                     Signaling Group: 32
                                                   Number of Members: 20
```

Navigate to **Page 3**, and enter "public" for the **Numbering Format** field as shown below. Use default values for all other fields.

```
add trunk-group 32                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                       Maintenance Tests? y



                     Numbering Format: public
                                                  UUI Treatment: service-provider

                                                   Replace Restricted Numbers? n
                                                  Replace Unavailable Numbers? n
```

Navigate to **Page 4**, and enable **Mark Users as Phone**. Use default values for all other fields.

```
add trunk-group 32                                          Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? y
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                  Network Call Redirection? n
                     Send Diversion Header? n
                   Support Request History? y
              Telephone Event Payload Type:
```

4.8. Configure Route Pattern

Configure a route pattern to correspond to the newly added SIP trunk groups. Use the "change route-pattern n" command, where "n" is an available route pattern. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern Name**: A descriptive name
- **Grp No**: The trunk group number configured in **Section 4.7.2**
- **FRL**: Facility Restriction Level that allows access to this trunk with "0" being the least restrictive

```
change route-pattern 32                                     Page   1 of   3
                    Pattern Number: 32  Pattern Name: To ASM
                              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: 32   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 M 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
```

4.9. Configure Location and Public Unknown Numbering

Use the "change locations" command to specify the SIP route pattern to be used as a default SIP route for the location corresponding to the Headquarters (Main) site. In this way, calls to non-numeric users or unknown domains will still be routed to Session Manager. Add an entry for the Main site if one does not exist already. Enter an appropriate Timezone Offset and the route-pattern from **Section 4.8** for **Proxy Sel. Rte. Pat**. Retain default values for the remaining fields. Submit these changes.

```
change locations
                                   LOCATIONS

                ARS Prefix 1 Required For 10-Digit NANP Calls? y

 Loc  Name           Timezone Rule  NPA                                Proxy Sel
 No                   Offset                                            Rte Pat
 1:  Main            + 00:00   0                                        32
                                                            Subaddress
 1: y y y y y n  n            rest                                        none
 2: y y y y y n  n            rest                                        none
```

Use the "change public-unknown-numbering 0" command, to define the calling party number to be sent to the AudioCodes MP-118. In the example shown below, all calls originating from a 5-digit extension beginning with 3 and routed across any trunk group (**Trk Grp(s)** setting is blank) will result in a 5-digit calling number. The calling party number will be in the SIP "From" header. Submit these changes.

```
change public-unknown-numbering 0                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                            Total
Ext Ext           Trk       CPN             CPN
Len Code          Grp(s)    Prefix          Len
                                                     Total Administered: 3
 5  3                                       5           Maximum Entries: 9999
```

4.10. Configure Automatic Alternate Routing (AAR)

Use the "change aar analysis 0" command to add entries to specify use of the route pattern for the specified dialed numbers. Add the dialed numbers to access each branch location (for the sample branch in the test configuration "1908543400" covers both the branch voice number 19085434000 and the branch fax number 19085434009); add another entry to cover calls to the voice messaging hunt group extension ("33000"). Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Dialed String**: Dialed prefix digits to match on
- **Total Min**: Minimum number of digits
- **Total Max**: Maximum number of digits
- **Route Pattern**: The route pattern number from **Section 4.8**
- **Call Type**: "aar"

```
change aar analysis 0                                          Page   1 of   2
                              AAR DIGIT ANALYSIS TABLE
                                    Location:  all           Percent Full:    1

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    1908543400            11   11    32       aar          n
    33000                 5    5     32       aar          n
    4                     7    7     999      aar          n
    400                   5    5     32       aar          n
    4000                  5    5     41       aar          n
```

Use the "change dialplan analysis" command to define "8" as a feature access code. This will be used for AAR dialing as defined above. Note also the 3xxxx extension range defined in this form for the Headquarters phones the feature access code "9" for ARS dialing defined in **Section 4.11**

```
change dialplan analysis                                       Page   1 of  12
                              DIAL PLAN ANALYSIS TABLE
                                    Location:  all           Percent Full:    1

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

Use the “change feature-access-codes” command to assign the feature access code “8” to AAR and the feature access code “9” to ARS.

```
change feature-access-codes                                      Page   1 of   8
                              FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: *01
         Abbreviated Dialing List2 Access Code: *02
         Abbreviated Dialing List3 Access Code: *03
Abbreviated Dial - Prgm Group List Access Code: *04
                      Announcement Access Code: *05
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA: *13    All: *11    Deactivation: *12
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code:
                       Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code:
                        Change COR Access Code:
                    Change Coverage Access Code:
            Conditional Call Extend Activation:        Deactivation:
                  Contact Closure   Open Code:          Close Code:
```

4.11. Configure Automatic Route Selection (ARS)

The ARS table entries are defined for local and long distance dialing to PSTN. The “change ars analysis n” command is used to make routing entries in the ARS table where “n” is the dialed digit string. The ARS table as used in the sample configuration is shown below. PSTN calls (1 + 10 digits) will match the **Dialed String** of 1 with 11 digits and use **Route Pattern** 3 for routing. Route Pattern 3 is configured to use a Trunk Group that connects to the T1/E1 network interface in the Avaya Media Gateway at the Headquarters location for PSTN terminations. The configuration of Route Pattern 3, the associated Trunk Group for PSTN calls, and the T1/E1 interface on the Avaya Media Gateway are out of scope of these Application Notes and are therefore not included.

```
change ars analysis 1                                            Page   1 of   2
                            ARS DIGIT ANALYSIS TABLE
                                  Location:  all           Percent Full:    2

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    1                     11   11    3        hnpa         n
    101xxxx0              8    8     deny     op           n
    101xxxx0              18   18    deny     op           n
    101xxxx01             16   24    deny     iop          n
    101xxxx011            17   25    deny     intl         n
    101xxxx1              18   18    deny     fnpa         n
    10xxx0                6    6     deny     op           n
    10xxx0                16   16    deny     op           n
```

4.12. Administer Incoming Call Handling Treatment

Use the "change inc-call-handling-trmt trunk-group 32" command to specify treatment of incoming calls to the Headquarters Communication Manger from Session Manager on trunk group 32 as defined in **Section 4.7.2**. These calls routed from the Session Manager can be:

- Calls to the branch access number 19085434000 that have exceeded the maximum number of calls allowed by the Call Admission Control for the branch Location (see **Section 5.2**). These calls are therefore re-directed by Session Manager to the Headquarters Communication Manager for termination at the Modular Messaging hunt group extension 33000.
- Branch calls to Headquarters with the 11-digit dialed number 1908123xxxx, where 908123 indicate that the call is destined for the Headquarters. In this case, the leading 6 digits should be stripped from the dialed string 1908123xxxx, so the call will be routed to the 5-digit extension 3xxxx (including the Modular Messaging hunt group extension 33000).
- Branch calls to PSTN with the 11-digit dialed number 1xxxxxxxxxx that conforms to the North America Numbering Plan. In this case, an ARS access code "9" should be added to the front of the dialed number for routing the call to PSTN through the ARS table as defined in **Section 4.11**.

```
change inc-call-handling-trmt trunk-group 32                    Page   1 of  30
                      INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 tie             11 19085434000      11  33000
 tie             11 1908123          6
 tie             11 1                    9
 tie
 tie
```

4.13. Configure Voice Messaging Hunt Group

Use the "add hunt group n" command to add a hunt group to be used by the voice messaging coverage path to be defined in the next section, where "n" is an unused hunt group number. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Number**: An unassigned hunt group number
- **Group Name:** A meaningful name
- **Group Extension**: An unassigned extension number
- **Group Type**: "ucd-mia"
- **ISDN/SIP Caller Display**:: "mbr-name"

```
add hunt-group 32                                          Page   1 of  60
                                   HUNT GROUP

            Group Number: 32                              ACD? n
              Group Name: Modular Messaging              Queue? n
         Group Extension: 33000                         Vector? n
              Group Type: ucd-mia                 Coverage Path:
                      TN: 1          Night Service Destination:
                     COR: 1                     MM Early Answer? n
           Security Code:                Local Agent Preference? n
 ISDN/SIP Caller Display: mbr-name
```

On **Page 2**, assign the following field values:

- **Message Center**: "sip-adjunct"
- **Voice Mail Number:** **Group Extension** from Page 1
- **Voice Mail Handle**: **Group Extension** from Page 1
- **Routing Digits**: AAR feature access code from **Section 4.10**

Submit these changes.

```
add hunt-group 32                                          Page   2 of  60
                                   HUNT GROUP





                      Message Center: sip-adjunct

     Voice Mail Number        Voice Mail Handle         Routing Digits
                                                   (e.g., AAR/ARS Access Code)
     33000                    33000                     8
```

4.14. Configure Voice Messaging Coverage Path

Use the "add coverage path n" command to specify a coverage path to be used for telephone users, where "n" is an unused coverage path. This will specify use of the voice messaging hunt group. Enter the hunt group number defined in Section 4.13 in Point1 under COVERAGE POINTS. Default values can be used for the remaining fields. It may be desirable to adjust the Number of Rings before a no-answer call goes to coverage.

```
add coverage path 32
                                  COVERAGE PATH

                   Coverage Path Number: 32
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA

    Station/Group Status    Inside Call     Outside Call
             Active?            n              n
               Busy?            y              y
       Don't Answer?            y              y         Number of Rings: 4
                All?            n              n
 DND/SAC/Goto Cover?            y              y
   Holiday Coverage?            n              n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h32            Rng:    Point2:
  Point3:                        Point4:
  Point5:                        Point6:
```

The following sample station form illustrates how to configure voice mail coverage for a given station user. Set **Coverage Path 1** to the value of the coverage path defined above.

```
change station 34002                                          Page   1 of   5
                                    STATION

Extension: 34002                        Lock Messages? n               BCC: 0
     Type: 6408D+                       Security Code:                  TN: 1
     Port: 01A0907                    Coverage Path 1: 32              COR: 1
     Name: HQ-DCP                     Coverage Path 2:                 COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 2          Personalized Ringing Pattern: 1
             Data Module? n                     Message Lamp Ext: 34002
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english

          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n
```

5. Configure Avaya Aura™ Session Manager

This section provides the procedures for configuring Session Manager as provisioned in the sample configuration. Session Manager is comprised of two functional components: the Session Manager server and the System Manager management server. The configuration of Session Manager is performed via the System Manager Web interface. The configuration is then downloaded into Session Manager.

The following sections assume that Session Manager and System Manager have been installed and that network connectivity exists between the two platforms.

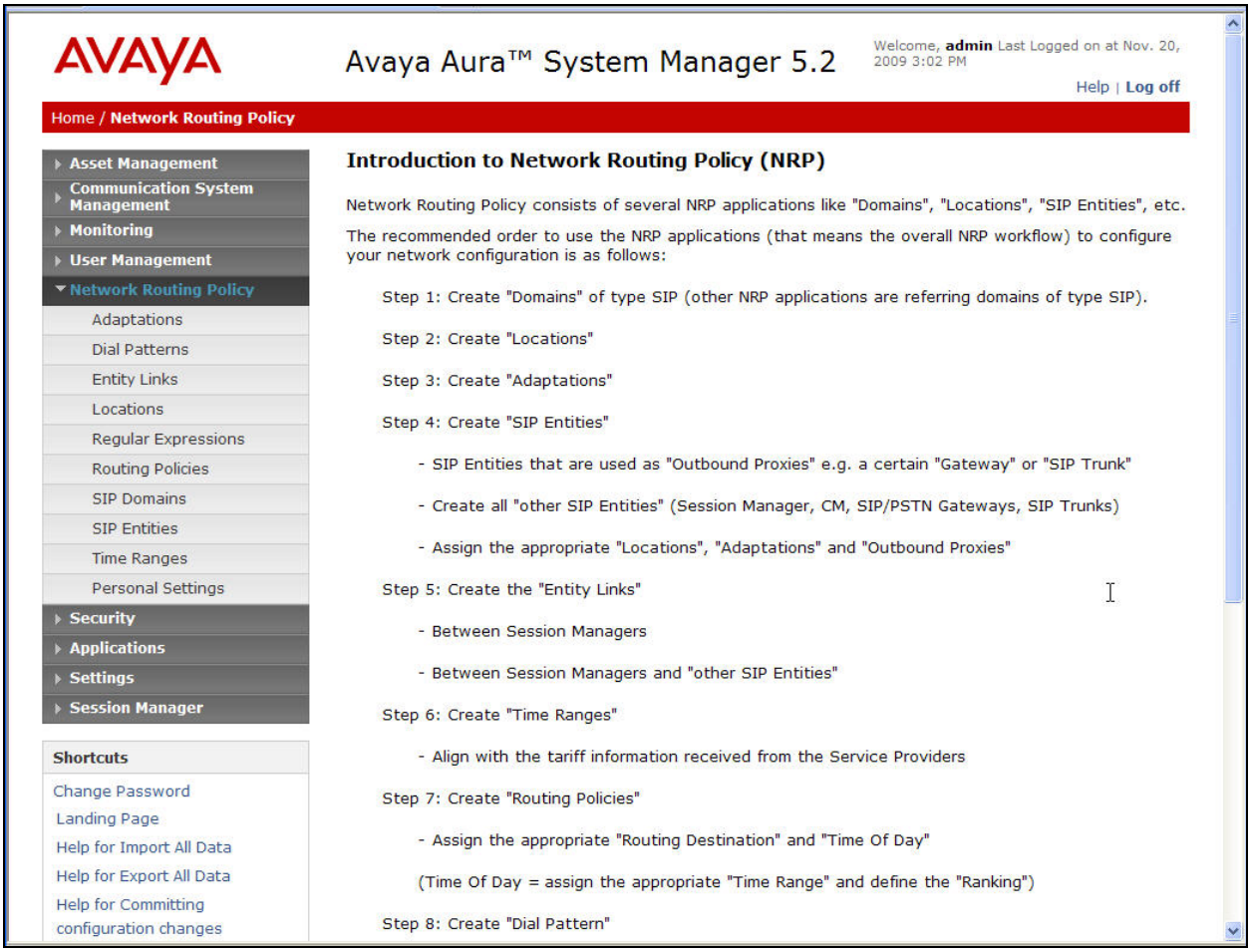
The Session Manager server contains an SM-100 security module that provides the network interface for all inbound and outbound SIP signaling to all provisioned SIP entities. For the Session Manager used in the sample configuration, the IP address assigned to the SM-100 interface is 10.1.2.170 as shown in **Figure 1**. The Session Manager server has a separate network interface used for connectivity to System Manager for managing/provisioning Session Manager. For the sample configuration, the IP address assigned to the Session Manager management interface is 10.1.2.171. The SM-100 interface and the management interface were both connected to the same IP network in the reference configuration. If desired, the SM-100 interface for real-time SIP traffic can be configured to use a different network than the management interface. For more information on Session Manager and System Manager, see **[1]** and **[2]**.

The procedures described in this section include configurations in the following areas:

- **SIP domain**
- Logical/physical **Locations** that can be occupied by SIP Entities
- **SIP Entities** corresponding to the SIP telephony systems including Communication Manager, branch AudioCodes MP-118 and Session Manager itself
- **Entity Links** defining the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- **Routing Policies** which control call routing between the SIP Entities
- **Dial Patterns** which govern to which SIP Entity a call is routed
- **Session Manager** corresponding to the Session Manager Servers managed by System Manager
- **Local Host Name Resolution** entries for host name to IP resolution

The Session Manager configuration described in these Application Notes does not include configuration on Session Manager for Avaya Modular Messaging. See **[10]** for details on Session Manager configuration for Modular Messaging.

Session Manger configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL “https://<ip-address>/SMGR”, where “<ip-address>” is the IP address of System Manager. Log in with the appropriate credentials and click on **OK** in the subsequent confirmation screen. The menu shown below is then displayed. Expand the **Network Routing Policy** link on the left side as shown. The sub-menus displayed in the left column will be used to configure the first six of the above items (**Sections 5.1** through **5.6**).

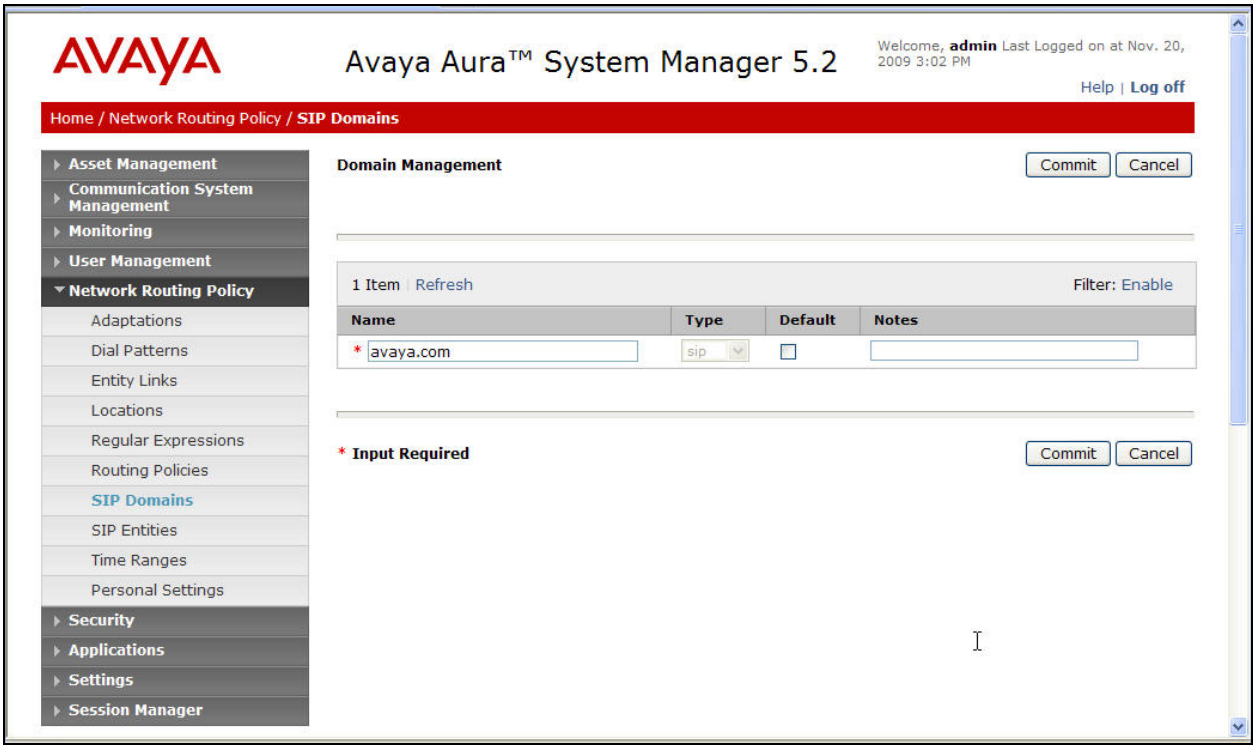


5.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Select **SIP Domains** on the left and click the **New** button (not shown) on the right. Fill in the following:

- **Name**: The authoritative domain name matching the domain configuration on Communication Manager (see **Section 4.6**)
- **Notes**: Descriptive text (optional)

Click **Commit**.



5.2. Add Location

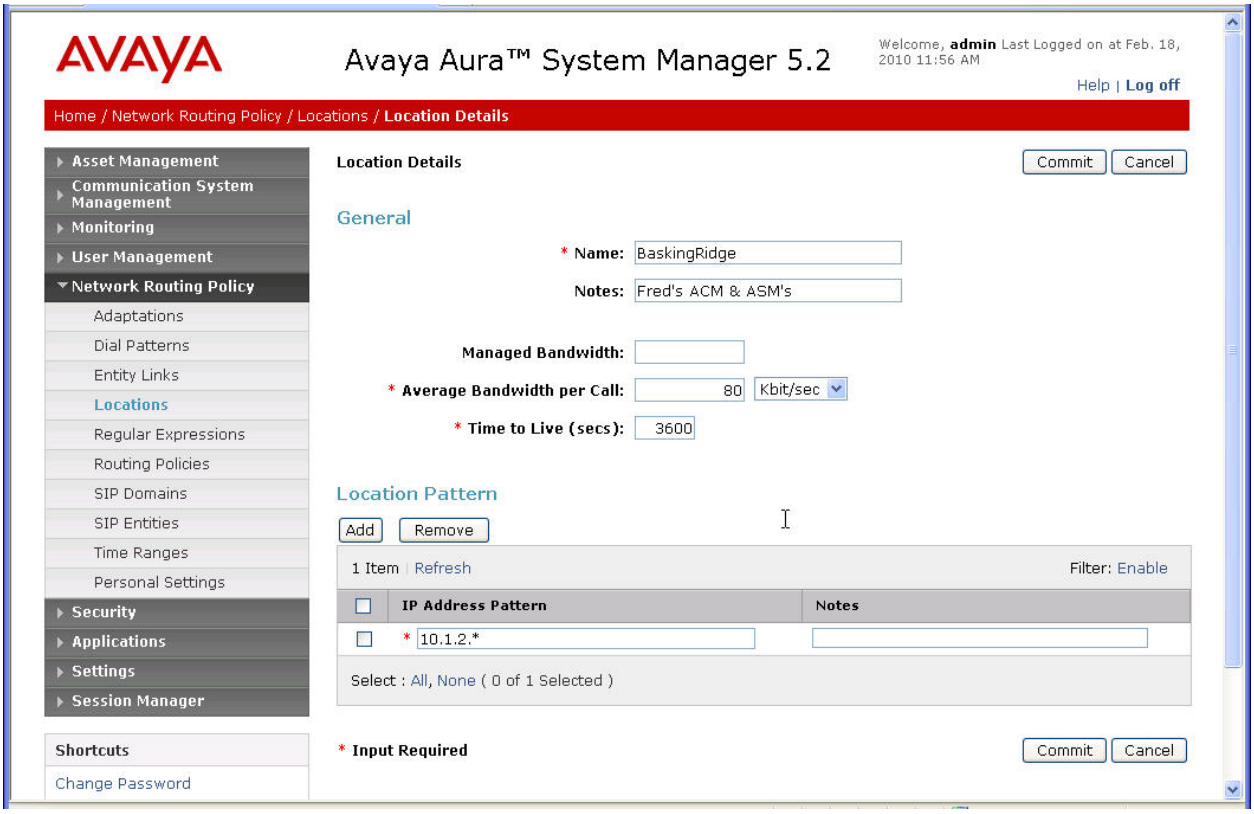
Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. To add a location, select **Locations** on the left and click on the **New** button (not shown) on the right.
Under *General*, enter:

- **Name**: A descriptive name
- **Notes**: Descriptive text (optional)

The remaining fields under *General* can be filled in to specify bandwidth management parameters between Session Manager and this location. These were used only for the branch Location in the sample configuration (see branch Location details below).
Under *Location Pattern*:

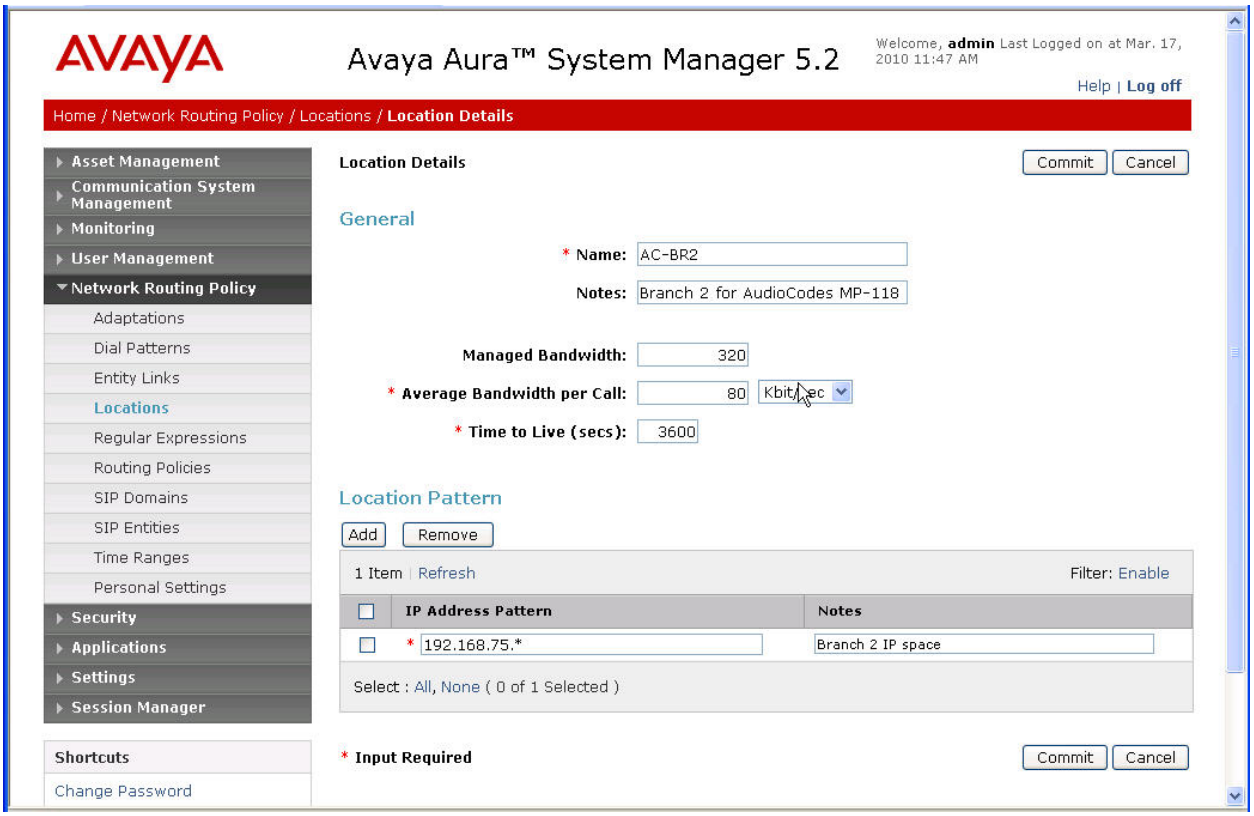
- **IP Address Pattern**: An IP address pattern used to identify the location
- **Notes**: Descriptive text (optional)

The screen below shows the addition of the “BaskingRidge” Location for the Headquarters site, which includes the Session Manager (10.1.2 subnet), Communication Manager (10.1.2 subnet), and all SIP telephones located at this location (10.1.2 subnet). Click **Commit** to save the Location definition.



In addition to the Location created for the Headquarters site, each branch needs to have its own Location defined. The screen below shows the addition of the "AC-BR2" location for the sample branch site with its own **Name** ("AC- BR2") and **IP Address Patterns** ("192.168.75.*").

The value 320 entered for **Managed Bandwidth** specifies a maximum of 4 simultaneous calls allowed for the branch location (with the default 80 Kbit/sec for the **Average Bandwidth per Call** as shown in the screen below). The 4 calls cover the 3 voice lines into the Avaya Partner Advanced Communication System from the 3 FXS ports on the AudioCodes MP-118 and the 1 fax connection from the 4th AudioCodes MP-118 FXS port. Any additional calls to the branch voice access number will be throttled by this Call Admission Control and be re-directed to the Headquarters Communication Manager for termination at the Modular Messaging system (see **Section 5.6**).



5.3. Add SIP Entities

A SIP Entity must be added for Session Manager and for each SIP-based telephony system supported by it using SIP trunks. In the sample configuration, a SIP Entity was added for the Session Manager itself, the Communications Manager, and the AudioCodes MP-118 for the test branch (in a real deployment, a separate SIP Entity must be added for each branch-located AudioCodes MP-118). Select **SIP Entities** on the left and click on the **New** button (not shown) on the right.

Under *General*:

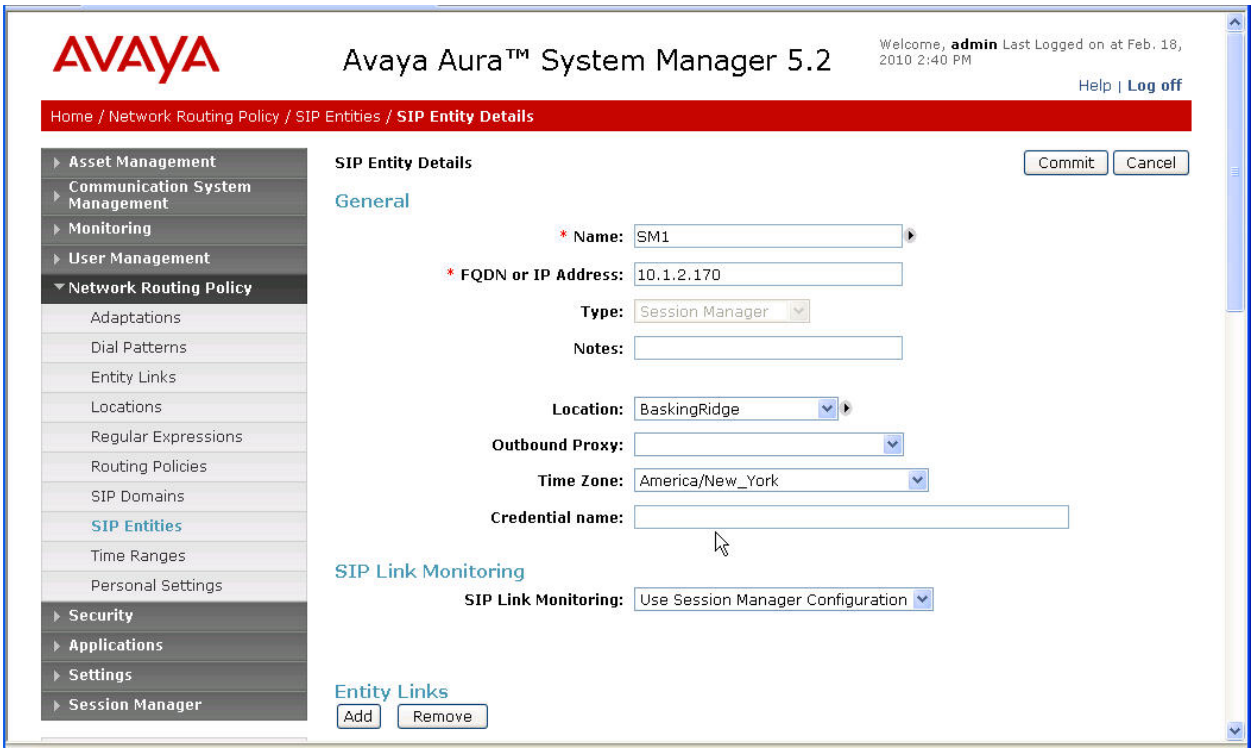
- **Name** A descriptive name
- **FQDN or IP Address**: FQDN or IP address of the Session Manager or the signaling interface on the telephony system
- **Type**: "Session Manager" for Session Manager, "CM" for Communication Manager, "Other" for AudioCodes MP-118
- **Adaptation**: Leave blank (Session Manager Entity does not have this field)
- **Location:** Select the Location configured in **Section 5.2**
- **Time Zone:** Select the proper time zone for this installation

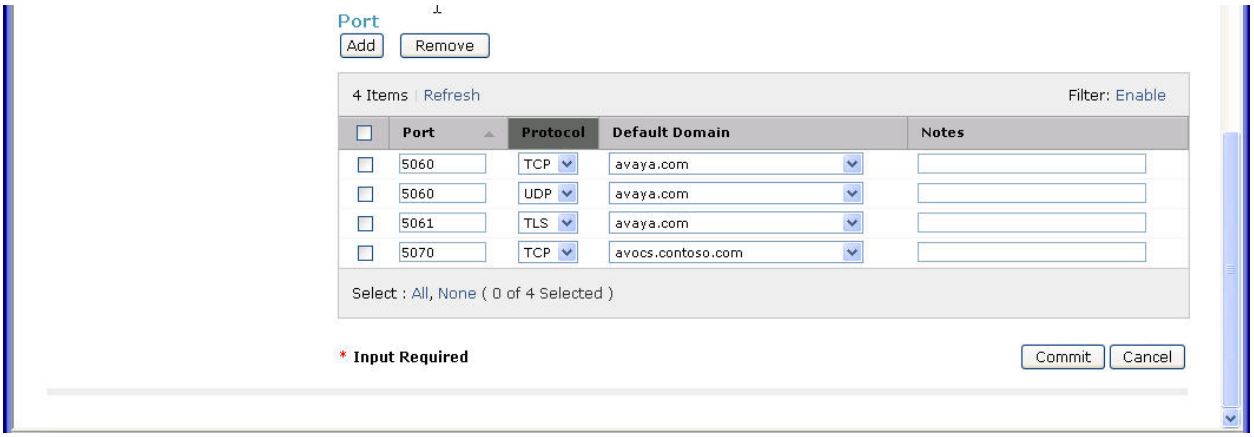
Under *Port* (for adding Session Manager Entity only), click **Add**, then edit the fields in the resulting new row as shown below:

- **Port**: Port number on which the system listens for SIP requests
- **Protocol**: Transport protocol to be used to send SIP requests
- **Default Domain**: Select the SIP Domain configured in **Section 5.1**

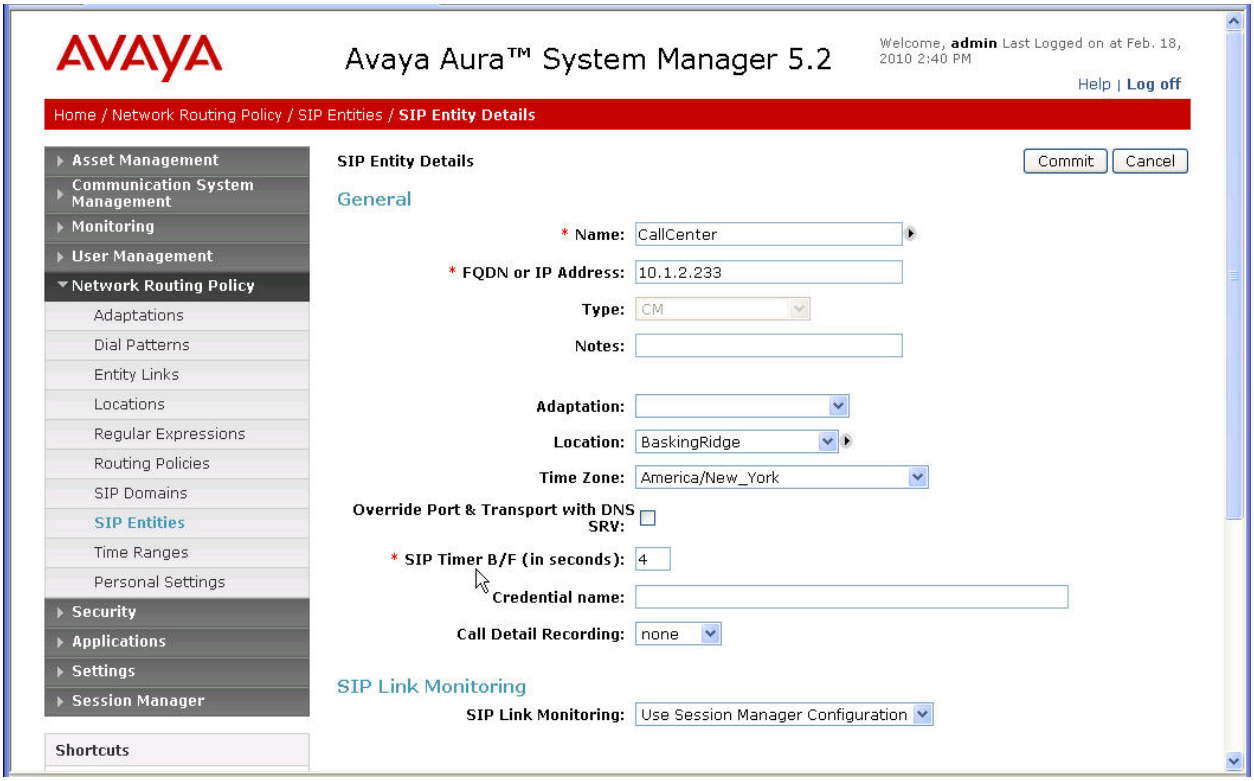
Default settings can be used for the remaining fields. Click **Commit** to save the SIP Entity definition.

The following screen shows the addition of Session Manager. The IP address of the SM-100 Security Module is entered for **FQDN or IP Address**. TCP port 5060 is used for communication with Communication Manager and branch-located AudioCodes MP-118. Note that only the first port entry under *Port* was used for the sample configuration; other port entries were configured for different purposes.

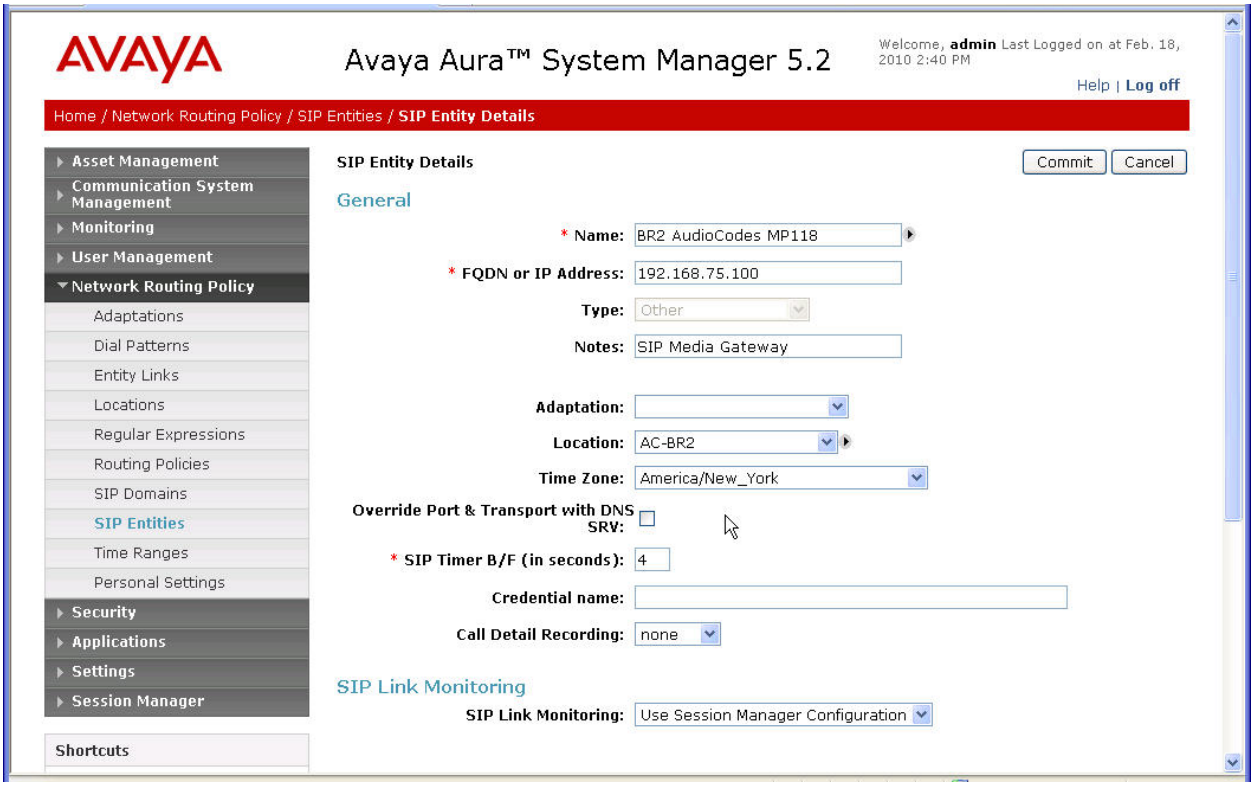




The following screen shows the results of adding Communication Manager. In this case, **FQDN or IP Address** is the IP address for the signaling interface "clan1" as defined in **Section 4.3**. For other Avaya Media Servers with the signaling interface integrated into the Communication Manager processor, the IP address of the Communication Manager should be specified. Note the "CM" selection for **Type**. Since this Communication Manager is for shared use, the **Name** "CallCenter" was used.



The following screen shows the results of adding the branch AudioCodes MP-118. In this case, **FQDN or IP Address** is the IP address assigned to the branch AudioCodes MP-118. Note the "Other" selection for **Type** as well as the selection of the branch Location as created in **Section 5.2**.



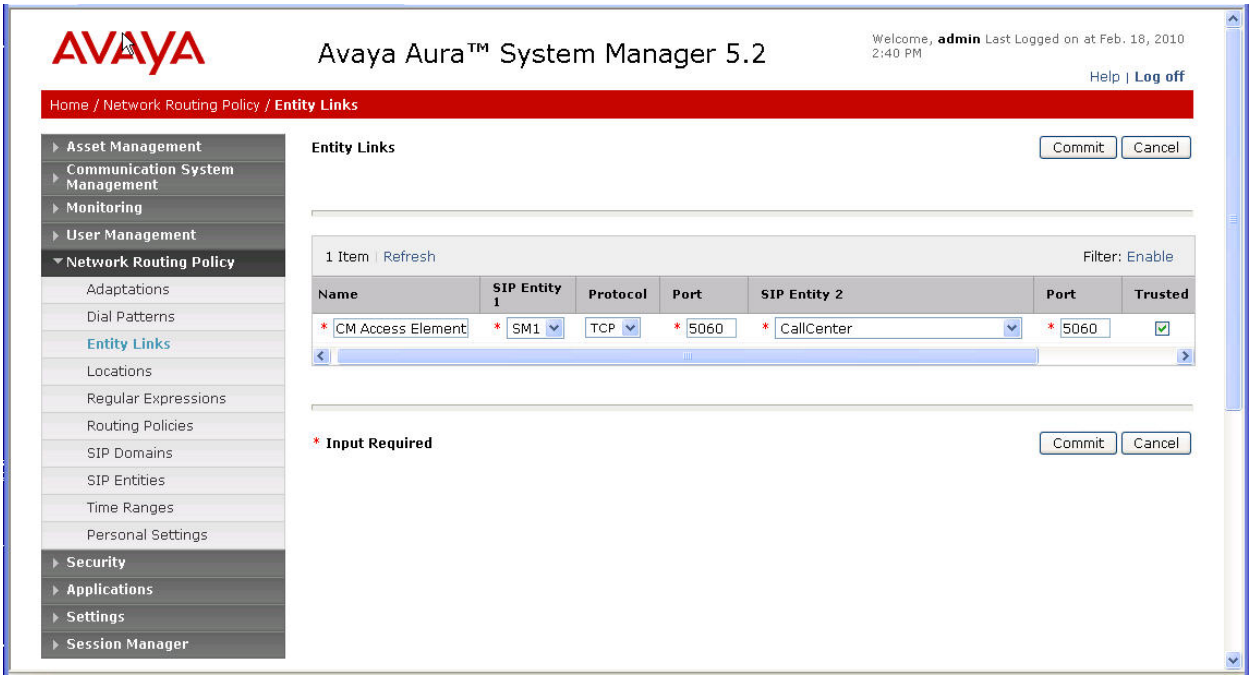
5.4. Add Entity Links

A SIP trunk between Session Manager and a telephony system is described by an Entity Link. In the sample configuration, 1 Entity Link was configured between Session Manager and Communication Manger (corresponding to the Signaling Group and the Trunk Group configured in Communication Manager in **Section 4.7**). In addition, a separate Entity Link should be created between Session Manager and the branch AudioCodes MP118 (one for each branch).

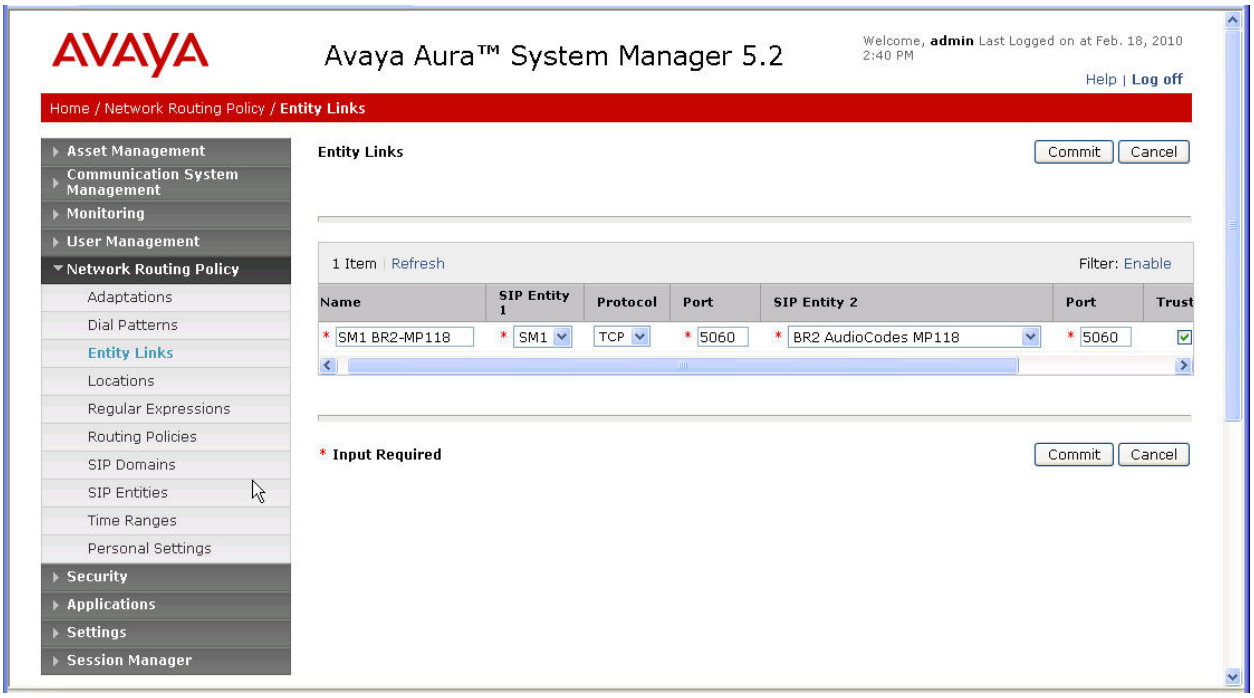
To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name**: A descriptive name
- **SIP Entity 1**: Select the Session Manager SIP Entity configured in **Section 5.3**
- **Protocol**: Select "TCP"
- **Port**: Port number to which the other system sends SIP requests.
- **SIP Entity 2**: Select the Communication Manager SIP Entity configured in **Section 5.3**
- **Port**: Port number on which the other system receives SIP requests.
- **Trusted**: Check this box

Click **Commit** to save the configuration. The screen below shows the Entity Link configured between Session Manager and Communication Manager.



The screen below shows the Entity Link between Session Manager and the Branch 2 AudioCodes MP-118.



5.5. Add Routing Policy

Routing policies describe the conditions under which calls will be routed to the SIP Entities. A routing policy must be added for routing calls, both voice and fax, to

- Communication Manager at the central site
- AudioCodes MP-118 at each branch

To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. Fill in the following on the displayed screen:

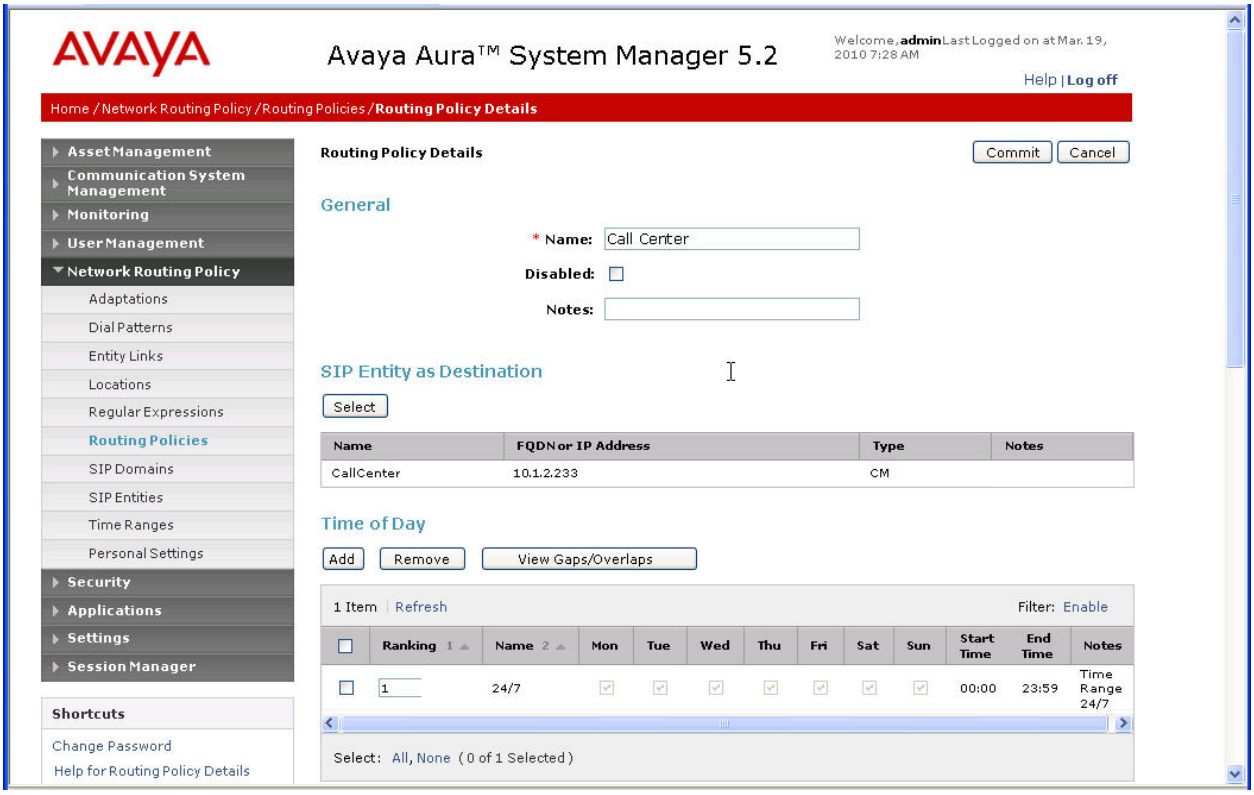
Under *General*:
Enter a descriptive name in **Name** and optional text in **Notes**.

Under *SIP Entity as Destination*:
Click Select, and then select the appropriate SIP entity to which this routing policy applies.

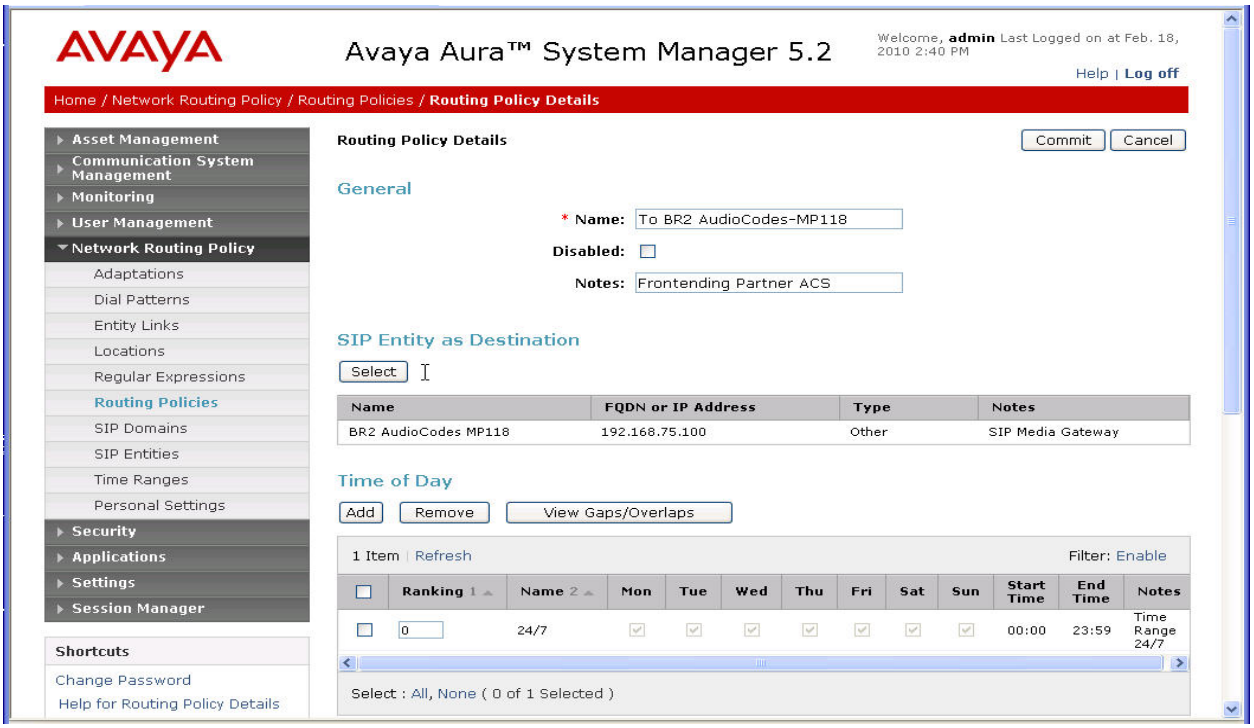
Under *Time of Day*:
Click **Add**, and select the default "24/7" time range, or add specific time ranges when the Routing Policy should be effective.

Defaults can be used for the remaining fields (except as noted below). Click **Commit** to save the Routing Policy definition.

The following screen shows the Routing Policy for routing calls to Communication Manger. Note the setting of "1" in the **Ranking** field for the *Time of Day* entry. This setting will be explained in **Section 5.6** on Dial Patterns.



The following screen shows the Routing Policy for routing calls to the AudioCodes MP-118 at Branch 2. Routing Policies for other branches are similarly configured (not shown).



5.6. Add Dial Patterns

Define Dial Patterns for matching called numbers. A Dial Patterns is then associated with one or more Routing Policies to direct calls to their destinations. For the sample configuration, following Dial Patterns are defined:

- 19085434000: voice call destined for Branch 2
- 19085434009: fax call destined for Branch 2
- 1908xxxxxxx: calls to Headquarters or PSTN destined for Communication Manager

One pair of the first 2 patterns with branch-specific number schemes (e.g., 17325551000 and 17325551009 for Branch 1) should be defined per branch.

For the sample configuration, the branch is restricted to calling PSTN with the 908 area code. The calls to 908123xxxx are calls to Headquarters (see **Section 4.12**). The 3rd Dial Pattern listed above addresses these 2 types of calls. This Dial Pattern should obviously be modified or new Dial Patterns be added to accommodate deployment-specific requirements (e.g, a different Headquarters number scheme, relaxation of the PSTN calling restrictions, etc.).

To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following, as shown in the screens below:

Under *General*:

- **Pattern**: Dialed number or prefix
- **Min**: Minimum length of dialed number
- **Max**: Maximum length of dialed number
- **SIP Domain**: SIP domain specified in **Section 5.1**
- **Notes**: Comment on purpose of dial pattern

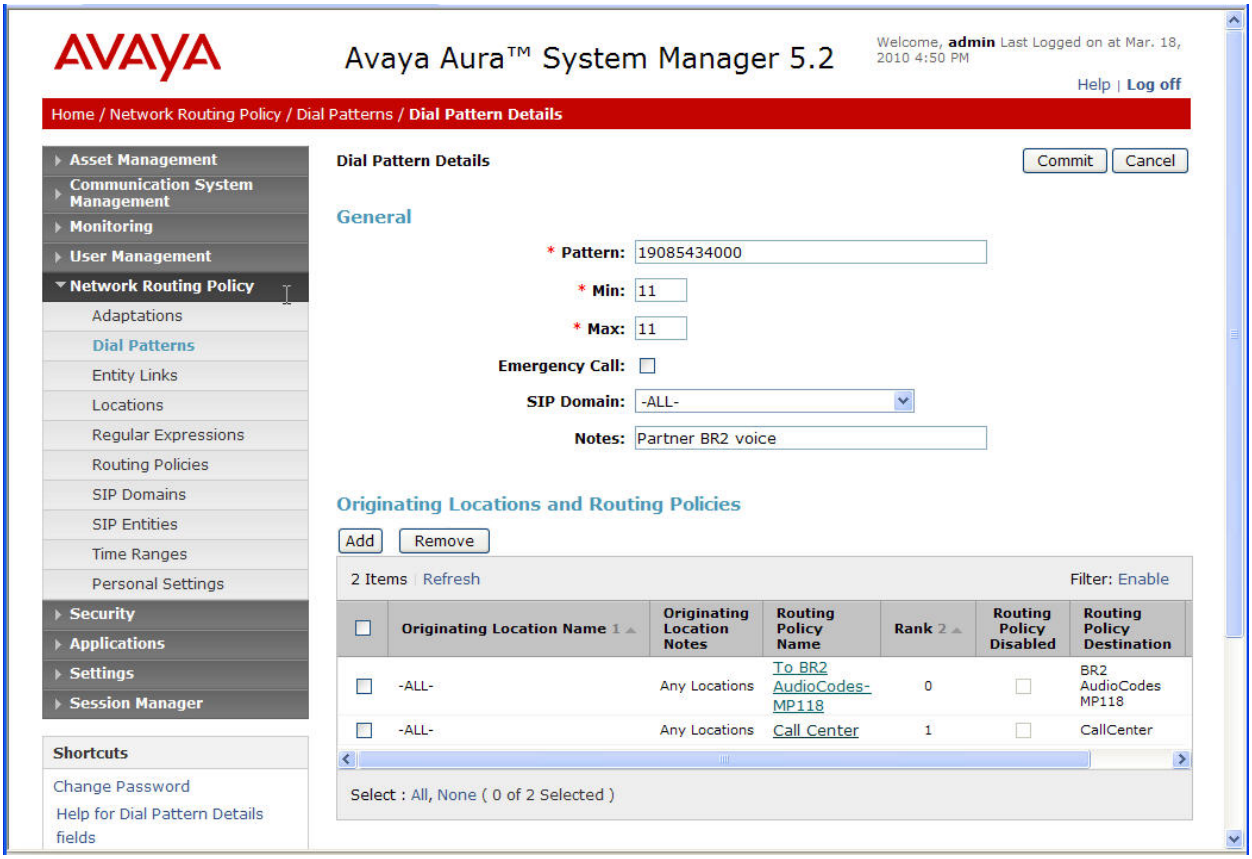
Under *Originating Locations and Routing Policies*:
Click **Add**, and then select the appropriate location (or "ALL") for **Originating Location Name** field and routing policy from the list.

Defaults can be used for the remaining fields. Click **Commit** to save the Dial Pattern. The following screens show the 3 Dial Patterns as listed above.

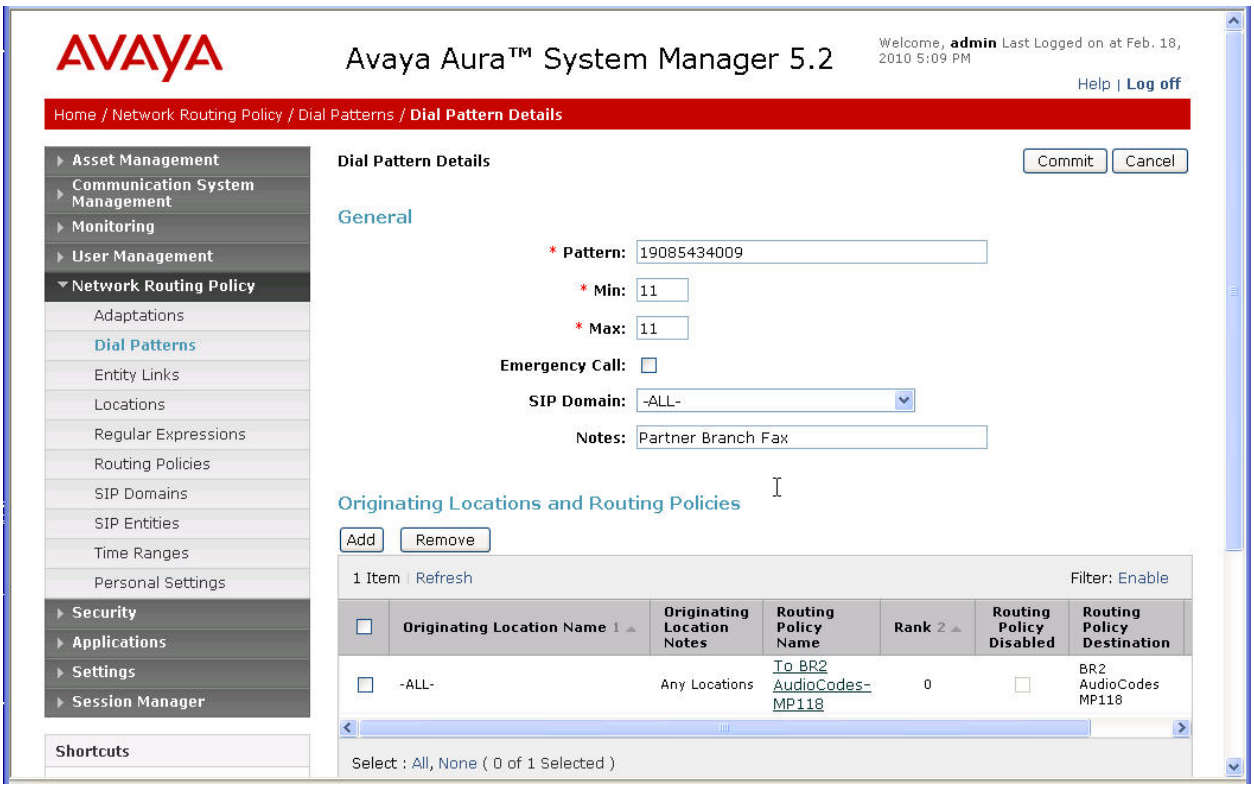
The following Dial Pattern for voice calls to Branch 2 is configured with 2 Routing Policies. The 1st Routing Policy (for the branch as destination) applies to calls below the Call Admission Control specified for the branch Location (see **Section 5.2**), i.e. 4 simultaneous calls allowed: 3 voice calls and 1 fax call. The 2nd Routing Policy (for Communication Manager at the Headquarters as destination) applies to calls when there are already 4 calls terminated to the branch. These additional voice calls above the branch call handling capacity are sent to the Communication Manager for onward routing to the Modular Messaging system at the Headquarters (see Section **4.12**).

Session Manager uses **Ranking** of the Routing Policy (administered with the associated *Time of Day* entry in Routing Policy configuration) to determine routing priorities. The "0" **Ranking** has the highest priority. In the sample configuration, when incoming calls to the branch are below the Call Admission Control threshold, the Routing Policy "To BR2 AudioCodes-MP118" will be applied (with **Ranking** "0"); when incoming calls to the branch exceed the Call Admission Control threshold, the Routing Policy "Call Center" will be applied (with **Ranking** "1" as administered in **Section 5.5**).

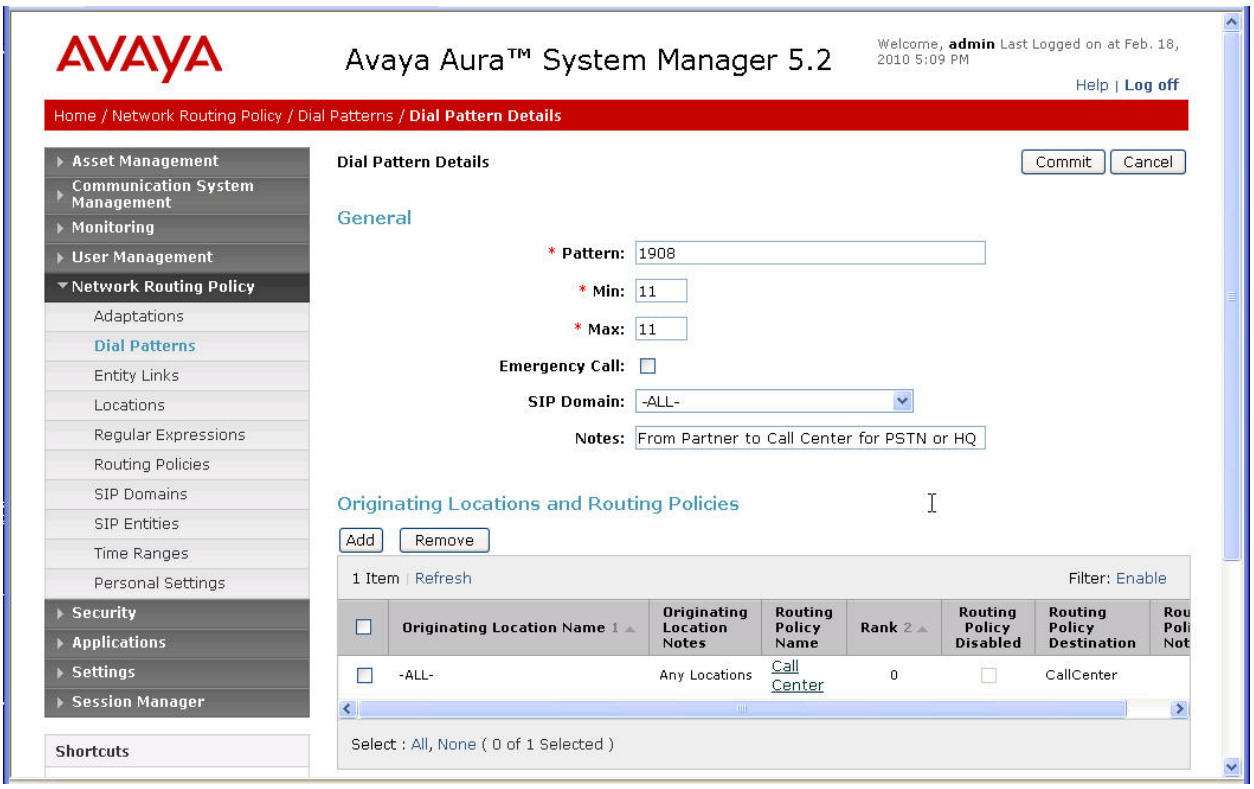
Note that if the 4th call to the branch access number is a voice call, it will be routed via the 1st Routing Policy to the branch. In that case, the branch AudioCodes MP-118 is configured to re-direct this 4th voice call to Modular Messaging at the Headquarters (see **Section 6.11**). It is explained in **Section 8.2** on Test Results why the sample configuration needs these combined configurations on the Session Manager and on the branch AudioCodes MP-118 to achieve call-redirection when no idle voice lines are available in the branch.



Dial Pattern for fax calls to Branch 2:



Dial Pattern for calls to Headquarters or PSTN destined for Communication Manager:



It should be noted that the sample configuration implemented a straightforward routing arrangement where calls are routed to configured destinations at all times (24/7). In real deployments, this simple routing arrangement might not be adequate, e.g., a business might want to route customer calls to the branches during normal business hours, but route off-hour calls to a Call Center or automatic voice response system at the business's Headquarters. This type of more sophisticated routing arrangements can be achieved by associating the same Dial Pattern (called number) with more than one Routing Policies (routing destinations) using appropriate Time Ranges assigned to Routing Policies. Rankings of assigned Time Ranges can be used to further refine routing arrangements. Please consult **[2]** for more information.

5.7. Add Avaya Aura™ Session Manager

Adding the Session Manager provides the linkage between System Manager and Session Manager. This configuration procedure should have already been properly executed if the Session Manager used has been set up for other purposes. This configuration step is included here for reference and completeness. To add Session Manager, expand the **Session Manager** menu on the left and select **Session Manager Administration**. Then click **Add** (not shown), and fill in the fields as described below and shown in the following screen (note that the screen below is for **Edit Session Manager** since it was already administered):

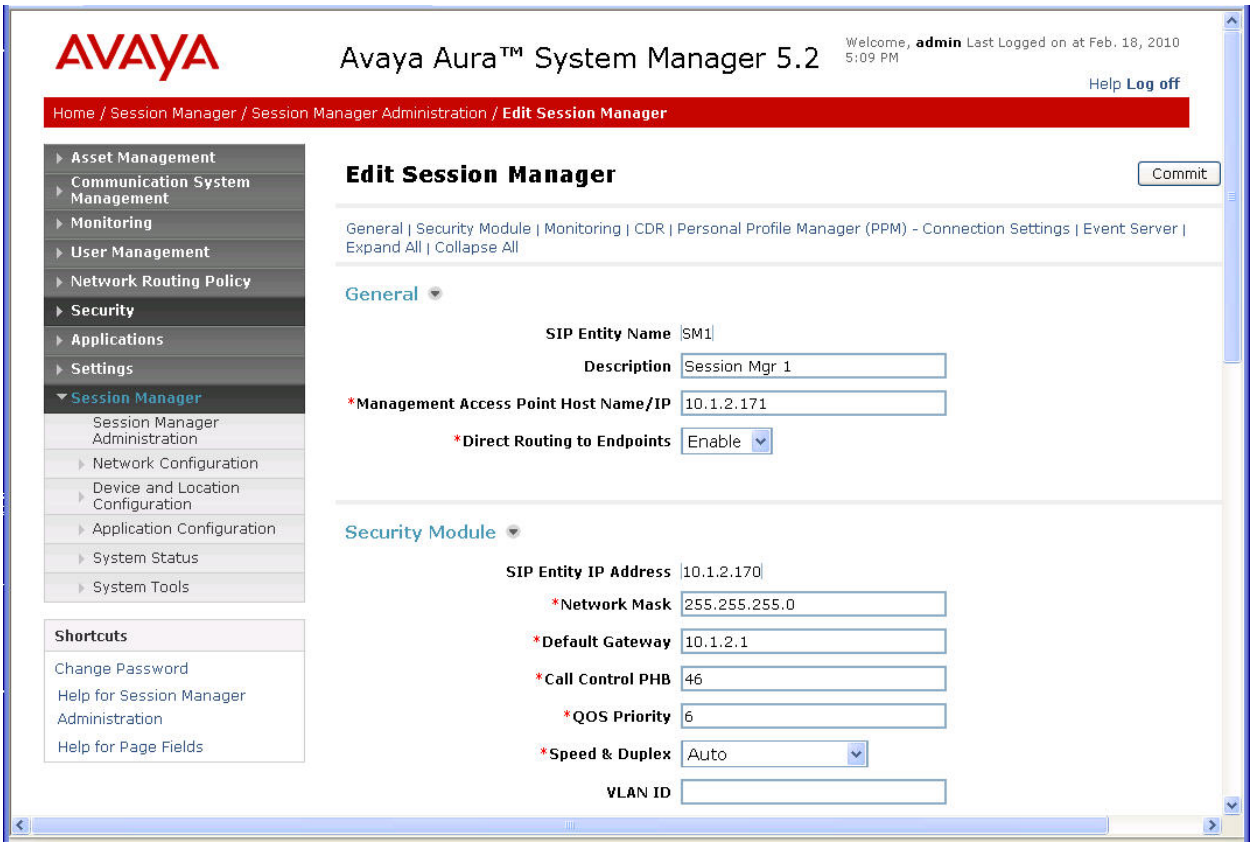
Under *General*:

- **SIP Entity Name**: Select the name of the SIP Entity created for Session Manager
- **Description**: Any descriptive text
- **Management Access Point Host Name/IP**: IP address of the Session Manager management interface.

Under *Security Module*:

- **Network Mask**: Enter the proper network mask for Session Manager.
- **Default Gateway**: Enter the default gateway IP address for Session Manager

Accept default settings for the remaining fields.

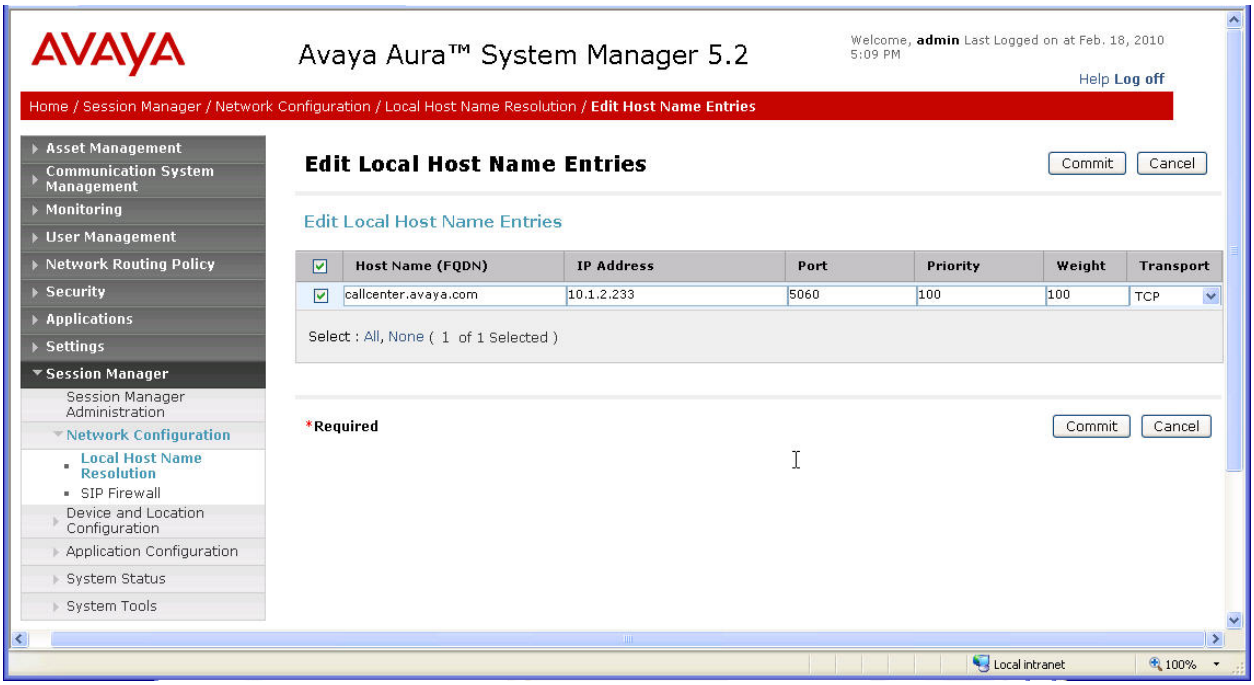


5.8. Define Local Host Name Resolution

The host names referenced in the definitions of the previous sections must be defined. To do so, Select **Session Manager → Network Configuration → Local Host Name Resolution** on the left. For each host name, click **New** and enter the following:

- **Host Name**: Name used for the host
- **IP Address**: IP address of the host's network interface
- **Port**: Port number to which SIP requests are sent
- **Transport**: Transport to be used for SIP requests

Defaults can be used for the remaining fields. The **Priority** and **Weight** fields are used when multiple IP addresses are defined for the same host. The following screen shows the host name resolution entry used in the sample configuration.



6. Configure AudioCodes MP-118

This section shows the necessary steps to configure the AudioCodes MP-118 to support the sample configuration. It is assumed that the basic configuration of the AudioCodes MP-118 has already been administered. See **[11]** and **[12]** for additional information. The icon on the AudioCodes MP-118 configuration screens contained in this section indicates the corresponding parameter value has been changed. All parameters with this icon shown in the following screens are relevant to the sample configuration. In some cases, the parameter values used are specific to the sample configuration and may not apply to all environments.

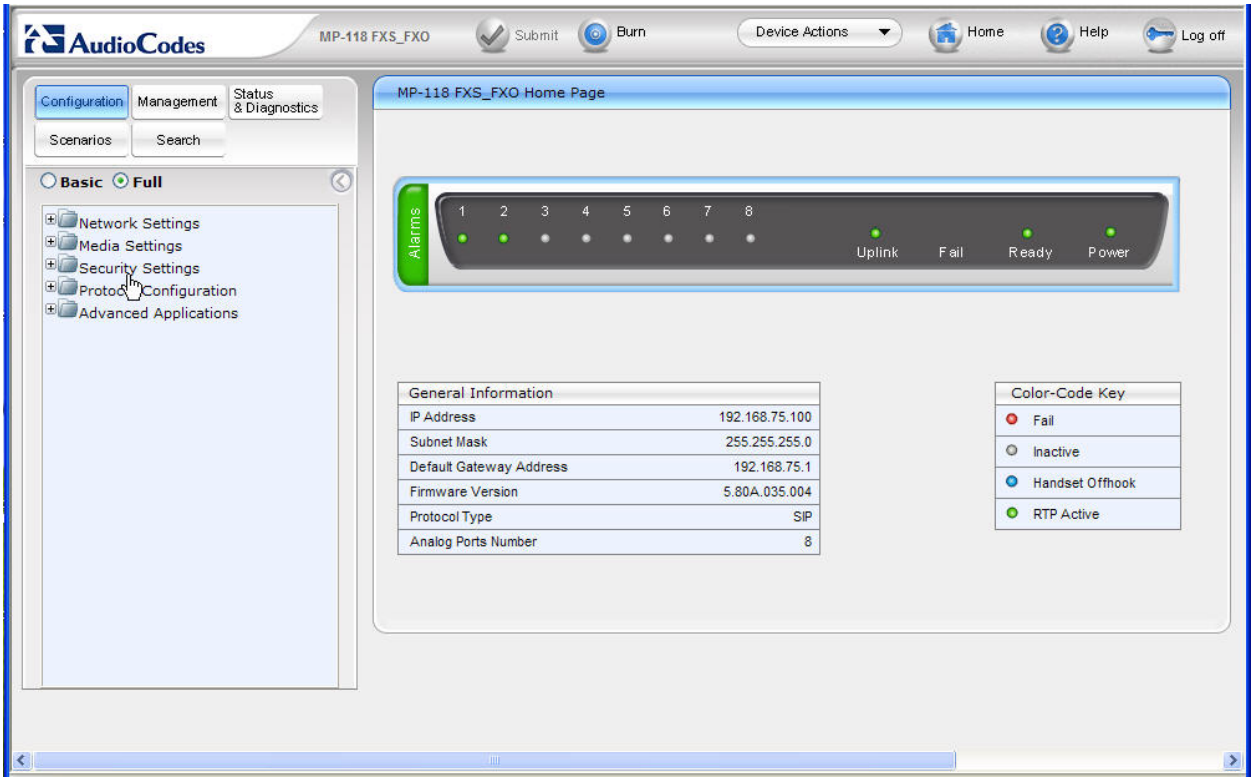
The administration procedures in this section include the following areas.

- Verify MP-118 Access
- Configure SIP General Parameters
- Configure Proxy & Registration
- Configure Proxy Sets Table
- Configure Coders
- Configure Advanced Parameters
- Dest Number IP → Tel Specification
- Configure IP to Hunt Group Routing
- Administer Endpoint Phone Numbers
- Configure Hunt Group Settings
- Configure Call Redirect
- Enable Caller ID Forwarding
- Modify .ini Configuration File
- Save Configuration Changes

6.1. Verify MP-118 Access

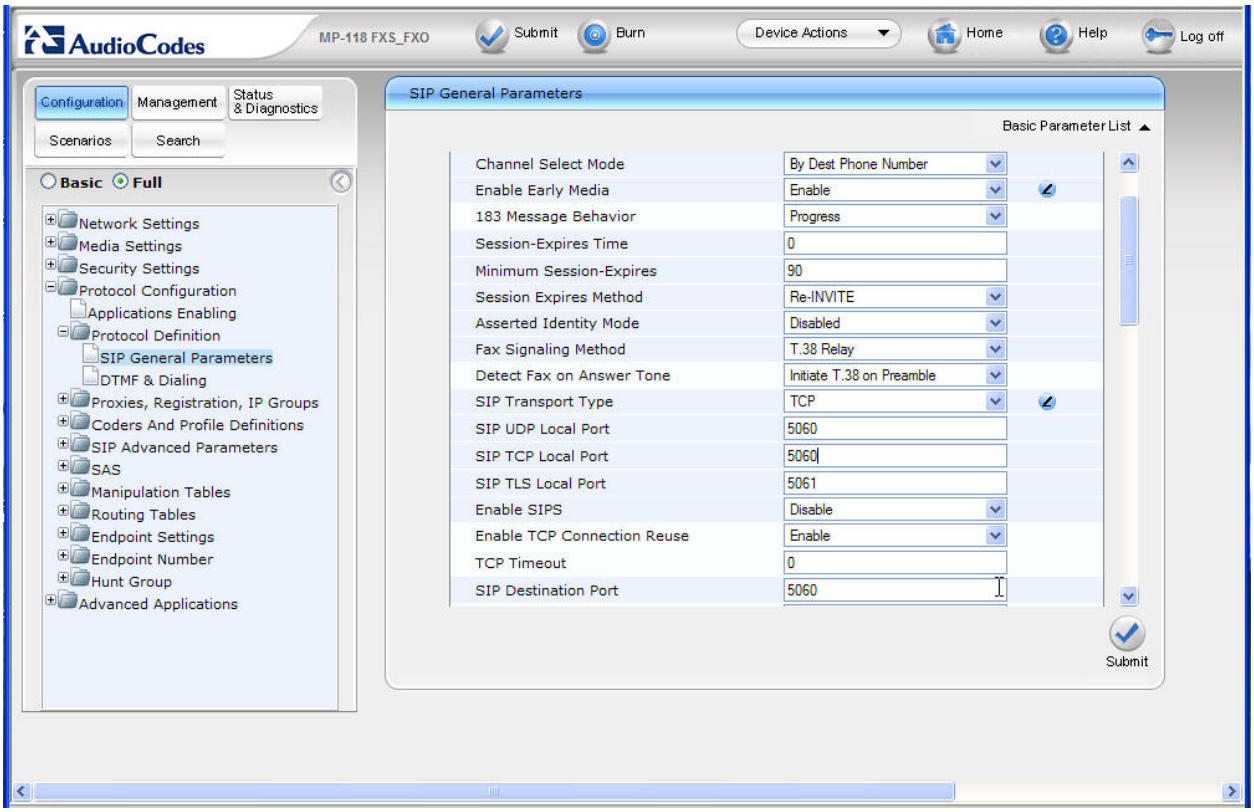
From a web browser, enter the AudioCodes MP-118 IP address in the URL. A pop-up login window will appear (not shown) to allow entering the appropriate User Name and Password to gain access to the MP-118 administration web pages. Default username is “Admin”. Default password is “Admin”.

Once logged in, select the **Full** radio button and **Configuration** tab from the left navigation panel. The example screen below was captured when two calls were up. One call was between a Headquarters phone and a Partner phone at the branch; the other call was between a PSTN user and a second Partner phone at the branch. Both calls went through a FXS port on the AudioCodes MP-118. This is the reason that ports 1 and 2 show green for “RTP Active”.



6.2. Configure SIP General Parameters

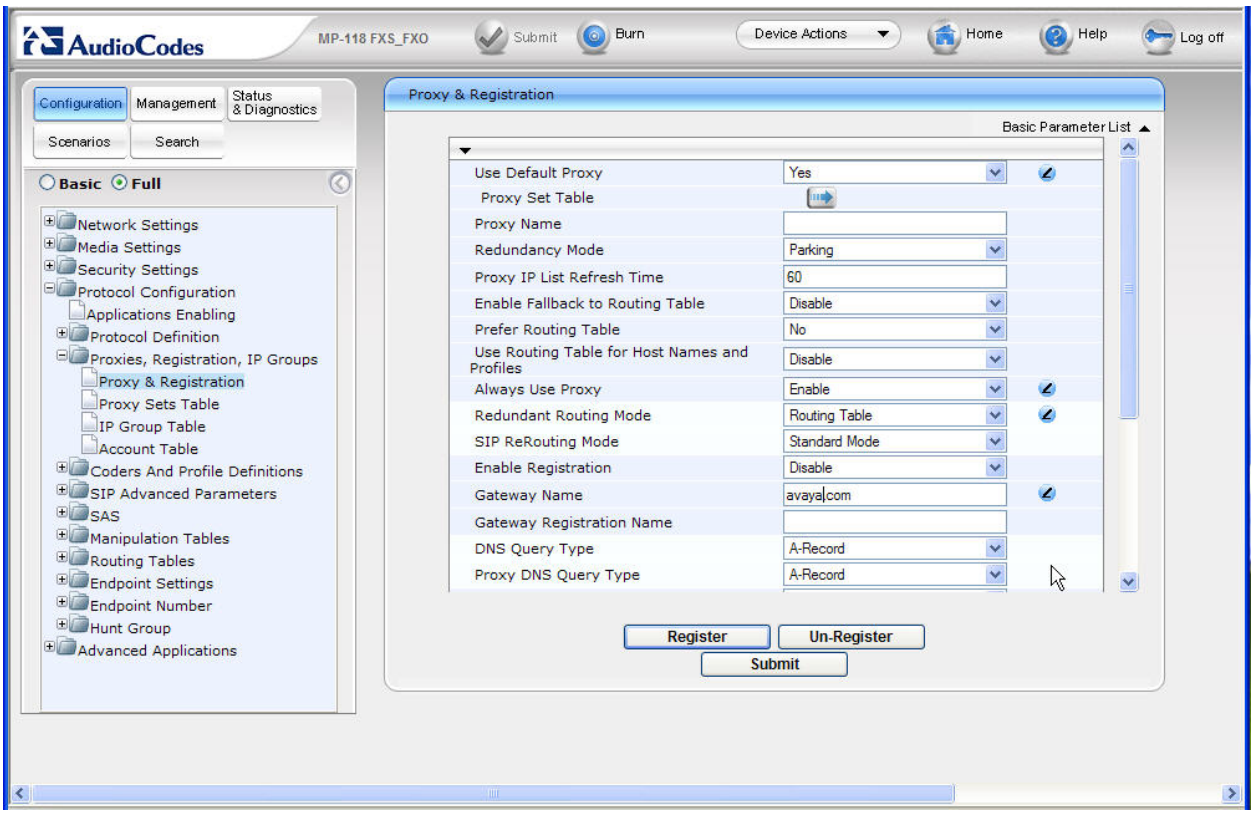
From the left navigation panel, navigate to the SIP General Parameters screen by selecting **Protocol Configuration → Protocol Definition → SIP General Parameters**. The values of the fields with an adjacent icon have been changed from the default. After making the necessary changes in the parameter settings, click the **Submit** button to make the changes effective (this applies to all configuration screens for AudioCodes MP-118). These key parameter values on this screen instruct the AudioCodes MP-118, when functioning as a media gateway, to use TCP as the transport and listen on port 5060 for SIP messages.



The remaining fields of the SIP General Parameters screens maintain the default values.

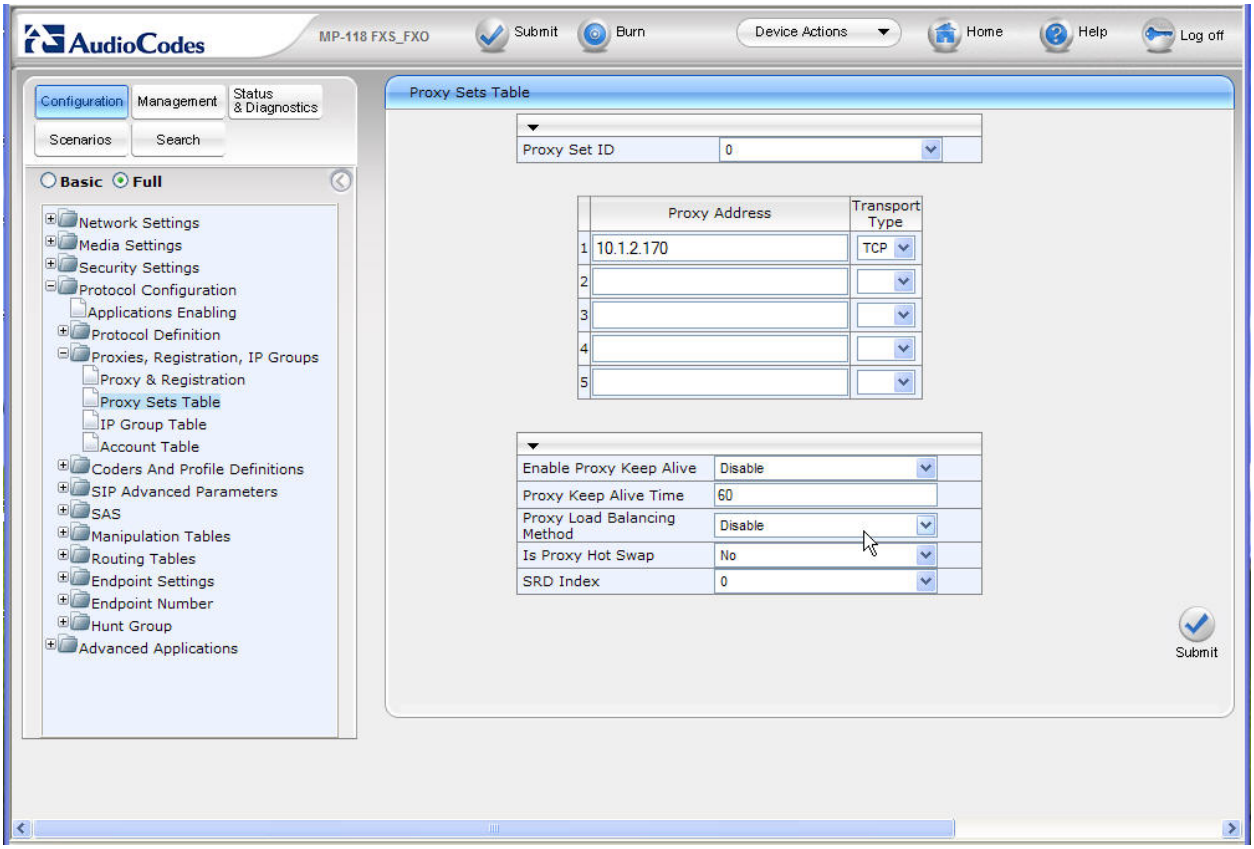
6.3. Configure Proxy & Registration

From the left navigation panel, navigate to the Proxy & Registration screen by selecting **Protocol Configuration → Proxies, Registration, IP Groups → Proxy & Registration**. The values of the fields with an adjacent icon have been changed from the default. The value of "avaya.com" specified for the **Gateway Name** parameter is the SIP Domain name used in the sample configuration and matches the SIP Domain name configured on Session Manager (**Section 5.1**) and Communication Manager (**Section 4.6**). This and other configured parameters instruct the AudioCodes MP-118 to always use SIP Proxy but SIP Registrar on MP-118 is disabled (the **Enable Registration** parameter is left at its default "Disable" setting) since no SIP endpoint registration is needed in the sample configuration.



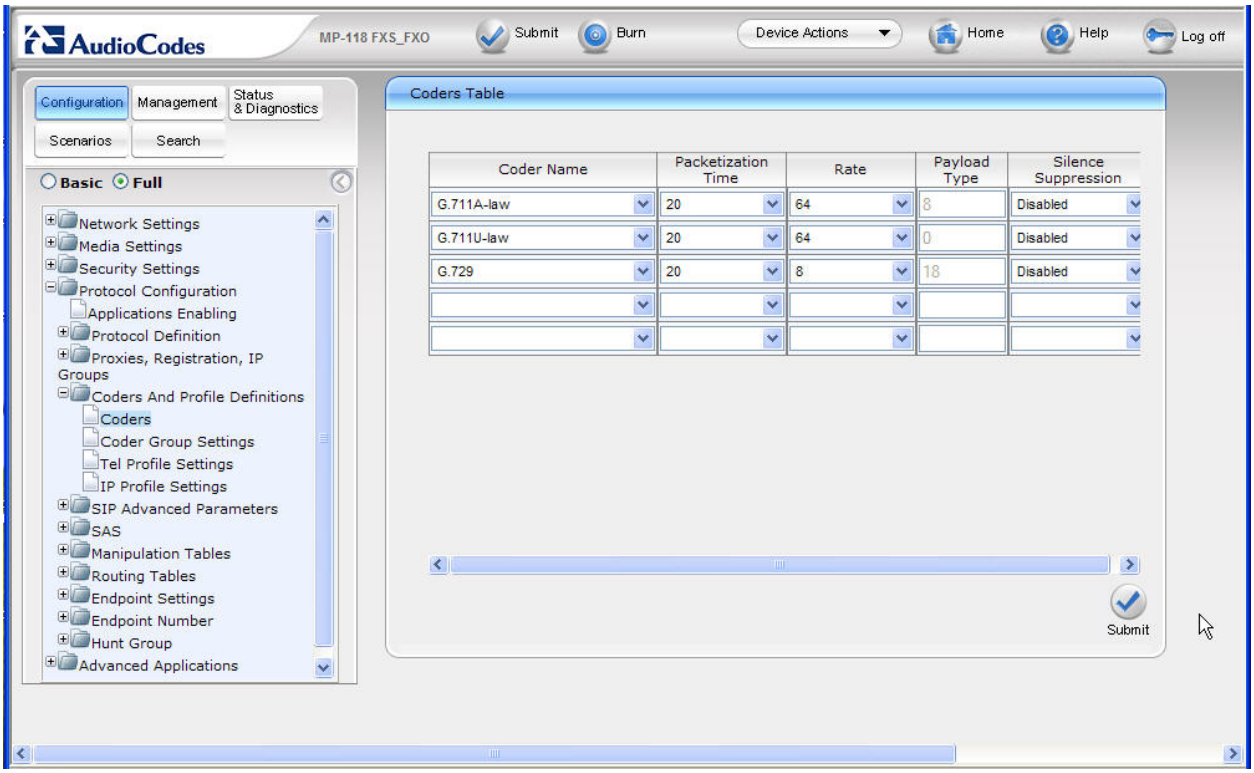
6.4. Configure Proxy Sets Table

From the left navigation panel, navigate to the Proxy Sets Table screen by selecting **Protocol Configuration → Proxies, Registration, IP Groups → Proxy Sets Table**. The Proxy Sets Table specifies the SIP Proxy server. Enter the IP addresses of the Session Manager in the **Proxy Address** table as shown below. Select "TCP" from the **Transport Type** drop-down list.



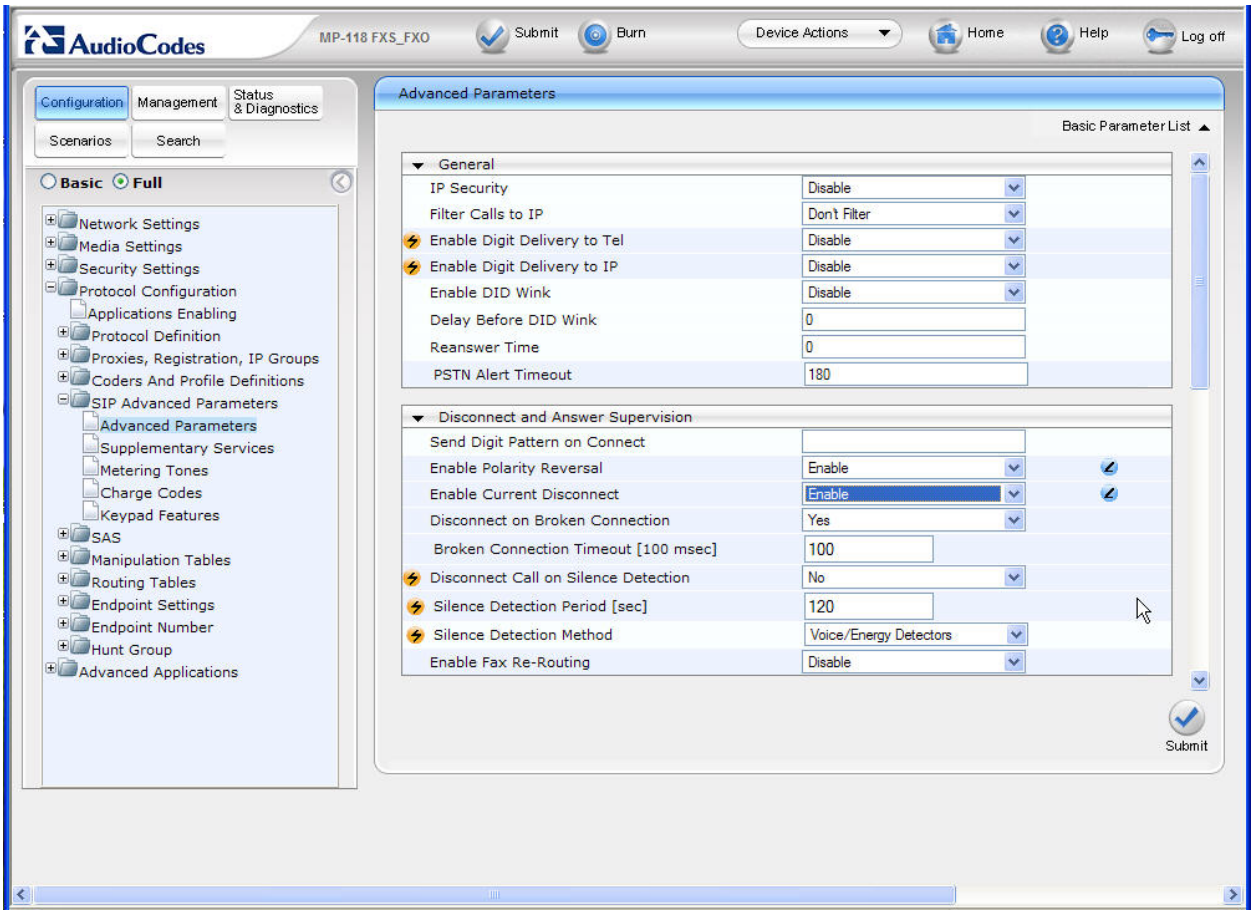
6.5. Configure Coders

From the left navigation panel, navigate to the Coders Table screen by selecting **Protocol Configuration → Coders And Profile Definitions → Coders**. Select the codecs from the drop-down list. There should be one entry that matches the codec configured on Communication Manager (see **Section 4.5**).



6.6. Configure Advanced Parameters

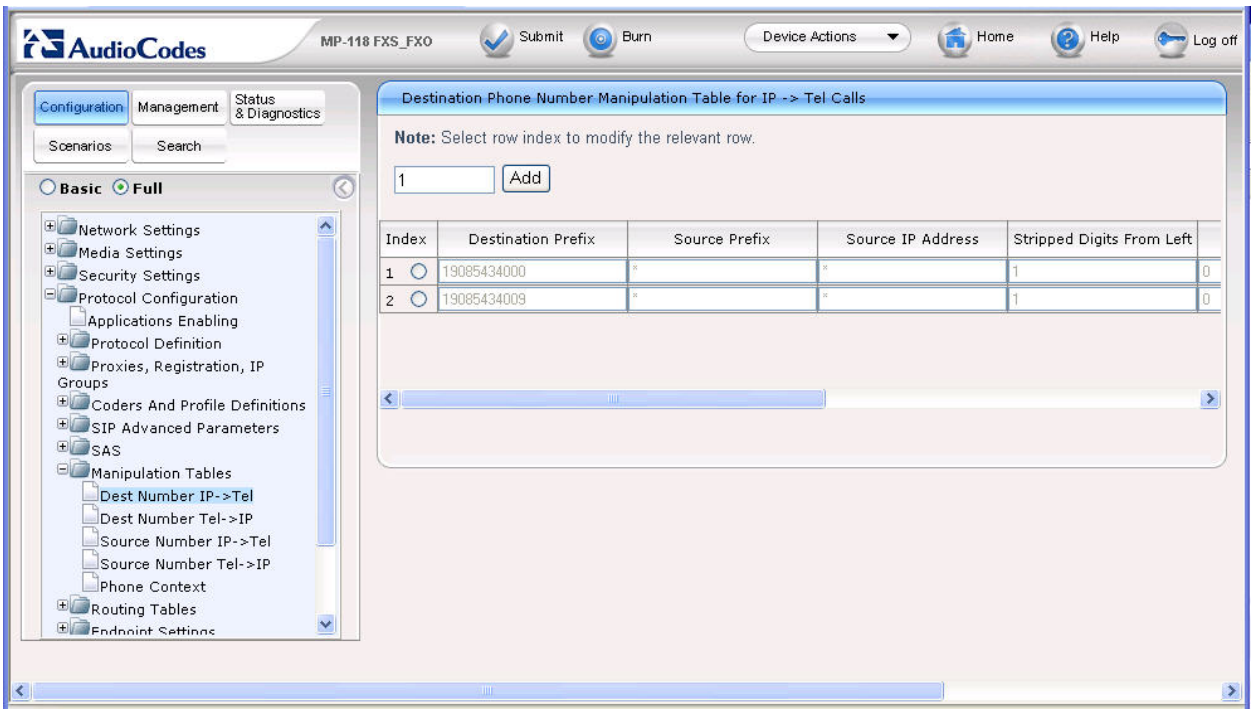
From the left navigation panel, navigate to the Advanced Parameters screen by selecting **Protocol Configuration → SIP Advanced Parameters → Advanced Parameters**. The values of the fields with an adjacent icon have been changed from the default for the sample configuration.



The remaining fields of the Advanced Parameters screens maintain the default values.

6.7. Dest Number IP → Tel Specification

From the left navigation panel, navigate to **Protocol Configuration → Manipulation Tables → Dest Number IP->Tel**. Add the voice and fax numbers for the branch with the leading 1 as delivered from Session Manager to the branch AudioCodes MP-118. The entries in this table strip the leading 1 from the dialed digit strings. For the sample configuration, the voice number for the Branch is 19085434000, the Branch fax number is 19085434009.

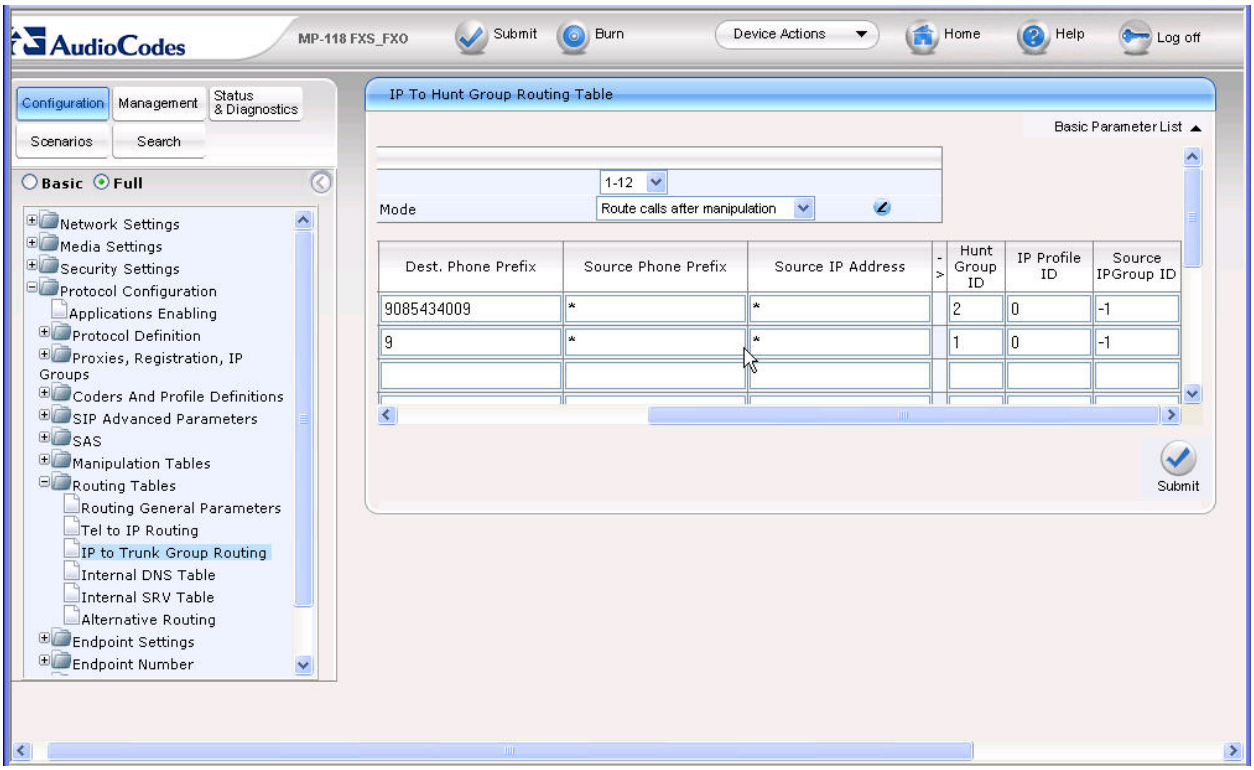


6.8. Configure IP to Hunt Group Routing

From the left navigation panel, navigate to the IP to Hunt Group Routing Table screen by selecting **Protocol Configuration → Routing Tables → IP to Trunk Group Routing**. The entries in this table are used by the AudioCodes MP-118 to route calls originating on IP and terminating on the gateway. Note that the AudioCodes "Hunt Group" concept is not the same as a "Hunt Group" in Communication Manager. The leading digits of the called numbers are used to determine the selected AudioCodes MP-118 Hunt Group. In the sample configuration, calls from IP (or Session Manager) to the branch fax machine 9085434009 is entered explicitly for routing to Hunt Group 2; voice calls to the branch access number 9085434000 is routed to Hunt Group 1.

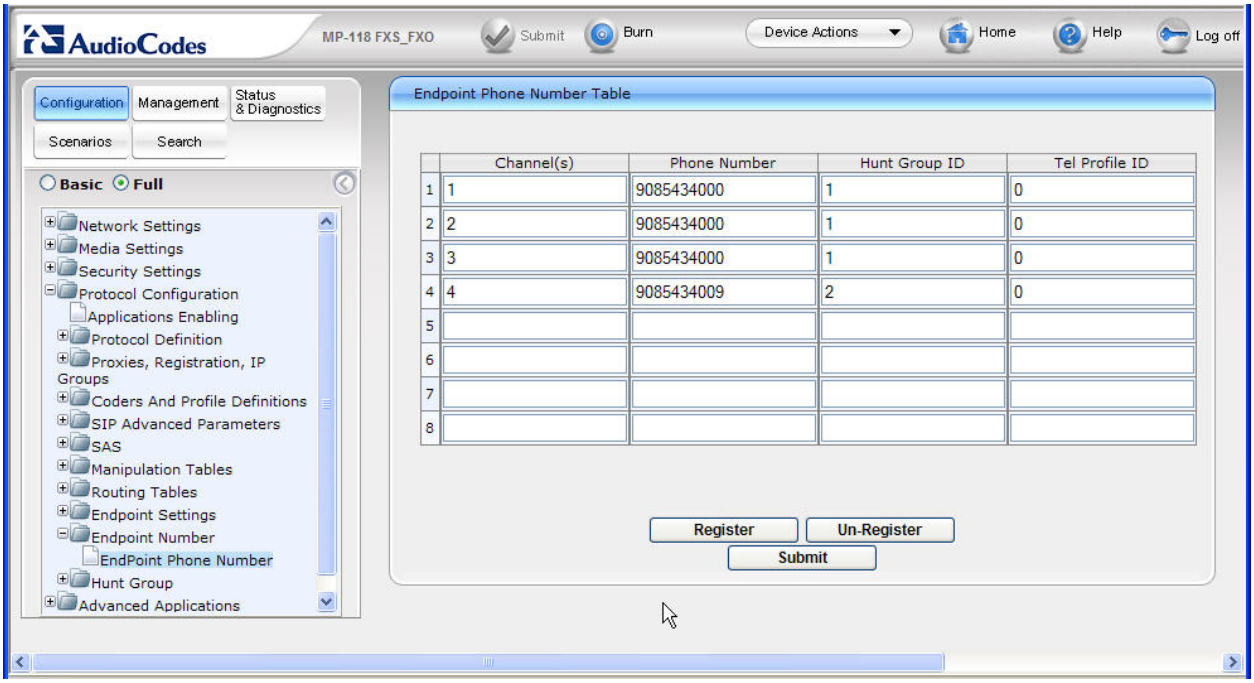
Hunt Group ID 1 consists of 3 AudioCodes MP-118 FXS ports that are connected to the branch Partner line ports. Hunt Group ID 2 consists of one FXS port that is directly connected to the branch fax machine. Channel (port) to hunt group associations are configured in **Section 6.9**. Hunt group settings are configured in **Section 6.10**. The table below shows a summary of the Hunt Group assignments.

Channel	Hunt Group ID
FXS 1, 2, 3	1
FXS 4	2
FXO 5, 6, 7, 8	Un-assigned



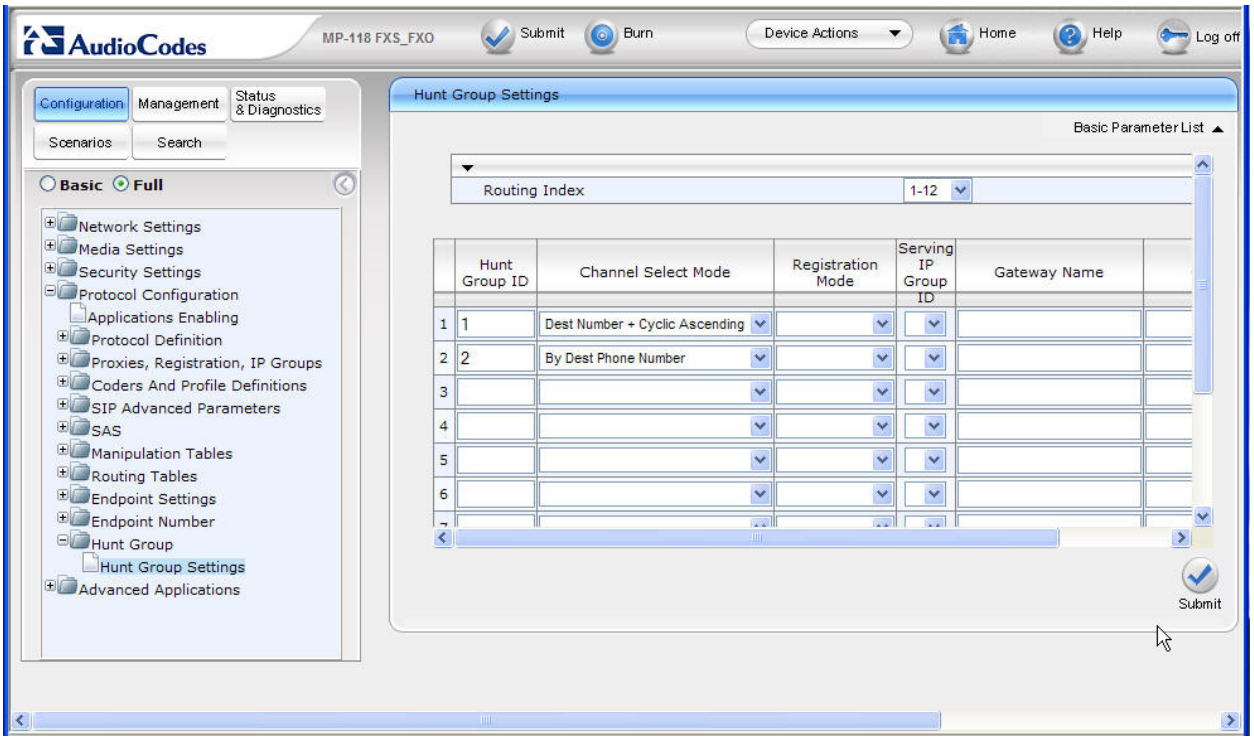
6.9. Endpoint Phone Number Specification

From the left navigation panel, navigate to the Endpoint Phone Number Table screen by selecting **Protocol Configuration → Endpoint Number → Endpoint Phone Number**. Enter the phone number assignment for each channel of the AudioCodes MP-118 as well as the associated Hunt Group ID. On AudioCodes MP-118, Channels 1 through 3 are the FXS interfaces to the branch Partner line ports for voice calls; Channels 4 is the FXS interface direct to the branch fax machine. The sample configuration used Channels 1 through 4 (FXS) only.



6.10. Configure Hunt Group Settings

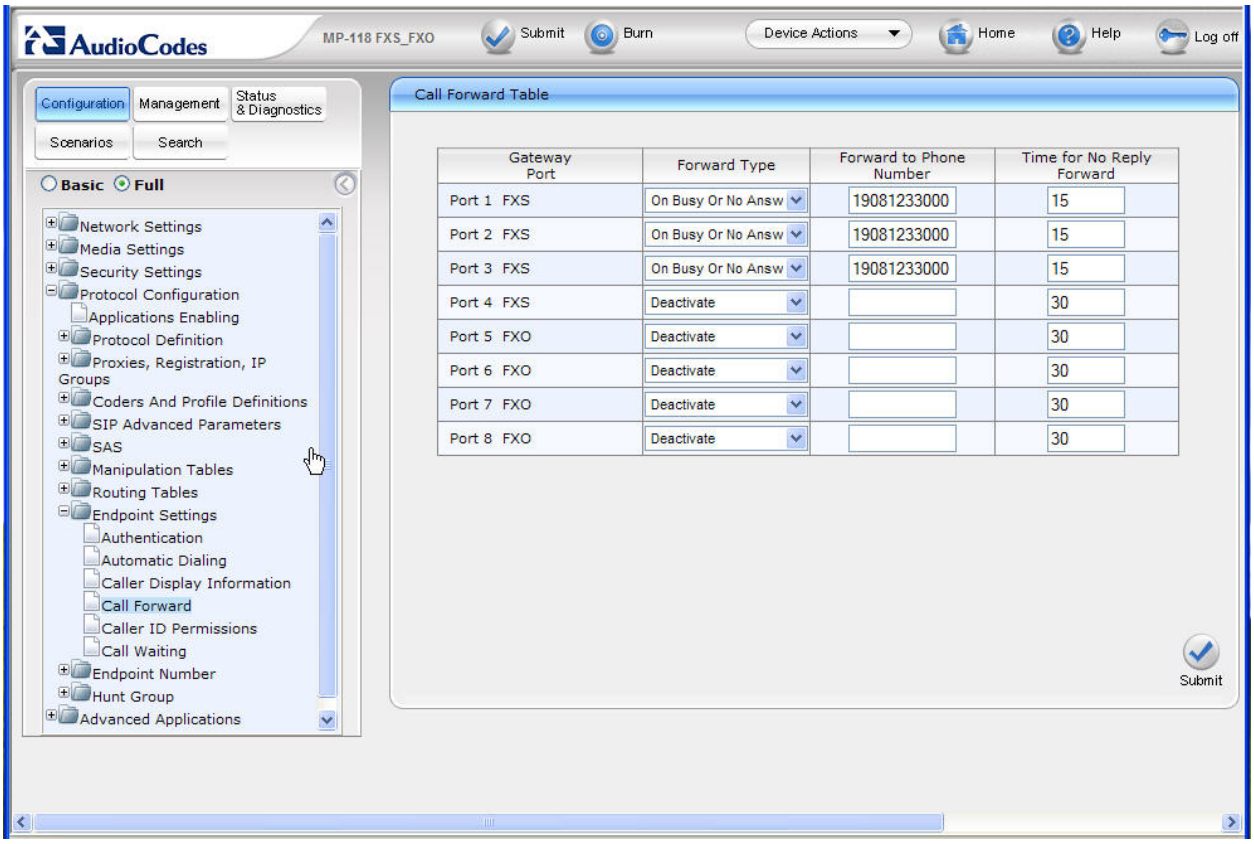
From the left navigation panel, navigate to the Hunt Group Settings screen by selecting **Protocol Configuration → Hunt Group → Hunt Group Settings**. The settings on this screen configure the method in which calls originating on IP and terminating on the gateway are assigned to channels within each Hunt Group. Hunt Group 1, containing 3 FXS interfaces to the branch Partner line ports, is configured to select any available interface in this Hunt Group by destination number and in a Cyclic Ascending order to terminate calls. Hunt Group 2, containing the 4th FXS interface to the branch fax machine, is configured to terminate calls based on the destination phone number.



6.11. Configure Call Redirect

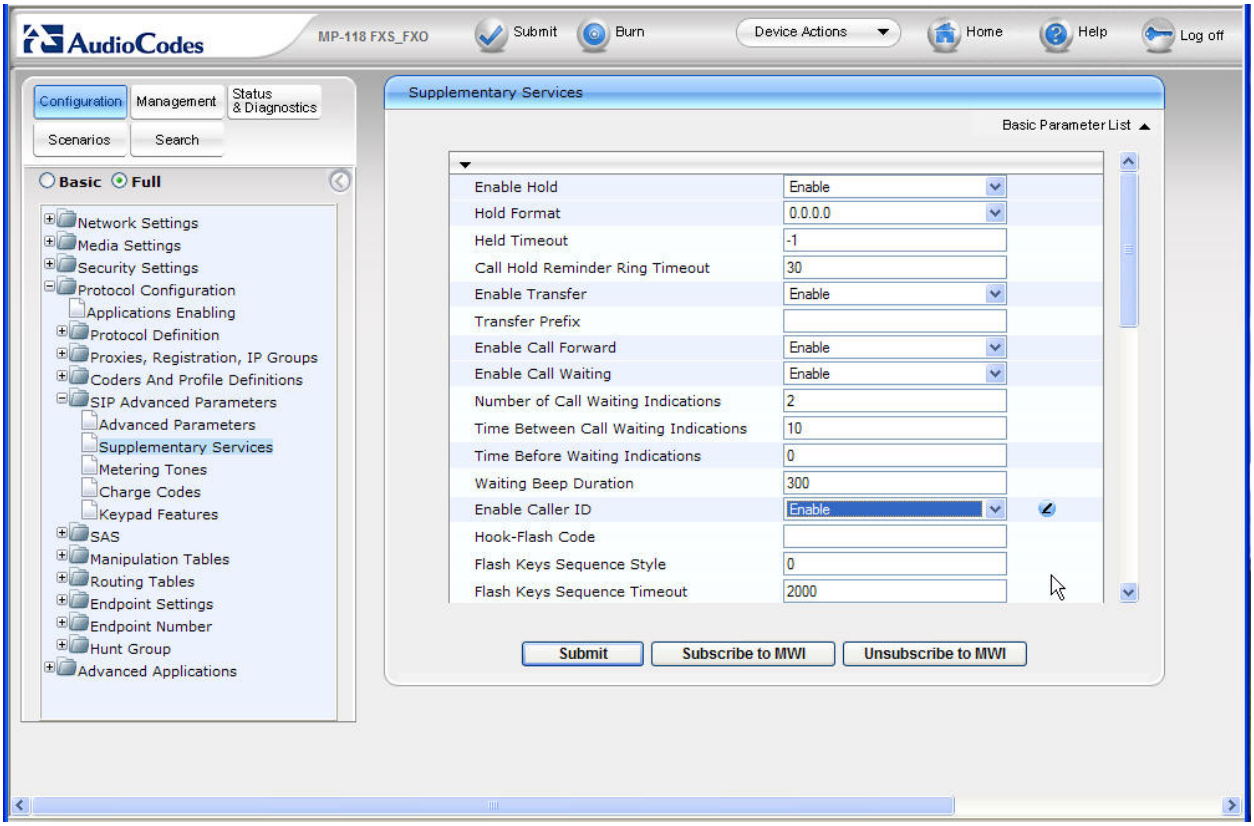
Navigate to **Protocol Configuration → Endpoint Settings → Call Forwarding** to configure redirection of unanswered or busy voice calls to the branch to a destination at the Headquarters location, like call center or an automatic voice response system.

For FXS ports 1 through 3 in the Call Forward Table, select "On Busy Or No answer" for **Forward Type**, enter the call redirection destination number (Modular Messaging access number for the sample configuration) at the Headquarters location for **Forward to Phone Number**, and enter the number of seconds for **Time for No Reply Forward** (15 seconds as configured for the sample configuration equate to about 4 rings before the call is redirected).



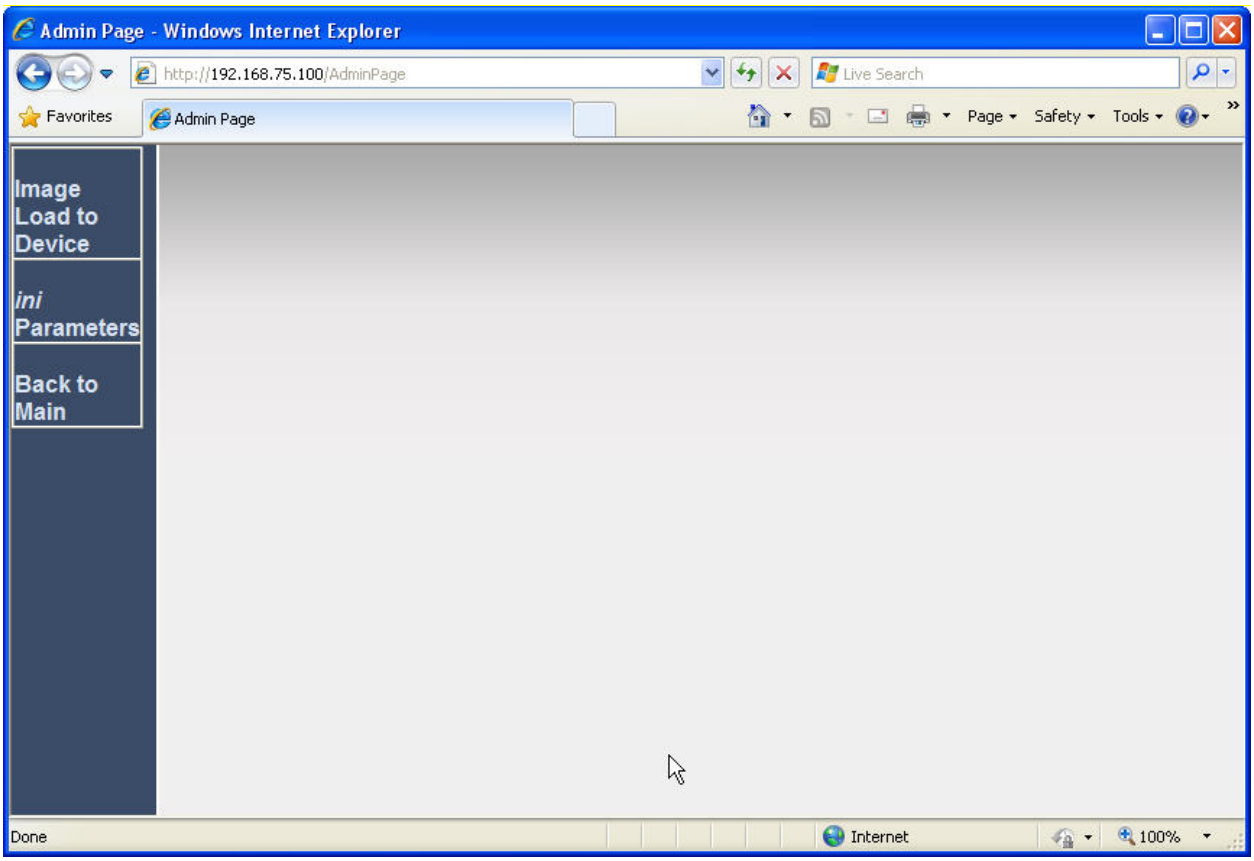
6.12. Enable Caller ID Forwarding

Navigate to **Protocol Configuration → SIP Advanced Parameters → Supplementary Services** to enable Caller ID forwarding to the branch Partner. Use the drop-down menu to select "Enable" for the **Enable Caller ID** parameter.

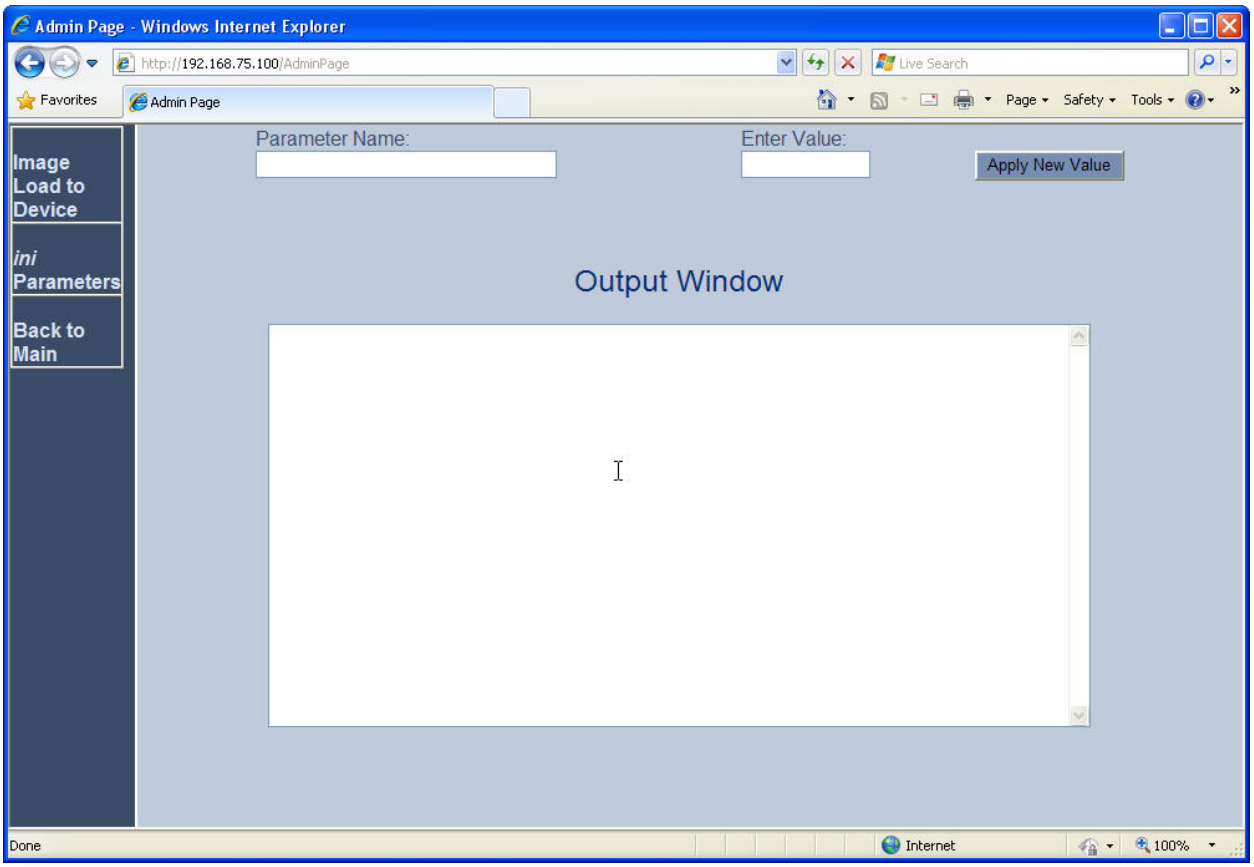


6.13. Modify .ini Configuration File

The AudioCodes MP-118 utilizes an initialization text file with a .ini extension. The .ini file contains MP-118 parameters that have been set by the WebUI, such as the parameters described in the previous sections. See **[12]** for additional information about the ini configuration file. For the AudioCodes MP-118 firmware version listed in **Table 1**, the **ReliableConnectionPersistentMode** parameter is not configurable from the WebUI and must be modified directly in the .ini file. While the .ini file can be edited directly with a text editor, it is recommended to use the .ini file editing capability of the AudioCodes Web AdminPage. The AdminPage can be accessed from a browser by entering the following URL: http://<MP-118 IP Address>/**AdminPage**. The AdminPage, similar to the one shown below, will be displayed.



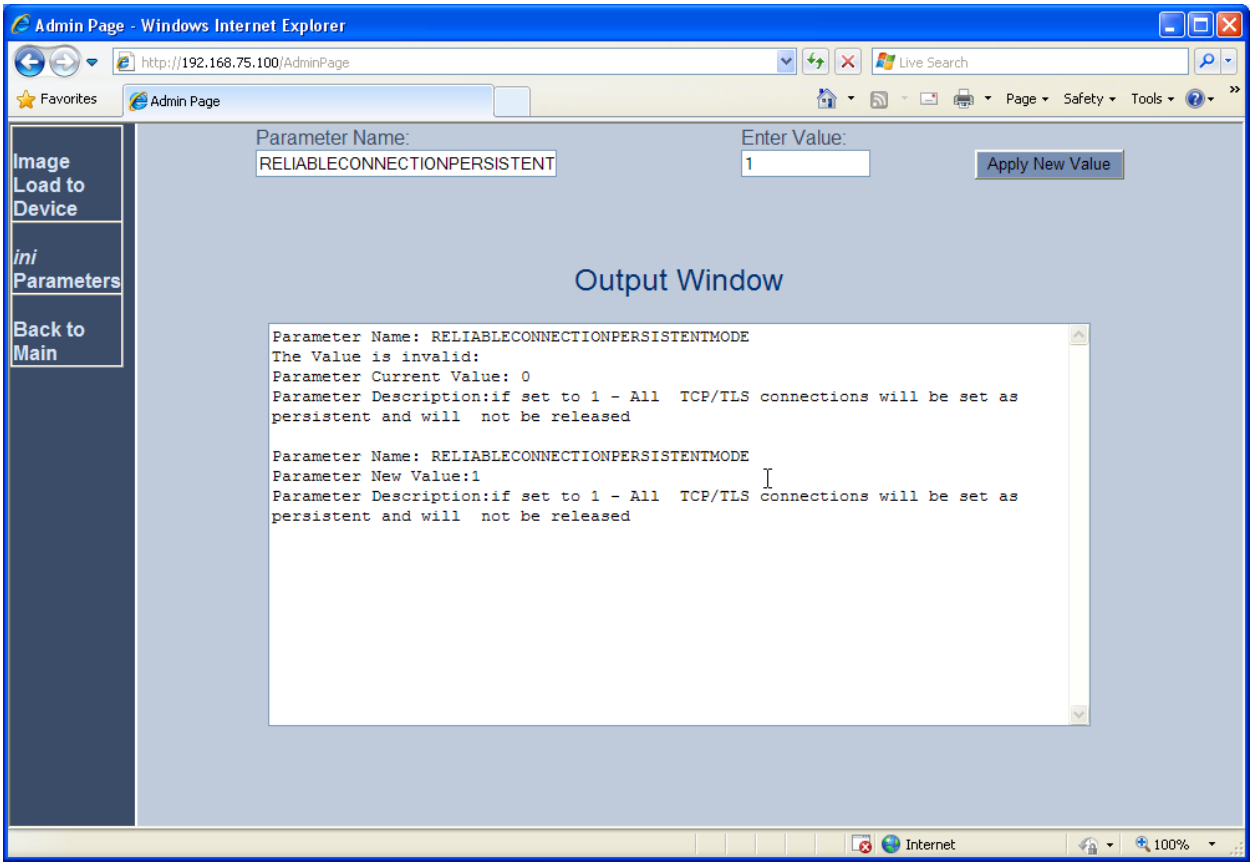
Select **ini Parameters** to access the .ini parameter editing screen. The .ini editing screen, similar to the one shown below, will be displayed.



To verify the current value of a parameter using the AdminPage, enter the parameter name in the top "Parameter Name" field and leave the "Enter Value" field blank. Click the adjacent **Apply New Value** button. The "Output Window" will display the current setting for the parameter entered in the Parameter Name field. To change the value of a parameter, enter the new parameter value in the "Enter Value" field, then click the adjacent "Apply New Value" button. The resulting screen will show both the old and new settings.

The **ReliableConnectionPersistentMode** parameter determines how the AudioCodes MP-118 establishes TCP connections. When **ReliableConnectionPersistentMode** is set to the default value of 0, all TCP connections established by the AudioCodes MP-118 are non-persistent connections. **ReliableConnectionPersistentMode** ***must be changed from the default value of 0 to a value of 1***. This configures the AudioCodes MP-118 to establish all TCP connections as persistent connections that will not be prematurely released.

The following screen shows the setting of the **ReliableConnectionPersistentMode** parameter to the value of "1" required for persistent TCP connections.

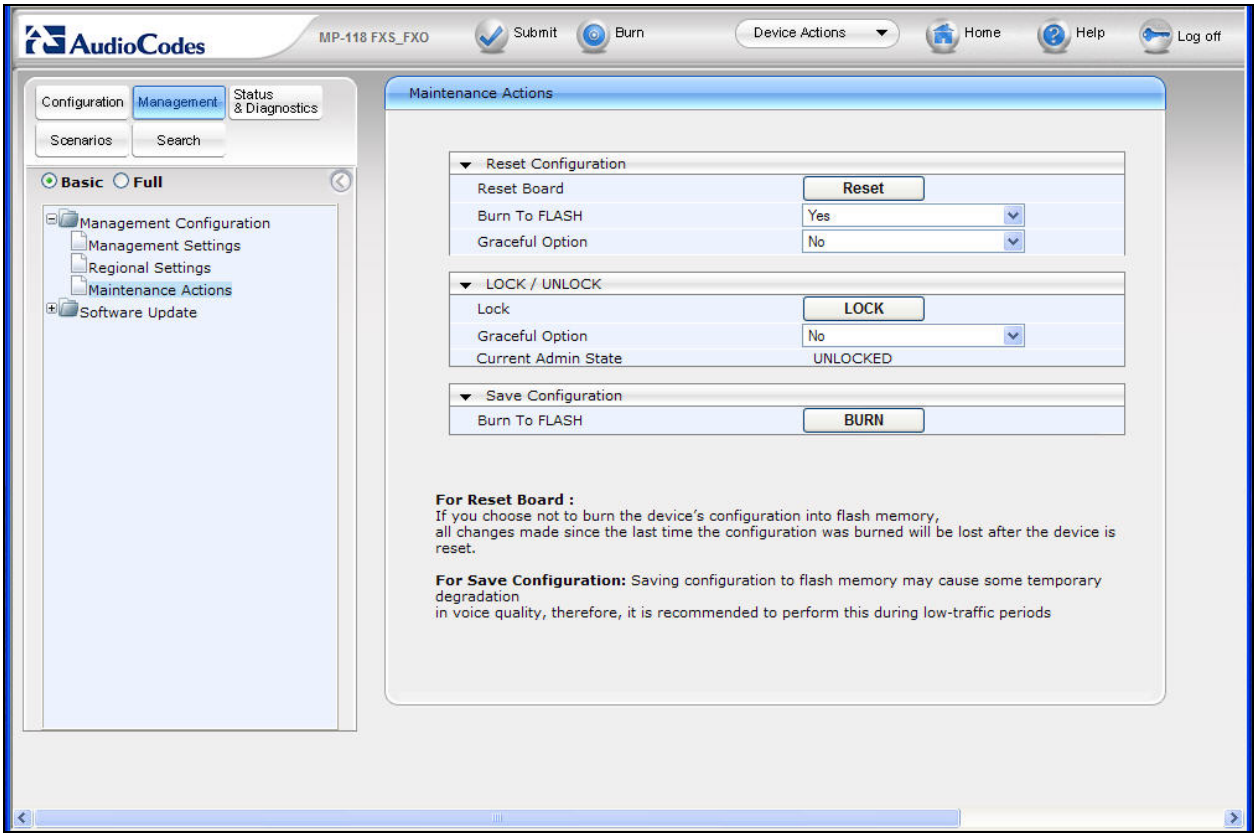


6.14. Save Configuration Changes

The 

button on the screens in the **Configuration** tab will save changes to the volatile memory (RAM) only. To save settings to non-volatile memory (flash), the 

button at the top of the screen can be used. Only configuration "burned" to non-volatile memory will be available after a hardware reset or power fail. An alternate means to access the "burn" function is via the **Management** tab. Navigate to **Management Configuration → Maintenance Actions**. The **BURN** button illustrated in the following screen may be used. The on-screen text below should be self-explanatory.



7. Configure Avaya Partner Advanced Communication System

The Partner Advanced Communication System was installed and configured to operates as in branch environment with no special configuration (in addition to the standard setup) required for supporting interoperability with the branch AudioCodes MP-118For example, the advanced feature **remote call forwarding** on the Partner Advanced Communication System was not utilized in the validation test to achieve off-switch call re-direction on unanswered incoming calls. Instead, this capability was configured on the AudioCodes MP-118. This was done to

ensure applicability of the sample configuration to other telephone key systems. Please consult **[7]** for detailed information on standard Partner ACS installation and configuration.

8. General Test Approach and Test Results

This section describes the testing used to verify the sample configuration for the AudioCodes MP-118 Media Gateway to interoperate with the Partner ACS key system in the branch and Session Manager and Communication Manger at the Headquarters location.

8.1. General Test Approach

The general test approach was to test on-net calls between Headquarters phones and Branch Partner phones through the AudioCodes MP-118 as well as off-net calls between PSTN and Branch Partner phones through the AudioCodes MP-118. Faxing between branch and Headquarters and between branch and PSTN was also tested,

8.2. Test Results

The following features and functionality were verified. Any observations related to these tests are listed at the end of this section:

- Basic calls between Headquarters/PSTN and Branch using G.711MU and G.729 codecs with codec negotiation.
- Supplementary call features (hold/unhold, attended/unattended call transfer, call conference, call forwarding, etc.).
- DTMF detection during voice calls.
- Faxing between Headquarters/PSTN and Branch
- Accessing Headquarters voice messaging system from Branch.
- Call hunting on multiple branch lines with incoming calls to same branch access number.
- Call redirection to Headquarters destination on unanswered calls at the branch.
- Call redirection to Headquarters destination if all voice lines to the branch are busy.
- On-net inter-branch calling.
- Proper system recovery after the branch AudioCodes MP-118 is shutdown/restarted or broken IP connectivity is reestablished.

The following observation was made during the testing using the sample configuration:

- **Call redirection on busy lines is not fully supported on AudioCodes MP-118**: with the current 5.80A GA version of firmware, AudioCodes MP-118 correctly redirects calls when all voice lines to the branch Partner Advanced Communication System are engaged. However, Audio-Codes MP-118 also re-directs calls when there is still one idle voice line available. For this reason the Call Admission Control is administered on the Session Manager (see **Sections 5.2** and **5.6**) in combination with the Call Forward configuration on the AudioCodes MP-118 (see **Section 6.11**) to achieve the complete call-redirect on busy lines function. The Call Admission Control on the Session Manager will allow a 4th call to be routed to the branch when all of the 3 branch voice lines are busy. This is because the 4th call could be a fax call. The capability on the branch AudioCodes MP-118 is utilized, in this situation, to forward the 4th call to the configured Headquarters destination while all 3 available voice lines are busy. This workaround

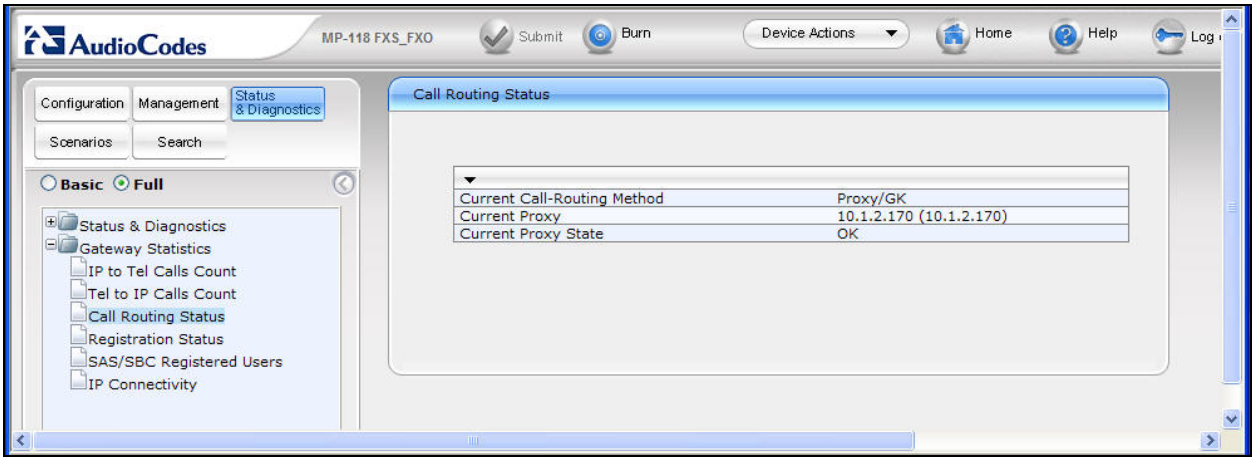
configuration on both the Session Manager and the AudioCodes MP-118 should be replaced by the fully-implemented Call Forward capability to be supported in the upcoming Release 6.0 firmware of the AudioCodes MP-11X SIP Media Gateway. Release 6.0 is in field trials at the present time.

With the above workaround, all test cases passed. The validation testing verified that the AudioCodes MP-118 SIP Media Gateway is able to interoperate successfully with the Avaya Partner Advanced Communication System 8.0 at the Branch, and Session Manager 5.2 and Communication Manager 5.2.1 at the Headquarters location.

9. Verification Steps

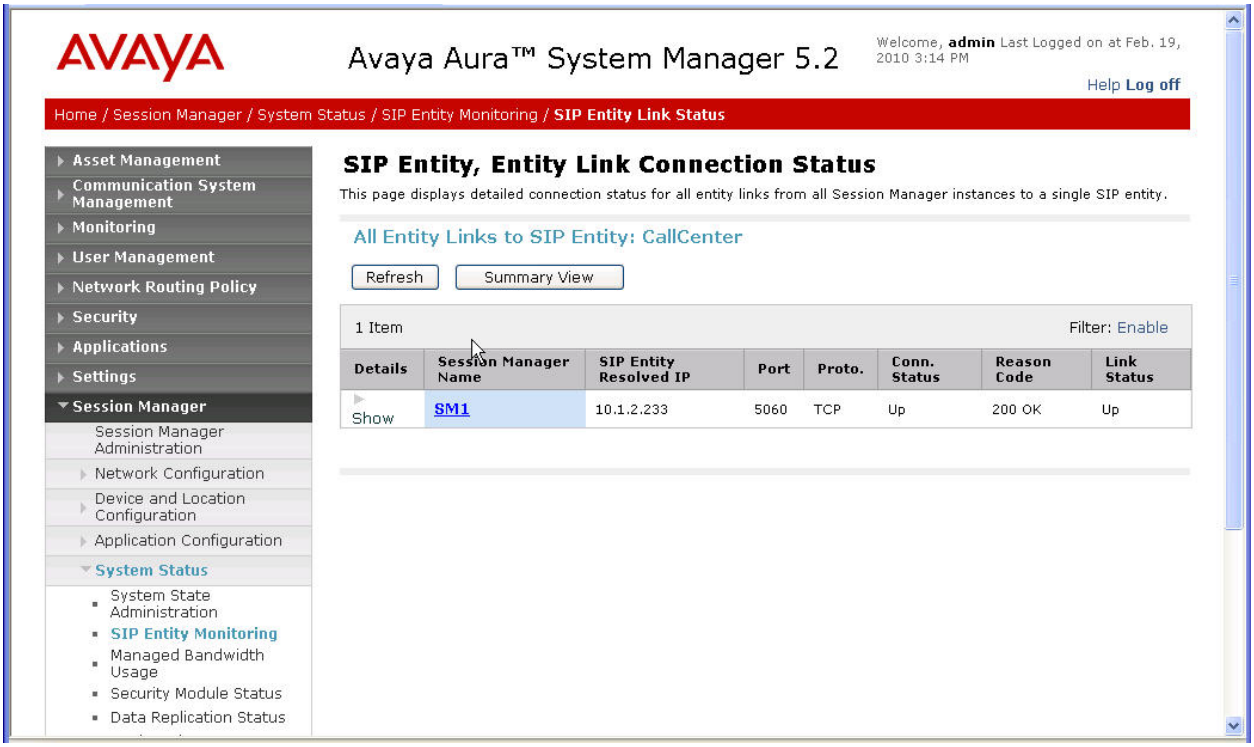
9.1. AudioCodes MP-118 Call Routing Status

From the left navigation panel, select the **Status & Diagnostics** tab, then navigate to the Call Routing Status screen by selecting **Gateway Statistics → Call Routing Status**. The **Call Routing Status** screen from the AudioCodes MP-118 indicating a good operating state is shown below:

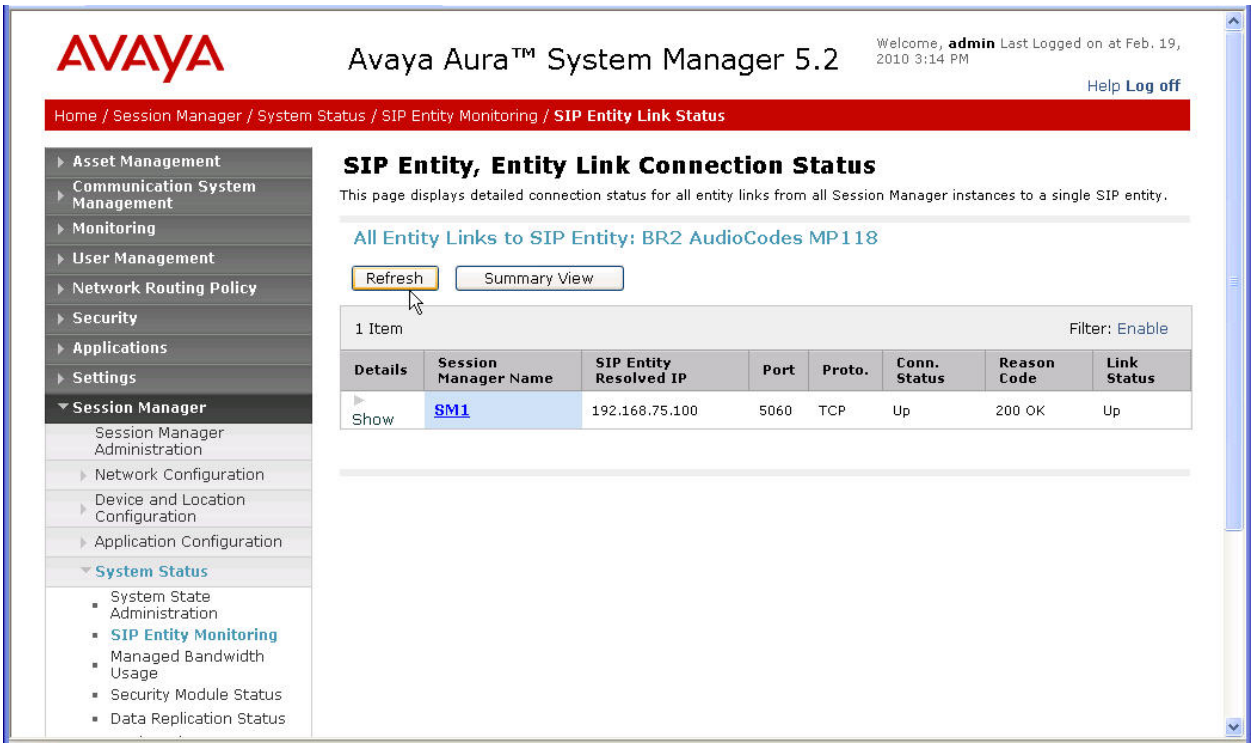


9.2. Avaya Aura™ Session Manager Entity Link Status

The following 2 screens show Session Manager Entity Link statuses on the Entity Link between Session Manager and Communication Manager and between Session Manager and the branch AudioCodes MP-118. The Entity Link status screen can be accessed from the left navigation menu **Session Manager → System Status → SIP Entity Monitoring** on System Manger. At the SIP Entity Link Monitoring Status Summary page, select the relevant SIP Entity from the All Monitored SIP Entity list. The screen below shows the Entity Link status between Session Manager and Communication Manager:



The screen below shows the Entity Link status between Session Manager and the branch AudioCodes MP-118:



9.3. Verify Basic Calls

Make calls between Headquarters and the branch; verify that the calls are successful with two-way talk path. Make calls between the PSTN and the branch through the Headquarters; verify that the calls are successful with two-way talk path.

10. Conclusion

The validation testing verified that the AudioCodes MP-118 SIP Media Gateway is able to interoperate successfully with the Avaya Partner Advanced Communication System 8.0 at the branch, and Avaya Aura™ Session Manager 5.2 and Avaya Aura™ Communication Manager 5.2.1 at the Headquarters location. These Application Notes describe the configuration steps to implement the sample configuration as presented in **Figure 1**.

11. Additional References

Avaya Aura™ Session Manager:

[1] *Avaya Aura™ Session Manager Overview*, Doc ID 03-603473, available at http://support.avaya.com.

[2] *Administering Avaya Aura™ Session Manager*, Doc ID 03-603324, available at http://support.avaya.com.

[3] *Maintaining and Troubleshooting Avaya Aura™ Session Manager*, Doc ID 03-603325, available at http://support.avaya.com.

[4] *Administering Avaya Aura™ Communication Manager as a Feature Server*, Doc ID 03-603479, available at http://support.avaya.com.

Avaya Aura™ Communication Manager 5.2:

[5] *SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers*, Doc ID 555-245-206, May, 2009, available at http://support.avaya.com.

[6] *Administering Avaya Aura™ Communication Manager*, Doc ID 03-300509, May 2009, available at http://support.avaya.com.

Avaya Partner Advanced Communication System:

[7] *Partner Advanced Communications System Installation, Programming, and Use*, Doc ID 518-456-803, April 2009, available at http://support.avaya.com.

Avaya Messaging Application:

[8] *Avaya Aura™ Communication Manager Messaging Installation and Initial Configuration*, Doc ID 03-603353, May 2009, available at http://support.avaya.com.

[9] *Modular Messaging Admin Guide Release 5.2 with Avaya MSS*, November 2009, available at http://support.avaya.com.

Avaya Application Notes:

[10] *Front-Ending Nortel Communication Server 1000 with an AudioCodes Mediant 1000 Modular Media Gateway to Support SIP Trunks to Avaya Aura™ Session Manager – Issue 1.0*, December 2009, available at http://devconnect.avaya.com.

AudioCodes MP-118:

[11] *AudioCodes SIP MP-11x & MP-124 Release Notes*, Version 5.8, Document #: LTRT-65614, October 09, available at http://www.audiocodes.com.

[12] *AudioCodes SIP MP-11x & MP-124 SIP User's Manual*, Version 5.8, Document #: LTRT-65412, October 09, available at http://www.audiocodes.com.

12. AudioCodes MP-118 Configuration .ini File

Presented below is the MP-118 BOARD.ini configuration file used in the testing to validate the sample configuration..

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

;Board: MP-118 FXS_FXO
;Serial Number: 547031
;Slot Number: 1
;Software Version: 5.80A.035.004
;DSP Software Version: 204IM => 580.06
;Board IP Address: 192.168.75.100
;Board Subnet Mask: 255.255.255.0
;Board Default Gateway: 192.168.75.1
;Ram size: 32M   Flash size: 8M
;Num of DSP Cores: 2  Num DSP Channels: 8
;Profile: NONE
;-----------------------------------------


[SYSTEM Params]

SyslogServerIP = 192.168.75.20
EnableSyslog = 1
VXMLFIleName = ''

[BSP Params]

PCMLawSelect = 3
StorageServerNetworkAddress = 255.255.255.255

[Analog Params]

PolarityReversalType = 1
MinFlashHookTime = 100
CurrentDisconnectDuration = 600

[ControlProtocols Params]

AdminStateLockControl = 0

[MGCP Params]


[MEGACO Params]

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

```
[Voice Engine Params]

CallProgressTonesFilename = 'usa_tones_13.dat'
VoiceVolume = 1
RFC2833TxPayloadType = 101
RFC2833RxPayloadType = 101
DTMFDetectorSensitivity = 1

[WEB Params]

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

[SIP Params]

ENABLECALLERID = 1
MAXDIGITS = 15
REGISTRATIONTIME = 3600
ISPROXYUSED = 1
ISTWOSTAGEDIAL = 0
ROUTEMODEIP2TEL = 1
ENABLECURRENTDISCONNECT = 1
ENABLEREVERSALPOLARITY = 1
GWDEBUGLEVEL = 5
ENABLEEARLYMEDIA = 1
ISUSERPHONE = 0
SIPGATEWAYNAME = 'avaya.com'
CNONCE = '0a123bcf'
PASSWORD = '787899'
ALWAYSSENDTOPROXY = 1
ISFAXUSED = 1
SIPTRANSPORTTYPE = 1
DISCONNECTONDIALTONE = 1
PREFIX2EXTLINE = '9'
RELIABLECONNECTIONPERSISTENTMODE = 1

[IPsec Params]


[SNMP Params]

DisableSNMP = 0
SNMPTrapManagerHostName = ''

;
;  *** 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 0 = 10, 10.1.10.10, *, 0, 255, 0, -1, , -1, , -1, -1;
PREFIX 1 = *, 10.1.10.11, *, 0, 255, 0, -1, , -1, , -1, -1;

[ \PREFIX ]

;
;  *** TABLE CoderName ***
;
;

[ CoderName ]
FORMAT CoderName_Index = CoderName_Type, CoderName_PacketInterval,
CoderName_rate, CoderName_PayloadType, CoderName_Sce;
CoderName 0 = g7231, 30, 0, 255, 0;
CoderName 1 = g711Alaw64k, 20, 0, 255, 0;
CoderName 2 = g711Ulaw64k, 20, 0, 255, 0;

[ \CoderName ]

;
;  *** TABLE TrunkGroup ***
;
;

[ TrunkGroup ]
FORMAT TrunkGroup_Index = TrunkGroup_TrunkGroupNum, TrunkGroup_FirstTrunkId,
TrunkGroup_FirstBChannel, TrunkGroup_LastBChannel,
TrunkGroup_FirstPhoneNumber, TrunkGroup_ProfileId, TrunkGroup_LastTrunkId,
TrunkGroup_Module;
TrunkGroup 0 = 1, 255, 1, 1, 9085434000, 0, 255, 255;
TrunkGroup 1 = 1, 255, 2, 2, 9085434000, 0, 255, 255;
TrunkGroup 2 = 1, 255, 3, 3, 9085434000, 0, 255, 255;
TrunkGroup 3 = 2, 255, 4, 4, 9085434009, 0, 255, 255;

[ \TrunkGroup ]

;
;  *** 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 = 1908543400, *, *, 255, 255, 1, 0, 255, , , 255, -1, -1;
NumberMapIp2Tel 2 = 19085434009, *, *, 255, 255, 1, 0, 255, , , 255, -1, -1;
```

```
[ \NumberMapIp2Tel ]

;
;  *** TABLE PstnPrefix ***
;
;

[ PstnPrefix ]
FORMAT PstnPrefix_Index = PstnPrefix_DestPrefix, PstnPrefix_TrunkGroupId,
PstnPrefix_SourcePrefix, PstnPrefix_SourceAddress, PstnPrefix_ProfileId,
PstnPrefix_SrcIPGroupID, PstnPrefix_DestHostPrefix, PstnPrefix_SrcHostPrefix;
PstnPrefix 0 = 9085434009, 2, *, *, 0, -1, , ;
PstnPrefix 1 = 9, 1, *, *, 0, -1, , ;

[ \PstnPrefix ]

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

[ ProxyIp ]
FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType,
ProxyIp_ProxySetId;
ProxyIp 0 = 10.1.2.170, 1, 0;

[ \ProxyIp ]

;
;  *** TABLE TxDtmfOption ***
;
;

[ TxDtmfOption ]
FORMAT TxDtmfOption_Index = TxDtmfOption_Type;
TxDtmfOption 0 = 4;

[ \TxDtmfOption ]

;
;  *** TABLE TrunkGroupSettings ***
;
;

[ TrunkGroupSettings ]
FORMAT TrunkGroupSettings_Index = TrunkGroupSettings_TrunkGroupId,
TrunkGroupSettings_ChannelSelectMode, TrunkGroupSettings_RegistrationMode,
TrunkGroupSettings_GatewayName, TrunkGroupSettings_ContactUser,
TrunkGroupSettings_ServingIPGroup;
TrunkGroupSettings 0 = 1, 5, 255, , , -1;
TrunkGroupSettings 1 = 2, 0, 255, , , -1;

[ \TrunkGroupSettings ]

;
```

```
;  *** TABLE FwdInfo ***
;
;

[ FwdInfo ]
FORMAT FwdInfo_Index = FwdInfo_Type, FwdInfo_Destination,
FwdInfo_NoReplyTime;
FwdInfo 0 = 4, 19081233000, 15;
FwdInfo 1 = 4, 19081233000, 15;
FwdInfo 2 = 4, 19081233000, 15;

[ \FwdInfo ]

;
;  *** TABLE CallWaitingPerPort ***
;
;

[ CallWaitingPerPort ]
FORMAT CallWaitingPerPort_Index = CallWaitingPerPort_IsEnabled;
CallWaitingPerPort 0 = 1;
CallWaitingPerPort 1 = 1;
CallWaitingPerPort 2 = 1;

[ \CallWaitingPerPort ]

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

[ ProxySet ]
FORMAT ProxySet_Index = ProxySet_EnableProxyKeepAlive,
ProxySet_ProxyKeepAliveTime, ProxySet_ProxyLoadBalancingMethod,
ProxySet_IsProxyHotSwap, ProxySet_SRD, ProxySet_ClassificationInput;
ProxySet 0 = 0, 60, 0, 0, 0, 0;

[ \ProxySet ]
```

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