Avaya Solution & Interoperability Test Lab

Avaya Aura™ Session Manager Survivable SIP Gateway Solution using AudioCodes MP-118 in a Distributed Trunking Configuration – Issue 1.2

Abstract

These Application Notes present a sample configuration of the Avaya Aura™ Session Manager Survivable SIP Gateway Solution using the AudioCodes MP-118 SIP Media Gateway in a Distributed Trunking configuration.

This solution addresses the risk of service disruption for SIP endpoints deployed at remote branch locations if connectivity to the centralized Avaya SIP call control platform (Avaya Aura™ Session Manager) located at the main site is lost. Connectivity loss can be caused by WAN access problems being experienced at the branch or by network problems at the centralized site blocking access to the Avaya SIP call control platform, or by Avaya Aura™ Session Manager going out of service.

The Avaya Aura™ Session Manager Survivable SIP Gateway Solution monitors the connectivity health from the remote branch to the centralized Avaya SIP call control platform. When connectivity loss is detected, Avaya one-X Deskphone SIP 9600 Series IP Telephones as well as the AudioCodes SIP Media Gateway dynamically switch to survivability mode, restoring telephony services to the branch for intra-branch and PSTN calling.

Testing was conducted via the DevConnect Program 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 of the Avaya Aura™ Session Manager Survivable SIP Gateway Solution using the AudioCodes MP-118 Media Gateway in a Distributed Trunking scenario.

SIP endpoints deployed at remote branch locations risk a loss of service if a break in connectivity to the centralized SIP call control platform (Session Manager) occurs. Connectivity loss can be caused by WAN access problems being experienced at the branch or by network problems at the centralized site blocking access to the Avaya SIP call control platform, or by Session Manager going out of service. The survivable SIP gateway solution monitors connectivity health from the remote branch to the centralized Avaya SIP call control platform. When connectivity loss is detected, SIP endpoints and SIP gateway components within the branch dynamically switch to survivability mode restoring basic telephony services to the branch for intra-branch and PSTN calling. When connectivity from the branch to the centralized Avaya SIP call control platform is restored, SIP components dynamically switch back to normal operation.

The primary components of this solution are the Avaya one-X Deskphone SIP 9600 Series IP Telephones and the AudioCodes SIP Media Gateways models MP-114 or MP-118 as well as Session Manager 5.2 which provides the centralized SIP control platform with SIP registrar and proxy functions. The sample configuration presented in these Application Notes utilizes the AudioCodes SIP Media Gateway model MP-118. These configuration steps can also be applied to the AudioCodes SIP Media Gateway model MP-114 using the AudioCodes firmware version specified in **Section 3**.

1.1. Interoperability Testing

The interoperability testing focused on the dynamic switch from the Normal Mode (where the network connectivity between the main site and the branch site is intact) to the Survivable Mode (where the network connectivity between the main site and the branch site is broken) and vice versa. The testing also verified interoperability between the Avaya 9600 Series SIP Phones and the AudioCodes SIP Media Gateway in the Survivable Mode.

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

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 Session Manager. Starting with release 5.2, Session Manager also includes onboard SIP Registrar and Proxy functionality for SIP call control. In the test configuration, all Avaya 9600 Series SIP Phones, either at the main site or at the branch sites, register to the Session Manager (the branch phones will failover to register with the AudioCodes MP-118 in Survivable Mode[1]) with calling features supported by Communication Manager, which serves as a Feature Server within the Session Manager architecture. The Avaya 9600 Series SIP Phones are configured on

[1] The main site phones still register to Session Manager in the case of broken connectivity between the main site and the branch. In the case of Session Manager going out of service, the main site phones will cease to function.

Communication Manger as Off-PBX-Stations (OPS) and acquire advanced call features from Communication Manger.

1.1.2. AudioCodes SIP Media Gateway

The AudioCodes SIP Media Gateway, referred to as AudioCodes MP-118 throughout the remainder of this document, takes on various roles based on call flows and network conditions. The following lists these roles:

- SIP PSTN Media Gateway (FXO interfaces to PSTN)
- SIP Analog Terminal Adapter (FXS interfaces to analog endpoints)
- SIP Registrar and Proxy (dynamically activated on detection of lost connectivity to the centralized SIP control platform)

Note: AudioCodes labels the Survivable SIP Registrar and Proxy functionality of the MP-118 as Stand-Alone Survivability (SAS). SAS will be used throughout these Application Notes.

1.1.3. Avaya one-X Deskphone SIP 9600 Series IP Telephone

The Avaya one-X Deskphone SIP 9600 Series IP Telephone, referred to as Avaya 9600 SIP Phone throughout the remainder of this document, is a key component of the survivable SIP gateway solution. The 2.5.5.11 firmware release of the Avaya 9600 SIP Phone tested with the sample configuration includes feature capabilities specific to SIP survivability, enabling the phone to monitor connectivity to Session Manager and dynamically failover to the local AudioCodes MP-118 as an alternate or survivable SIP server. See **Section 11** [7] for additional information on the Avaya 9600 SIP Phone.

1.1.4. Network Modes

Normal Mode: Branch has WAN connectivity to the main Headquarters/Datacenter location and the centralized Avaya SIP call control platform is being used for all branch calls.

Survivable Mode: A Branch has lost WAN connectivity to the Headquarters/Datacenter location. The local branch AudioCodes MP-118 SIP gateway with SAS capability is being used for all calls at that branch. Note that if the Session Manager which provides the centralized SIP control loses connectivity to the WAN, all branches will go into survivable mode simultaneously.

1.1.5. PSTN Trunking Configurations

The Session Manager Survivable SIP Gateway Solution can interface with the PSTN in either a Centralized Trunking or a Distributed Trunking configuration. These trunking options determine how branch calls to and from the PSTN will be routed over the corporate network.

Assuming an enterprise consisting of a main Headquarters/Datacenter location and multiple distributed branch locations all inter-connected over a corporate WAN, the following defines Centralized Trunking and Distributed Trunking as related to this survivable SIP gateway solution:

Centralized Trunking: In Normal Mode, all PSTN calls, inbound to the enterprise and outbound from the enterprise, are routed to/from the PSTN connection as configured on the Avaya Media Gateway centrally located at the Headquarters/Datacenter location. In Survivable Mode, the PSTN calls to/from the branch phones are through the analog trunks from the Service Provider connected to the FXO interface ports on the local AudioCodes MP-118 branch gateway.

Distributed Trunking: Outgoing PSTN call routing can be determined by the originating source location using Communication Manager Location Based Routing. Local calls from branch locations can be routed back to the same branch location and terminate on the FXO interface of the local AudioCodes MP-118 branch gateway (see **Section 1.1.6** for call flow details). This has the potential benefits of saving bandwidth on the branch access network, off-loading the WAN and centralized media gateway resources, avoiding Toll Charges, and reducing latency.

The sample configuration presented in these Application Notes implements a Distributed Trunking configuration. The sample configuration of the Session Manager Survivable SIP Gateway Solution in a Centralized Trunking configuration is described in a separate Application Notes document.

1.1.6. Sample Call Flow: Branch PSTN Outbound Local – Normal Mode

Some of the Communication Manager and Session Manager configuration steps presented in **Section 4** and **Section 5** are to support the source based routing requirements of the Branch PSTN Outbound Local – Normal Mode call flow. The details of this call flow, specific to the sample configuration, are included here as a reference for better understanding the linkage of the various configuration steps.

Branch PSTN Outbound Local – Normal Mode:

Branch 2 Avaya 9600 SIP Phone user dials the local PSTN number: 9 1-908-555-1111.

1. Branch 2 Avaya 9600 SIP Phone sends SIP INVITE to Session Manager with dialed digit string of 919085551111.
2. Session Manager receives the SIP INVITE and identifies the Avaya 9600 SIP Phone user has an assigned Communication Manager Extension. Session Manager forwards the SIP INVITE to Communication Manager.
3. Communication Manger receives the SIP INVITE from Session Manager on SIP Trunk Group Number 42.
4. Communication Manager identifies the IP address of the Avaya 9600 SIP Phone in the Contact field of the SIP INVITE message as an IP address mapped to IP Network Region 12 which is configured to Location 12. Communication Manager now knows the source of the call is Location 12.
5. The leading 9 in the dialed digit string is identified by Communication Manager as the ARS Access Code. The 9 is removed from the dialed digit string.
6. The ARS Digit Analysis Table for Location 12 is queried for a match on the remaining digits 19085551111.

7. A match on 1908 is found and Route Pattern 12 is chosen as specified in the ARS Digit Analysis Table.
8. Route Pattern 12 routes the call to SIP Trunk Group Number 32 which connects Communication Manager to Session Manager and is specifically configured for routing local PSTN calls from Branch 2 phones.
9. Communication Manager sends a new SIP INVITE to Session Manager over SIP Trunk Group Number 32 with the dialed digits of 19085551111.
10. Session Manager finds a configured Dial Pattern that matches the dialed number 19085551111 with associated Routing Policy that routes the call to the Branch 2 Audio Codes MP-118 media gateway with IP address 192.168.75.100 using TCP port 5070.
11. Session Manager forwards the SIP INVITE with dialed digits string 19085551111 to the Branch 2 AudioCodes MP-118.
12. The Branch 2 AudioCodes MP-118 internally routes the call to an FXO interface for termination on the PSTN.

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 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 shown in **Figure 1** is modeled after an enterprise consisting of a main Headquarters/Datacenter location and multiple distributed branch locations all inter-connected over a corporate WAN. While three branch locations have been included in the sample network, Branch 2 configurations are highlighted and documented in ensuing sections of these Application Notes.

The Headquarters location hosts a Session Manager (with its companion System Manager) providing enterprise-wide SIP call control, and a Communication Manager as a Feature Server providing advanced feature capabilities to Avaya 9600 SIP Phones. The Communication Manager runs inside an Avaya G-Series Media Gateway with PSTN trunks. The Avaya Aura™ Communication Manager Messaging is running co-resident with the Communication Manager to provide Voice Mail functionality[2] (Avaya Modular Messaging is also configured and tested in the sample configuration). The Headquarters location also hosts an Avaya IP Phone Configuration File Server for Avaya 9600 SIP Phones to download configuration information. The Session Manager is connected to the 10.1.2.0/24 subnet; the Communication Manager and the phone configuration file server are connected to the 10.32.2.0/24 subnet; the Avaya 9600 SIP Phones are connected to the 10.32.1.0/24 subnet.

The configuration details of the phone configuration file server, the Communication Manager Messaging application as well as Avaya Modular Messaging are considered out of scope of these Application Notes and therefore not included.

The Avaya IP Phone Configuration File Server contains the 46xxsettings.txt file used by Avaya IP phones to set the values of phone configuration parameters. **Section 6** includes the parameters of the 46xxsettings.txt file used by the Avaya 9600 SIP Phone for survivability. The Communication Manager Messaging (or Avaya Modular Messaging) can be reached by dialing the internal extension configured as the voice mail access number, or by dialing a PSTN number that also terminates to the voice messaging application. The internal extension is configured in the 46xxsettings.txt file as the default voice mail access number to dial when the Message button of the Avaya 9600 SIP Phone is pressed while the phone is in Normal Mode. The external PSTN number is configured in the 46xxsettings.txt file as an alternate voice mail access number to dial when the Message button of the Avaya 9600 SIP Phone is pressed while the branch phone is in Survivable Mode. This enables branch users to continue to access the centralized voice mail platform while in Survivable Mode.

The branch locations consist of two Avaya 9600 SIP Phones, an AudioCodes MP-118 SIP Media Gateway with a PSTN Analog trunk on the FXO interface and two analog phones on the FXS interfaces. A flat network has been implemented at each branch.

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[2] The voice messaging system is used in the test configuration to test voice mail access and MWI (Messaging Wait Indicator) on Avaya 9600 SIP Phones in both Normal Mode and Survivable Mode. Any compatible messaging system can be used to satisfy this test purpose, e.g., Avaya Modular Messaging can be used in the test configuration instead of Communication Manager Messaging.

Note that the Communication Manger serves as a Feature Server in the test configuration. As such, it does not support inter-working between SIP phones and non-SIP phones (H.323 and other Avaya digital and/or analog telephone sets) directly configured on the same Communication Manager[3]. This restriction will be lifted in future releases of Session Manager and Communication Manager. In the sample configuration, all phones at both the main and branch sites are SIP phones (branch analog sets are adapted by the AudioCodes MP-118 as SIP phones too).

The Distributed Trunking capabilities of the solution utilize the source based call routing feature of Communication Manager which requires the information presented in **Table 1**. The branch configurations presented throughout these Application Notes focus on Branch 2; however, Branch 1 and Branch 3 parameters are included on relevant screen shots.

IP Network	IP Network Region	Location	Area Code	AudioCodes MP-118 IP Address
10.32.1.0/24 10.32.2.0/24 10.1.2.0/24	1	1 (Headquarters)	201	
191.168.75.0/24	11	11 (Branch 1)	609	191.168.75.100
192.168.75.0/24	12	12 (Branch 2)	908	192.168.75.100
193.168.75.0/24	13	13 (Branch 3)	732	193.168.75.100

Table 1 – Network Information

[3] See **Section 11** [10] for application notes on configuring Communication Manager as an Access Element to support H.323 and digital telephones.

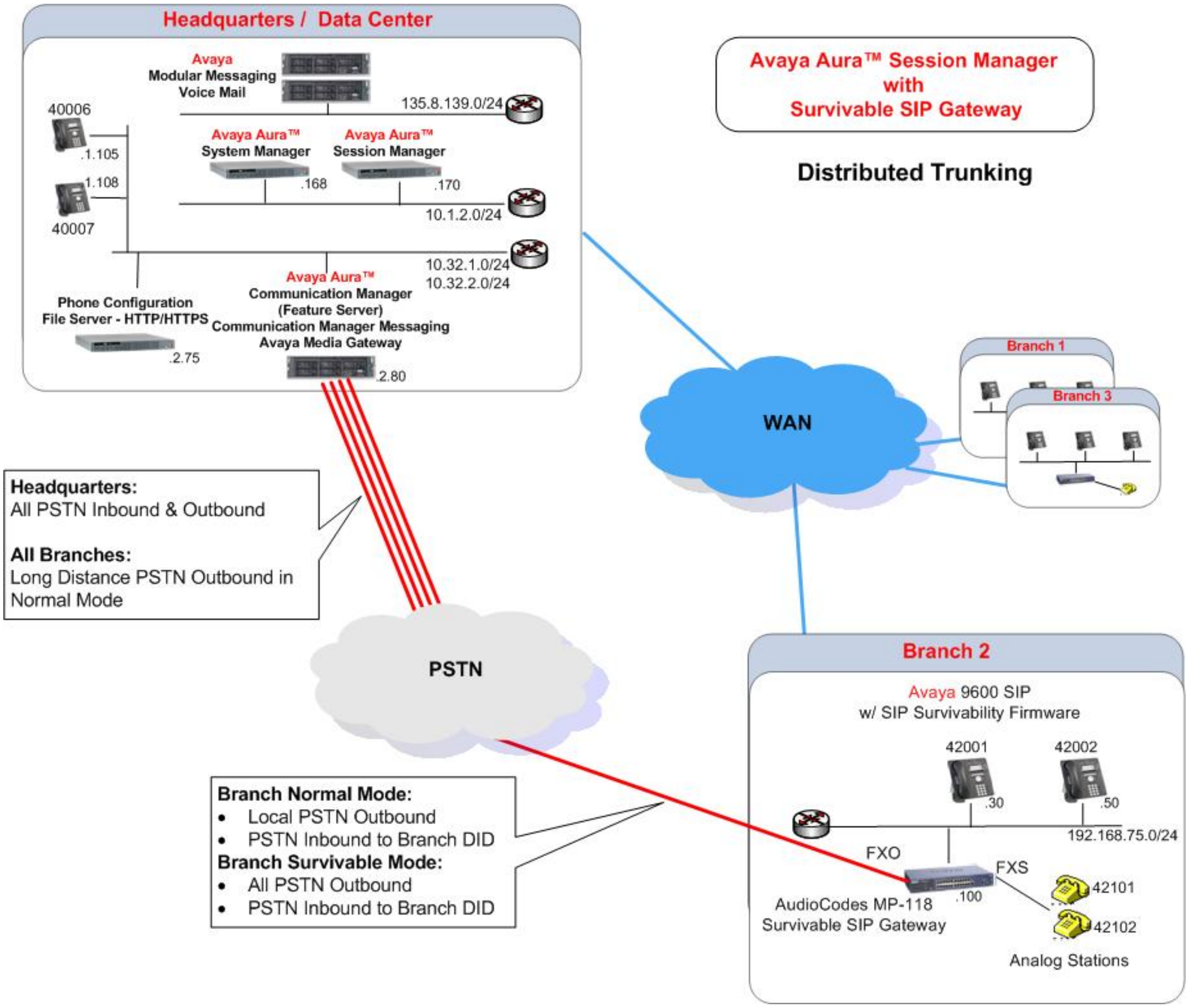


Figure 1 – Network Diagram

3. Equipment and Software Validated

The following components were used for the sample configuration:

Component	Software/Firmware
Avaya Aura™ Session Manager	R5.2.0.1.520017
Avaya Aura™ System Manager	R5.2.0.1.520017
Avaya Aura™ Communication Manager (Feature Server)	5.2.1 (R015x.02.1.016.4)
Avaya Aura™ Communication Manager Messaging	Release 5.2
Avaya Modular Messaging	V5.2 with Patch 8 (9.2.15013)
Avaya 9600 Series IP Telephones Models: 9620 and 9630	Avaya one-X™ Deskphone Edition SIP 2.5.0
Avaya 6210 Analog Telephone	-
HTTPS/HTTP Phone Configuration File Server	Windows Server 2003 SP2
AudioCodes MP-118 FXS-FXO [4]	5.80A.019.003

Table 3 – Software/Hardware Version Information

[4] Although not tested, the AudioCodes MP-114 gateway can be used in the sample configuration presented in these Application Notes. The MP112 was not specifically tested. However for the functions it can perform, Avaya will support it in place of the MP-118 shown and tested in this document because the MP112 software is the same as MP-118. Please note the MP-112 has no FXO interfaces so this function is not supported on the MP-112.

4. Configure Communication Manager

This section shows the necessary steps to configure Communication Manager to support the survivable SIP gateway solution in a Distributed Trunking scenario. It is assumed that the basic configuration on Communication Manager, the required licensing, the configuration for connection to PSTN through the T1/E1 interface as well as the configuration required for accessing Communication Manager Messaging (if it is used for voice messaging), has already been administered. See listed documents in **Section 11** for additional information.

All commands discussed in this section are executed on Avaya 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.

- Communication Manager license
- System parameters features
- IP node names
- IP codec set
- Locations
- IP network regions
- Stations
- SIP signaling group and trunk group
- Route pattern
- Private numbering
- Automatic Alternate Routing (AAR)
- Automatic Route Selection (ARS)

4.1. Verify Communication Manger License

Log into 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  11
                              OPTIONAL FEATURES

IP PORT CAPACITIES                                          USED
                     Maximum Administered H.323 Trunks: 800   100
           Maximum Concurrently Registered IP Stations: 18000 1
             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   252
  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
```

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.

```
display 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: none
             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

   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

4.3. Configure IP Node Names

Use the "change node-names ip" command to add an entry for the Session Manager that the Communication Manager will connect to. The **Name** "sm1" and **IP Address** "10.1.2.170" are entered for the Session Manager Security Module (SM-100) interface. The configured node-name "sm1" will be used later on in the SIP Signaling Group administration (**Section 4.8.1**).

```
change node-names ip                                               Page   1 of   2
                                  IP NODE NAMES
    Name                IP Address
default               0.0.0.0
msgserver             10.32.2.90
procr                 10.32.2.80
sm1                   10.1.2.170
```

4.4. Configure IP Codec Set

Configure the IP codec set to use for SIP 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.

```
display 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:


     Media Encryption
 1: none
 2:
 3:
```

4.5. Locations

The locations of each branch as well as Headquarters must be defined within Communication Manager using the "change locations" command. The values used in the sample configuration are shown below. The location numbers and names are entered as defined in **Table 1**. All remaining fields have been left at default values. The **Timezone Offset** can be used if locations reside within different time zones. All locations are within the same time zone in the sample configuration so the default value of 00:00 is used.

```
change locations                                              Page   1 of   4
                                   LOCATIONS

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

 Loc  Name           Timezone Rule  NPA  ARS  Atd      Disp   Prefix   Proxy Sel
 No                   Offset             FAC  FAC      Parm              Rte Pat
 1:  Headquarters    + 00:00   0                        1
 2:                      :
 3:                      :
 4:                      :
 5:                      :
 6:                      :
 7:                      :
 8:                      :
 9:                      :
10:                      :
11:  Branch 1        + 00:00   0                        1
12:  Branch 2        + 00:00   0                        1
13:  Branch 3        + 00:00   0                        1
14:                      :
```

4.6. Configure IP Network Regions

An IP address map can be used for network region assignment. The following screen illustrates a subset of the IP network map used to verify this sample configuration. Branch 2 has IP Addresses in 192.168.75.0/24 assigned to network region 12. The Headquarters location has IP Addresses in 10.32.1.0/24 (for phones), 10.32.2.0/24 (for servers) and 10.1.2.0/24 (where Session Manager is assigned) configured to network region 1. Although not illustrated in these Application Notes, network region assignment can be used to vary behaviors within and between regions.

```
display ip-network-map                                        Page   1 of  63
                               IP ADDRESS MAPPING

                                                Subnet Network     Emergency
 IP Address                                     Bits   Region VLAN Location Ext
 ---------------------------------------------- ------ ------ ---- -------------
 FROM: 10.1.2.0                                 /24    1      n
   TO: 10.1.2.255
 FROM: 10.32.1.0                                /24    1      n
   TO: 10.32.1.255
 FROM: 10.32.2.0                                /24    1      n
   TO: 10.32.2.255
 FROM: 192.168.75.0                             /24    12     n
   TO: 192.168.75.255
```

Although not unique to the AudioCodes equipped branch, the following screens illustrate relevant aspects of the network region used to verify this sample configuration. The IP Network **Region** is mapped to the **Location** previously created in **Section 4.5**. The values used in the sample configuration for Branch 2 IP Network Region 12 are shown below. The **Authoritative Domain** "avaya.com" matches the SIP domain configured in the Session Manager (**Section 5.1**) as well as the AudioCodes gateway (**Section 7.3**). The **Codec Set** for intra-region calls is set to the codec set 1 as configured in **Section 4.4**. The **IP-IP Direct Audio** parameters retain the default "yes" allowing direct IP media paths both within the region and between regions to minimize the use of media resources in the Media Gateway.

```
display ip-network-region 12                                        Page   1 of  19
                               IP NETWORK REGION
  Region: 12
Location: 12       Authoritative Domain: avaya.com
    Name: Branch 2
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: 3329
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
```

The following screen illustrates a portion of **Page 3** for network region 12. The connectivity between network regions is specified under the **Inter Network Region Connection Management** heading, beginning on **Page 3.** Codec set 1 is specified for connections between network region 12 and network region 1.

```
display ip-network-region 12                                   Page   3 of  19

 Source Region: 12    Inter Network Region Connection Management     I       M
                                                                     G  A    e
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    a
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    s
 1    1     y    NoLimit                                             n all
 2
 3
 4
 5
 6
 7
 8
 9
 10
 11
 12   1                                                                all
 13
 14
 15
```

The ip-network-region form for Network Region 1 is similarly configured (not shown). Network region 1 is for phones and servers as well as Session Manager at the Headquarters location as defined in **Table 1**.

4.7. Add Stations

A station must be created on Communication Manager for each SIP User account to be created in Session Manager which includes a provisioned Communication Manager Extension. The extension assigned to the Communication Manager station must match the Communication Manager Extension assignment in Session Manager (see **Section 5.10**).

Use the “add station” command to add a station to Communication Manager. The “add station” command for an Avaya 9620 SIP Phone located at Branch 2 assigned to extension 42001 is shown below. Because this is a SIP station, only the **Type** and **Name** fields are required to be populated as highlighted in bold. All remaining fields can be left at default values. Of course, feature programming will vary.

```
add station 42001                                            Page   1 of   6
                                  STATION

Extension: 42001                        Lock Messages? n              BCC: 0
     Type: 9620SIP                      Security Code:                 TN: 1
     Port:                            Coverage Path 1: 1              COR: 1
     Name: AC-Surv-BR21-LD            Coverage Path 2:                COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 19
                                                Message Lamp Ext: 42001

        Display Language: english

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

On **Page 6** of the station form, specify “aar” for **SIP Trunk**.

```
add station 42001                                            Page   6 of   6
                                  STATION
SIP FEATURE OPTIONS
         Type of 3PCC Enabled: None
                    SIP Trunk: aar
```

Repeat the above procedures for adding each and every SIP phone located at both the main site and the branch sites including the branch analog stations. Note that a phone type of “9600SIP” should be used for the branch analog stations.

After all the stations have been added, use the “list off-pbx-telephone station-mapping” command to verify that all the stations have been automatically designated as OPS (Off-PBX Station) sets. In the screen shown below, extensions 40006 and 40007 are SIP phones at the main site; extensions 42001 and 42002 are SIP phones at Branch 2; extensions 42101 and 42102 are analog phones at Branch 2.

```
list off-pbx-telephone station-mapping

                    STATION TO OFF-PBX TELEPHONE MAPPING

Station         Appl   CC   Phone Number     Config Trunk   Mapping      Calls
Extension                                    Set    Select  Mode         Allowed

40006           OPS         40006            1  /   aar     both         all
40007           OPS         40007            1  /   aar     both         all
42001           OPS         42001            1  /   aar     both         all
42002           OPS         42002            1  /   aar     both         all
42101           OPS         42101            1  /   aar     both         all
42102           OPS         42102            1  /   aar     both         all
```

4.8. Configure SIP Signaling Group and Trunk Group

Two SIP signaling groups and two associated trunk groups are used between Communication Manager and Session Manager in the sample configuration. The "Primary" SIP trunk group (and the associated signaling group) is used for regular call signaling and media transport to/from SIP phones registered to Session Manager including phones at all branches (when in Normal Mode); the "Secondary" SIP trunk group (and the associated signaling group) is used for routing calls from branch phones to local (non-toll) PSTN destinations in Normal Mode (see **Section 1.1.6** for call flow details).

Note that a single trunk group (the "Primary" trunk group) can be used for both purposes and it is not required to configure two separate trunk groups. However, the use of two trunk groups provides the added flexibility to change trunk parameters independently. Tracing call legs within Communication Manager is also simplified.

4.8.1. SIP Signaling Groups

In the sample configuration, Communication Manager acts as a Feature Server supporting the Avaya 9600 SIP Phones. An IMS-enabled SIP trunk to Session Manager is required for this purpose. Use the "add signaling-group n" command, 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**: "tls"
- **IMS Enabled?**: "y"
- **Near-end Node Name**: "procr" node name from **Section 4.3**
- **Far-end Node Name**: "sm1" Session Manager node name from **Section 4.3**
- **Near-end Listen Port**: "5061"
- **Far-end Listen Port**: "5061"
- **Far-end Network Region**: Network region number "1" from **Section 4.6**
- **Far-end Domain**: SIP domain name from **Section 4.5** and **Section 5.1**
- **DTMF over IP**: "rtp-payload"

The screen below shows signaling group 42 which is used in the sample configuration as the "Primary" signaling group.

```
add signaling-group 42
                                SIGNALING GROUP

 Group Number: 42              Group Type: sip
                         Transport Method: tls
  IMS Enabled? y






   Near-end Node Name: procr                   Far-end Node Name: sm1
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          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
```

The screen below shows signaling group 32 which is used in the sample configuration as the "Secondary" signaling group to be associated with trunk group 32 for routing local PSTN calls from branch phones to Session Manager (for onward routing to local branch AudioCodes MP-118 media gateway) in Normal Mode. Note that all the settings for this signaling group are identical to those for signaling group 42 except the following:

- **Transport Method** is set to "tcp" (the port numbers will change automatically to "5060")
- **IMS Enabled?** is set to "n"

```
add signaling-group 32
                                SIGNALING GROUP

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




   Near-end Node Name: procr                  Far-end Node Name: sm1
 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
```

4.8.2. SIP Trunk Groups

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 4.8.1**
- **Number of Members**: Equal to the maximum number of concurrent calls supported

```
add trunk-group 42                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 42                   Group Type: sip          CDR Reports: y
  Group Name: SIP endpoints               COR: 1        TN: 1        TAC: *142
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                       Signaling Group: 42
                                                     Number of Members: 20
```

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

```
add trunk-group 42                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



                     Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n








 Show ANSWERED BY on Display? y
```

Navigate to **Page 4**, and enter "127" for the **Telephone Event Payload Type** field. This setting must match the configuration on AudioCodes MP-118 (see **Section 7.6**). Use default values for all other fields.

```
add trunk-group 42                                          Page   4 of  21
                              PROTOCOL VARIATIONS

                        Mark Users as Phone? n
              Prepend '+' to Calling Number? n
        Send Transferring Party Information? y

                      Send Diversion Header? n
                    Support Request History? y
             Telephone Event Payload Type: 127
```

The trunk group 32 used for routing local PSTN calls from branch phones is similarly configured (not shown).

4.9. Configure Route Patterns

Configure a route pattern to correspond to each of the two 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 4.8.2**
- **FRL**: Facility Restriction Level that allows access to this trunk, "0" being least restrictive

```
change route-pattern 42                                       Page   1 of   3
                    Pattern Number: 42  Pattern Name: URE SIP Trunk
                             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: 42   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
```

```
change route-pattern 32                                       Page   1 of   3
                    Pattern Number: 32  Pattern Name: Branch Local PSTN
                             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
```

4.10. Configure Private Numbering

Use the "change private-numbering 0" command to define the calling party number to be sent. Add an entry for the trunk group defined in **Section 4.8.2**. In the example shown below, all calls originating from a 5-digit extension beginning with 4 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.

```
change private-numbering 0                                     Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext               Trk        Private           Total
Len Code              Grp(s)     Prefix            Len
 5  4                                              5     Total Administered: 1
                                                            Maximum Entries: 540
```

4.11. Configure AAR

Use the "change aar analysis" command to add an entry for the extension range corresponding to the SIP telephones as configured in **Section 4.7** (required for feature server/Off-PBX-Station support). 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 4.9**
- **Call Type**: "aar"

```
change aar analysis 4                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    2

          Dialed           Total     Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    4                     5    5     42        aar          n
    49998                 5    5     32        aar          n
    50000                 5    5     1         aar          n
    55000                 5    5     2         aar          n
    7                     7    7     254       aar          n
    8                     7    7     254       aar          n
    9                     7    7     254       aar          n
                                                            n
                                                            n
                                                            n
                                                            n
                                                            n
                                                            n
                                                            n
                                                            n
```

4.12. Automatic Route Selection (ARS)

The ARS entries highlighted in the section focus on the local and long distance dialing from branch locations.

4.12.1. ARS Access Code

The sample configuration designates '9' as the ARS Access Code as shown below on **Page 1** of the **change feature-access-codes** form. Calls with a leading 9 will be directed to the ARS routing table.

```
change feature-access-codes                                         Page   1 of   8
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: *56
                       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: *57    Deactivation: *58
Call Forwarding Activation Busy/DA:        All: *88    Deactivation: *89
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code: *59
                       Call Pickup Access Code: *55
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
                   Contact Closure   Open Code:          Close Code:
```

4.12.2. Location Specific ARS Digit Analysis

The "change ars analysis location x y" command is used to make location specific routing entries where the x is the location number and the y is the dialed digit string to match on. Each branch location has an ARS entry for the local area code of the branch. These ARS location tables are used by Communication Manager for source based routing. The location specific ARS entries for each branch are shown below. Route Pattern 32 as defined in **Section 4.9** is used when a match is made on any of these ARS entries.

```
change ars analysis location 11 1609                                Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  11           Percent Full:    2

          Dialed            Total     Route     Call   Node  ANI
          String           Min  Max  Pattern    Type   Num   Reqd
    1609                   11   11    32        natl         n
```

```
change ars analysis location 12 1908                            Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  12          Percent Full:    2

          Dialed           Total      Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    1908                  11   11     32      natl         n
```

```
change ars analysis location 13 1732                            Page   1 of   2

                             ARS DIGIT ANALYSIS TABLE
                                   Location:  13          Percent Full:    1

          Dialed           Total      Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    1732                  11   11     32      natl         n
```

4.12.3. Global ARS Digit Analysis

The “change ars analysis y” command is used to make global routing entries where the y is the dialed digit string to match. A match on this table can occur if there is no match on the ARS location table (**Section 4.12.2**) for the branch originating the call. The global ARS table as used in the sample configuration is shown below. Long distance calls, 1 + 10 digits, will match the Dialed String of 1 with 11 digits and select Route Pattern 3.

Route Pattern 3 is configured to use a Trunk Group that connects to the Avaya G-Series Media Gateway at the Headquarters location for PSTN terminations. The configuration of Route Pattern 3 the associated PSTN Trunk Group and the Avaya G-Series Media Gateway are out of scope of these Application Notes and are therefore not included.

```
display 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
```

5. Configure Session Manager

This section provides the procedures for configuring Session Manager as provisioned in the reference configuration. Session Manager is comprised of two functional components: the Session Manager server and the System Manager management server. All SIP call provisioning for Session Manager is performed via the System Manager web interface and are then downloaded into Session Manager.

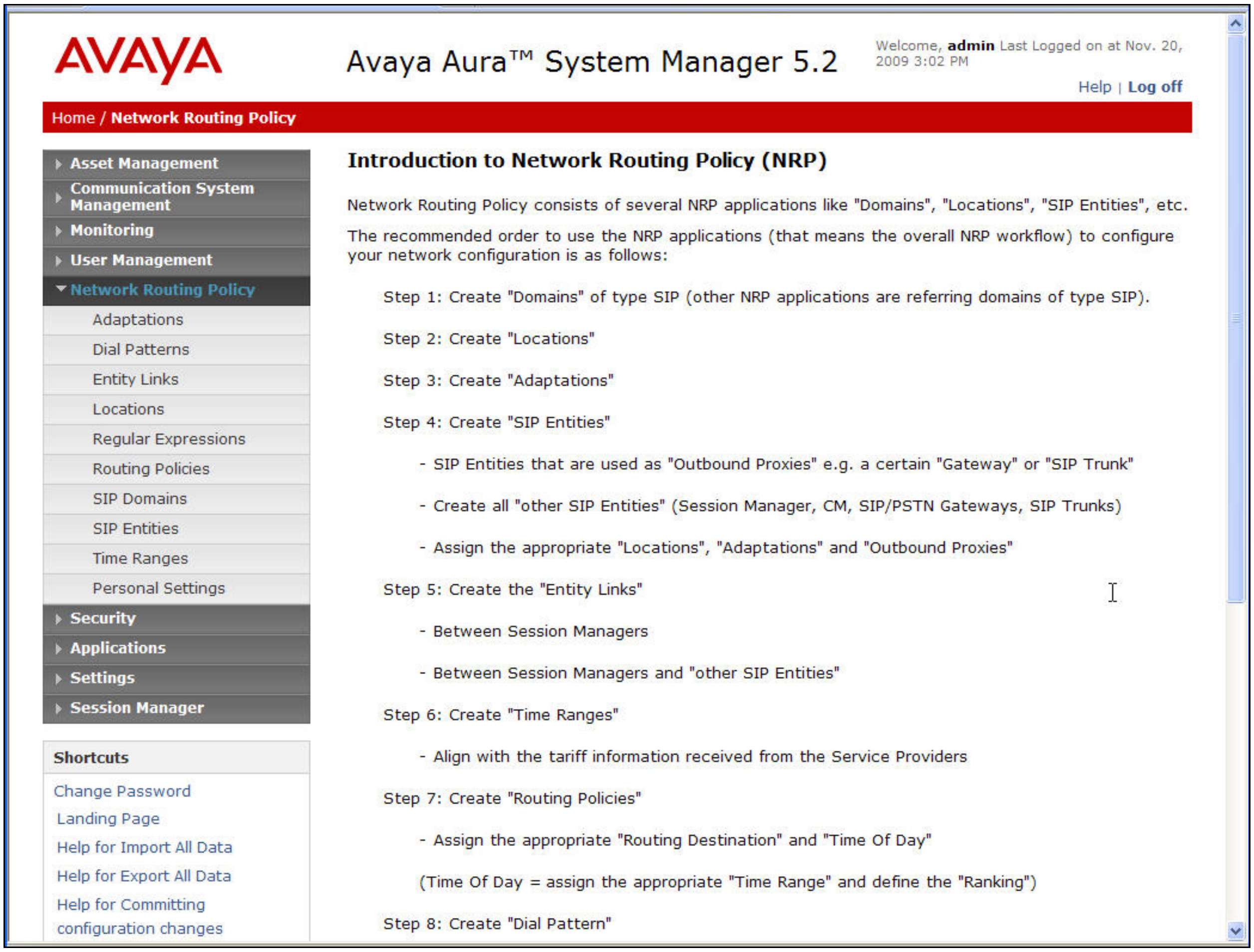
The following sections assume that Session Manager and System Manager have been installed and that network connectivity exists between the two platforms.

The Session Manager server contains an SM-100 security module that provides the network interface for all inbound and outbound SIP signaling and media transport to all provisioned SIP entities. For the Session Manager used in the reference 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 reference configuration, the IP address assigned to the Session Manager management interface is 10.1.1.171. In the reference configuration, the SM-100 interface and the management interface were both connected to the same IP network. 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** which define 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 host name to IP resolution
- Add Communication Manger as a Feature Server
- **User Management** for SIP telephone users

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**).

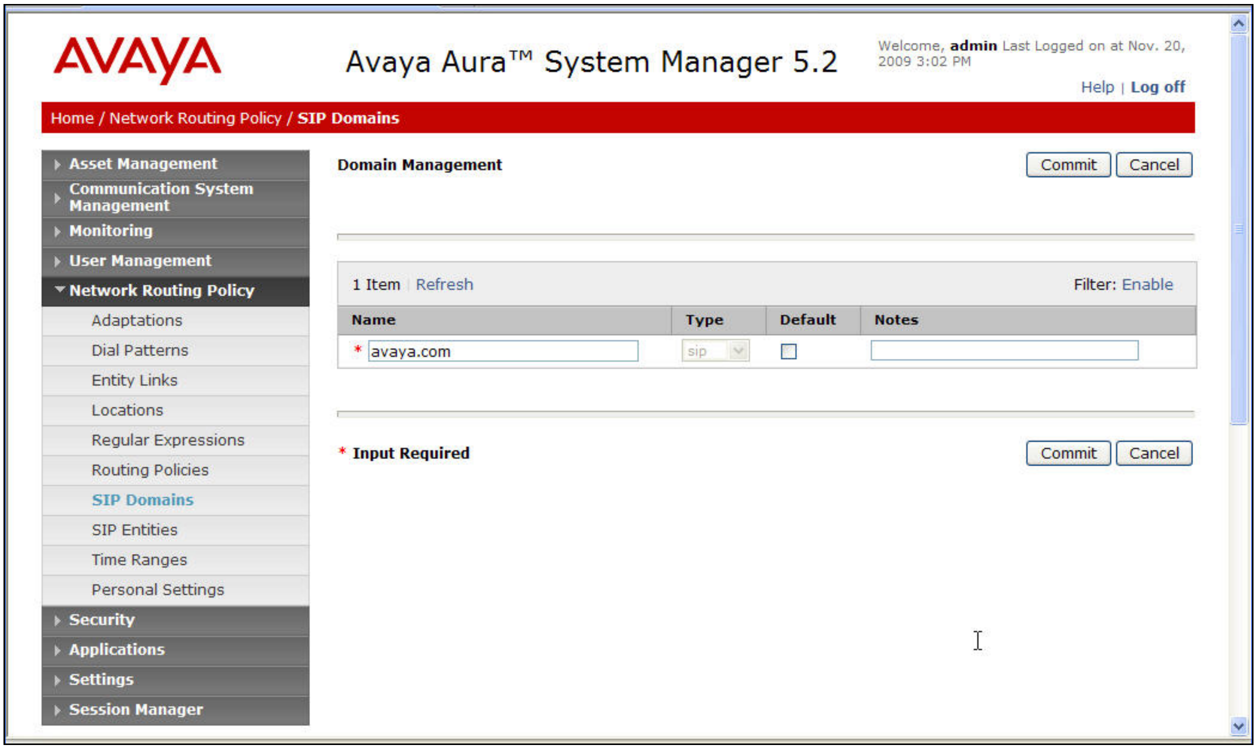


5.1. Specify SIP Domain

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:

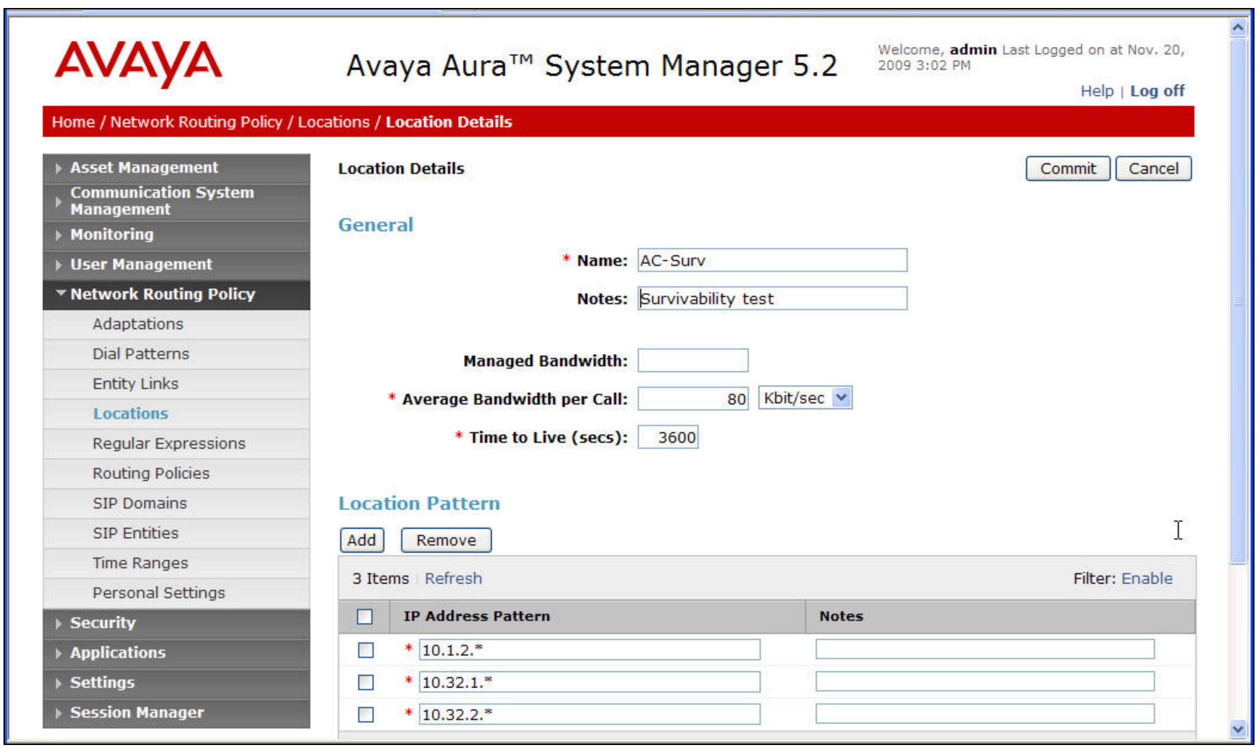
- **Name**: A descriptive name
- **Notes**: Descriptive text (optional)

The remaining fields under *General* can be filled in to specify bandwidth management parameters between Session Manager and this location. These were not used in the sample configuration, and reflect default values. Note also that although not implemented in the sample configuration, routing policies can be defined based on location.

Under *Location Pattern*:

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

The screen below shows addition of the "AC-Surv" location for the Headquarters site, which includes Session Manager (10.1.2 subnet), Communication Manager (10.32.2 subnet), and all SIP telephones located at this location (10.32.1 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 (not shown). Each branch Location is similarly configured as above with its own **Name** (e.g., “AC-Surv-BR2” for Branch 2) and **IP Address Patterns** (e.g., “192.168.75.*” for Branch 2).

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 and the Communications Manager.

Select **SIP Entities** on the left and click on the **New** button (not shown) on the right.

Under *General*:

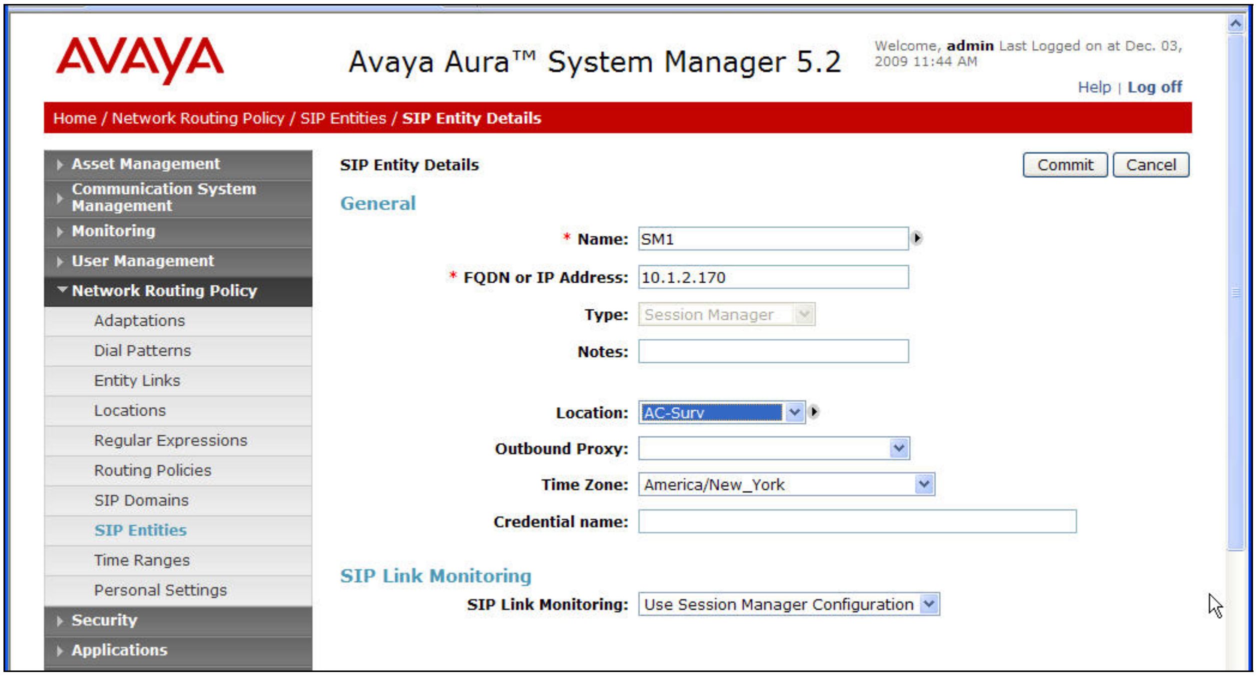
- **Name** A descriptive name
- **FQDN or IP Address**: FQDN or IP address of the Session Manager or the signaling interface on the telephony system
- **Type**: “Session Manager” for Session Manager, “CM” for Communication Manager
- **Adaptation**: Leave blank
- **Location:** Select the Location configured in previous step
- **Time Zone:** Select the proper time zone for this installation

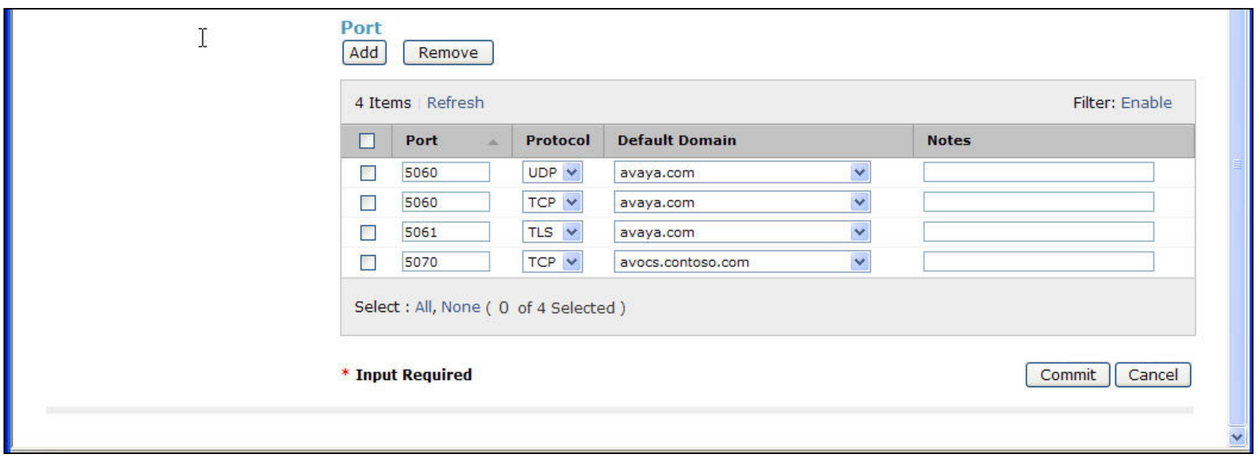
Under *Port* (for adding Session Manager Entity only), click **Add**, then edit the fields in the resulting new row as shown below:

- **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**

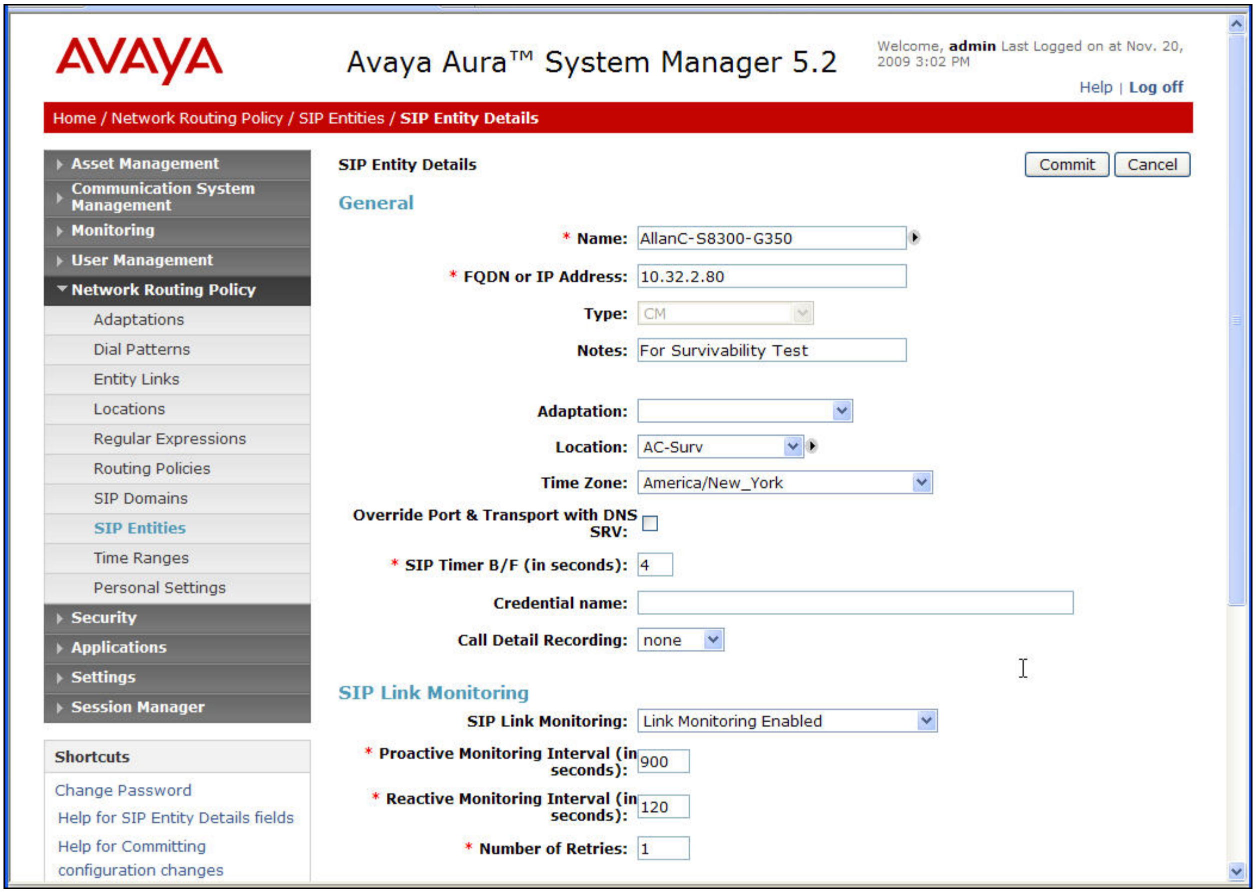
Default settings can be used for the remaining fields. Click **Commit** to save the SIP Entity definition.

The following screens show addition of Session Manager. The IP address of the SM-100 Security Module is entered for **FQDN or IP Address**. TLS port 5061 is used for communication with Communication Manager.

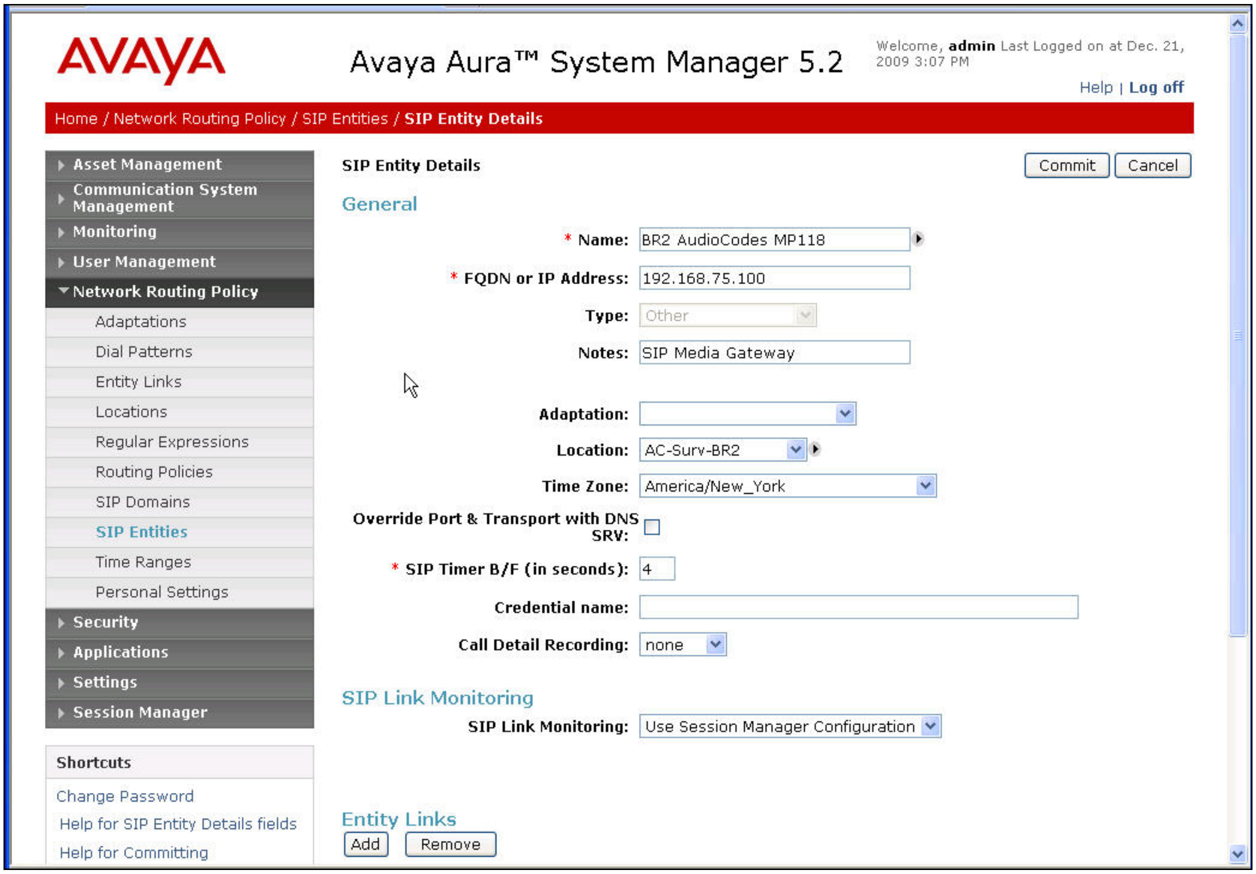

AVAYA
Avaya Aura™ System Manager 5.2
Welcome, admin Last Logged on at Dec. 03, 2009 11:44 AM
Help | Log off
Home / Network Routing Policy / SIP Entities / SIP Entity Details
Asset Management
Communication System Management
Monitoring
User Management
Network Routing Policy
Adaptations
Dial Patterns
Entity Links
Locations
Regular Expressions
Routing Policies
SIP Domains
SIP Entities
Time Ranges
Personal Settings
Security
Applications
SIP Entity Details
Commit
Cancel
General
* Name: SM1
* FQDN or IP Address: 10.1.2.170
Type: Session Manager
Notes:
Location: AC-Surv
Outbound Proxy:
Time Zone: America/New_York
Credential name:
SIP Link Monitoring
SIP Link Monitoring: Use Session Manager Configuration


Port
Add
Remove
4 Items Refresh
Filter: Enable
Port
Protocol
Default Domain
Notes
5060 UDP avaya.com
5060 TCP avaya.com
5061 TLS avaya.com
5070 TCP avocs.contoso.com
Select : All, None (0 of 4 Selected)
* Input Required
Commit
Cancel

The following screen shows the results of adding Communication Manager. In this case, **FQDN or IP Address** is the IP address for the Communication Manager since the G-Series Media Gateway used in the sample configuration has its signaling interface integrated into the Communication Manager processor. For other Avaya Media Gateways with C-LAN board installed, the IP address of the C-LAN board in the Media Gateway should be specified. Note the "CM" selection for **Type**.



The following screen shows the results of adding the branch AudioCodes MP-118 for Branch 2. 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

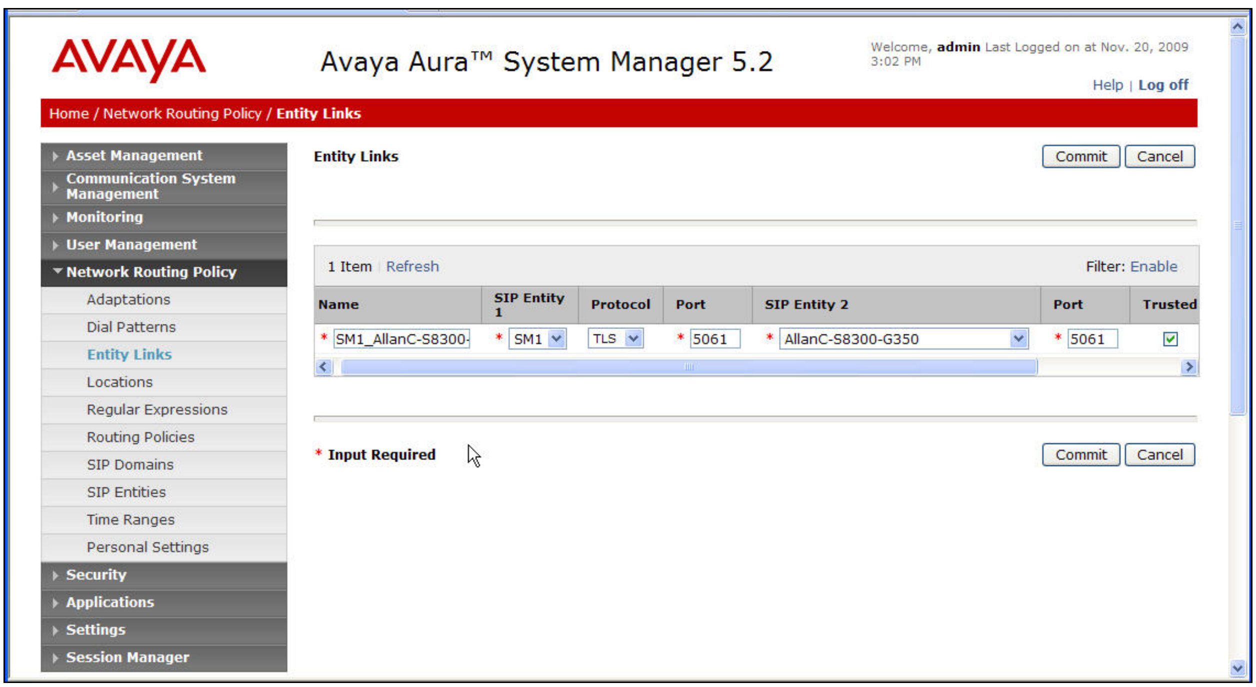
A SIP trunk between Session Manager and a telephony system is described by an Entity link. In the sample configuration, 2 Entity Links were configured between Session Manager and Communication Manger (corresponding to the 2 Signaling Groups and 2 Trunk Groups configured in Communication Manager in **Section 4.8**). Additional Entity Links were 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 previous section
- **Protocol**: Select "TLS" or "TCP"
- **Port**: Port number to which the other system sends SIP requests.
- **SIP Entity 2**: Select the Communication Manager SIP Entity configured in previous section.
- **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 1st Entity Link configured between Session Manager and Communication Manager for regular call signaling and audio media transport.



The 2nd Entity Link between Session Manager and Communication Manager (for routing branch local calls to PSTN in Normal Mode) is similarly configured (not shown). In the sample configuration, this second Entity Link was configured to use **Protocol** TCP and **Port** 5060.

The screen below shows the Entity Link between Session Manager and the Branch 2 AudioCodes MP-118. Note the **Port** setting 5070 specified for the branch AudioCodes gateway.

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 the branch local PSTN call (sent over to Session Manager from Communication Manager after its location-based routing decision) to the branch AudioCodes MP-118. Each branch should have its own Routing Policy defined.

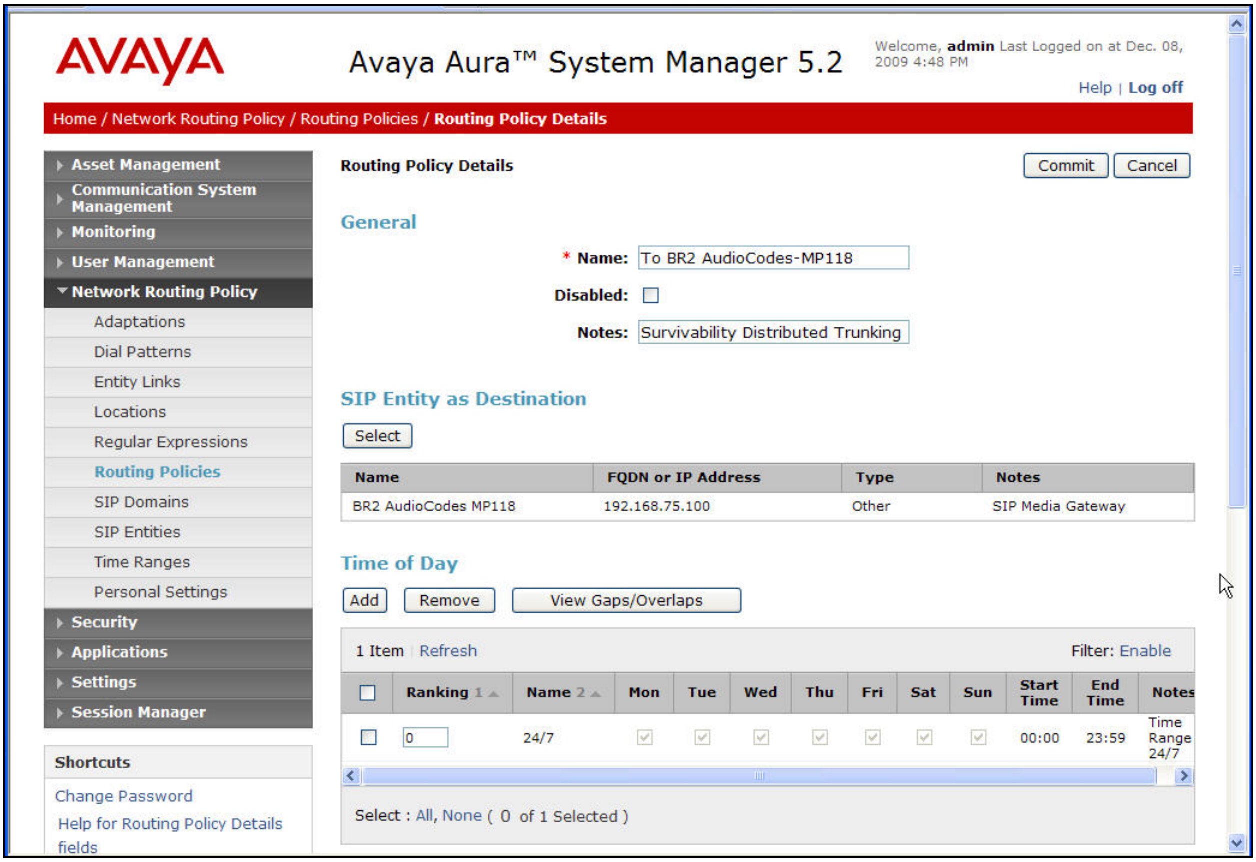
To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

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 branch SIP entity to which this routing policy applies.

Under *Time of Day*:
Click **Add**, and select the default "24/7" time range.

Defaults can be used for the remaining fields. Click **Commit** to save the Routing Policy definition. The following screen shows the Routing Policy for routing local PSTN calls to Branch 2.



Routing Policies for other branches are similarly configured (not shown).

5.6. Add Dial Patterns

Define a Dial Pattern for matching local PSTN calls based on Area Codes. A Dial Patterns is then associated with a Routing Policy to direct calls with the matched Area Code to the branch where the call to the PSTN will be a non-toll local call.

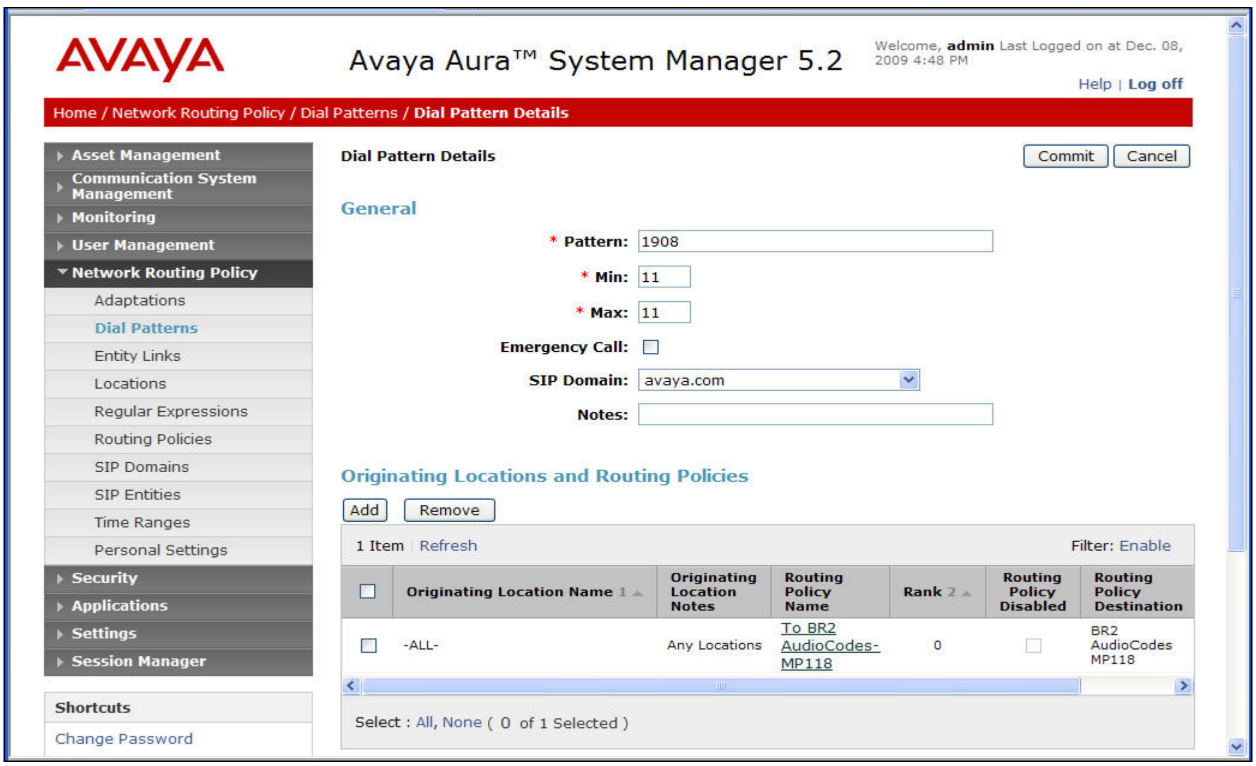
To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following, as shown in the screens below:

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 screen shows the Dial Pattern defined for routing local PSTN calls to Branch 2.



Dial Patterns for other branches are similarly configured (not shown).

5.7. Add 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):

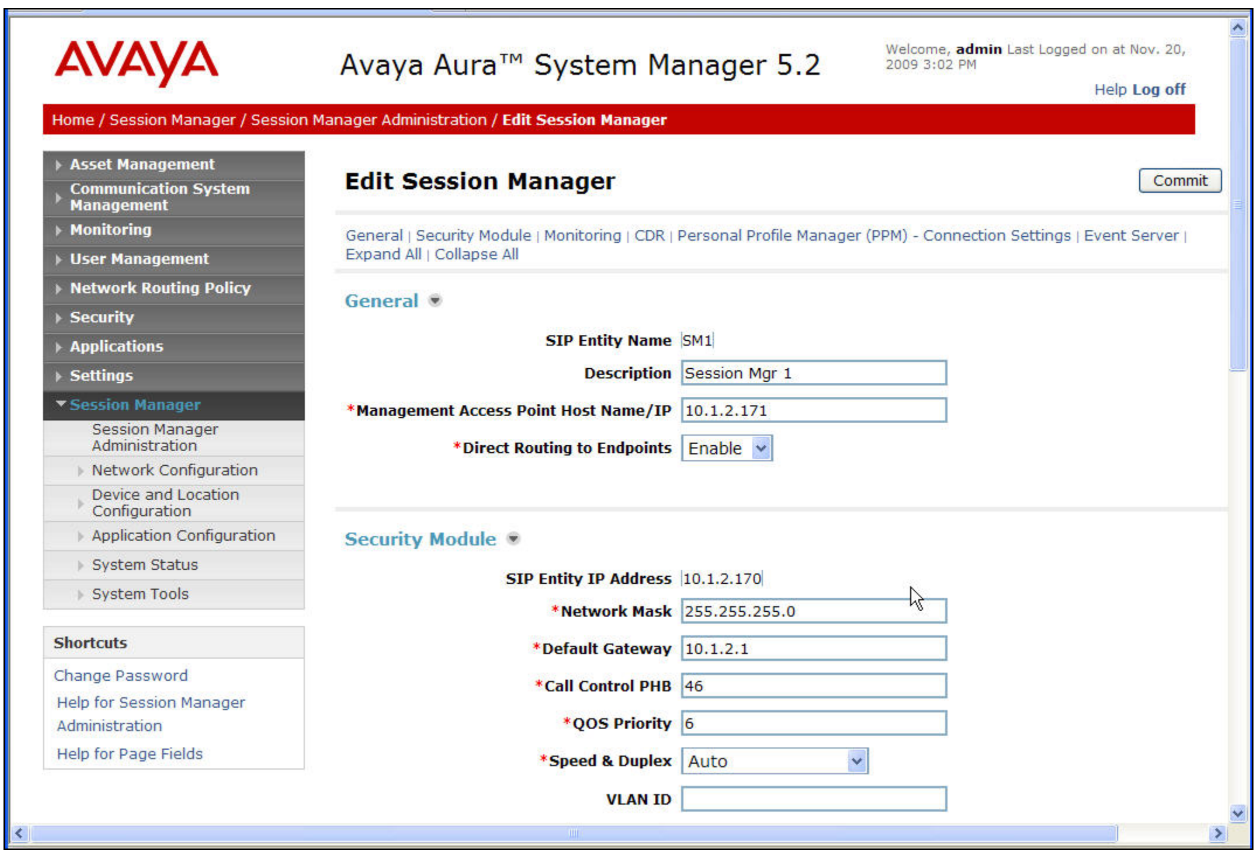
Under *General*:

- **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.

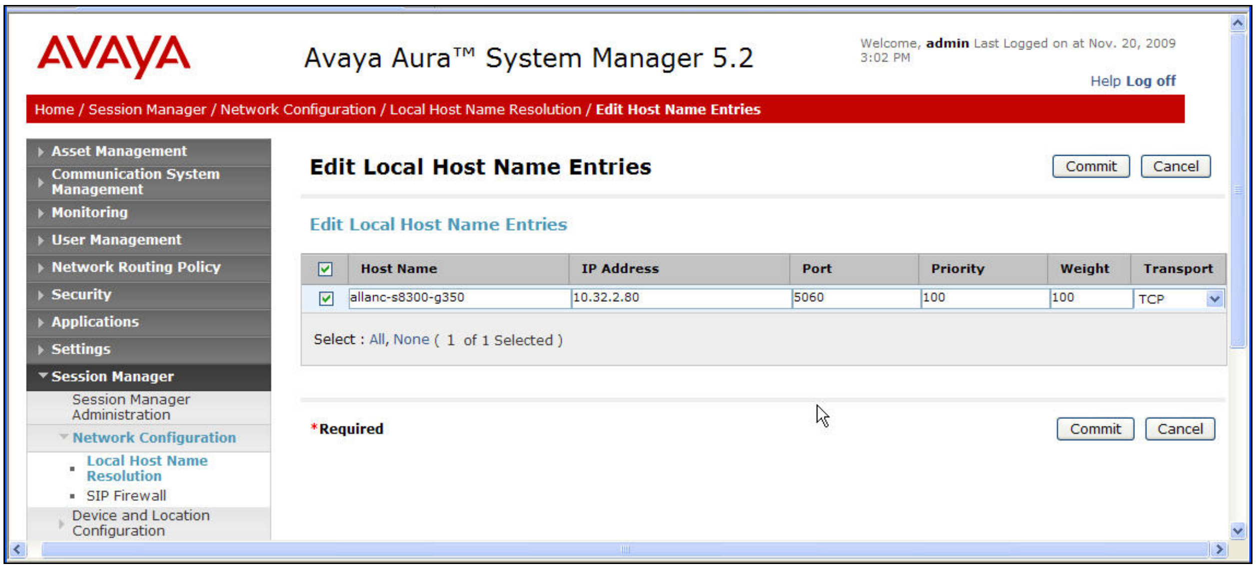


5.8. Define Local Host Name Resolution

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.



5.9. Add Communication Manger as a Feature Server

In order for Communication Manager to provide configuration and Feature Server support to SIP telephones when they register to Session Manager, Communication Manager must be added as an application for Session Manager. This is a four step process.

Step 1

Select **Applications → Entities** on the left. Click on **New** (not shown). Select "CM" **Type** and in the displayed "New CM Instance" page, enter the following fields. Use defaults for the remaining fields:

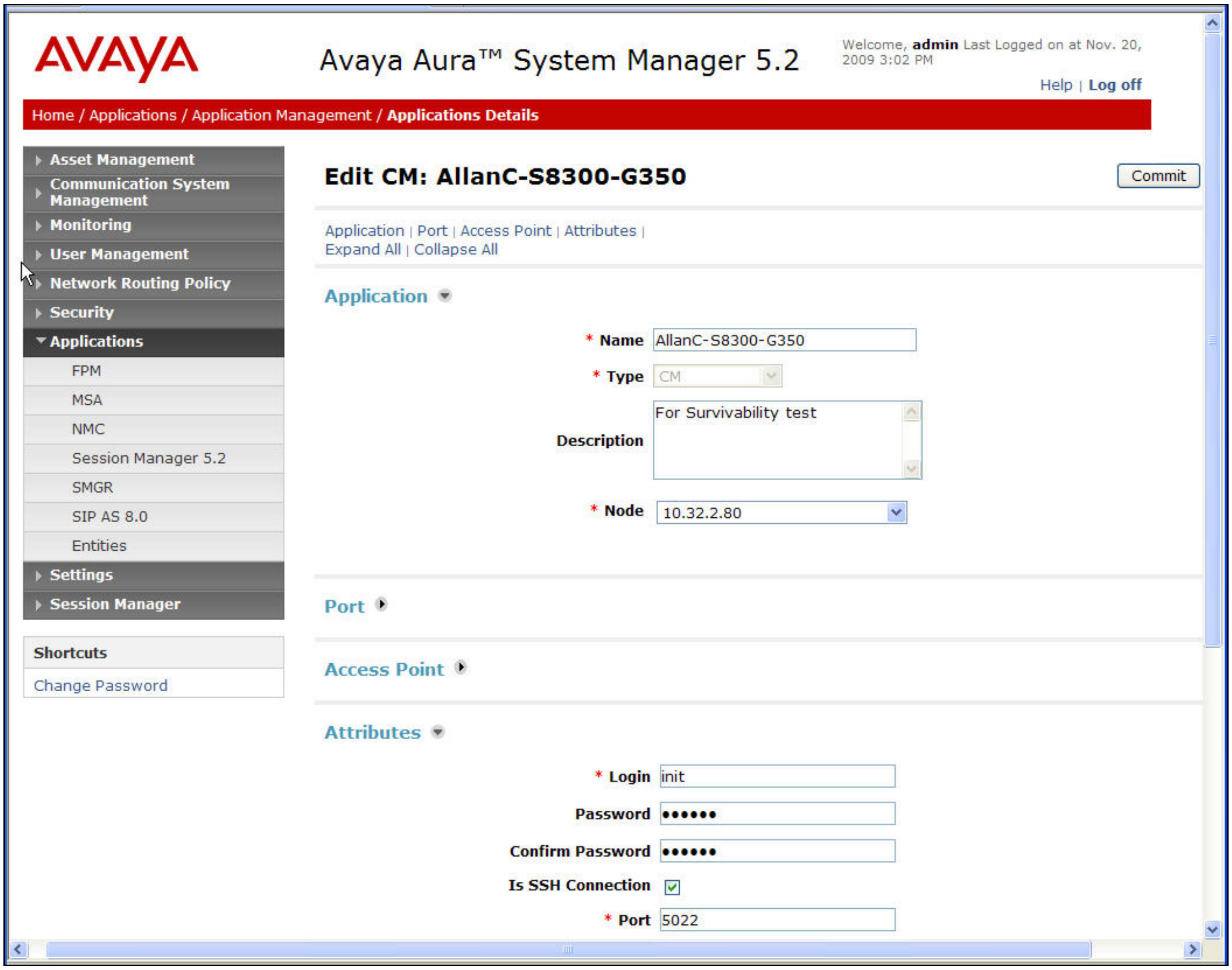
- **Name**: A descriptive name
- **Type**: "CM"
- **Node**: Select IP address for Communication Manager SAT access

Under the *Attributes* section, enter the following fields, and use defaults for the remaining fields:

- **Login**: Login used for SAT access
- **Password**: Password used for SAT access
- **Confirm Password**: Password used for SAT access

Click on **Commit**. This will set up data synchronization with Communication Manager to occur periodically in the background.

The screen shown below is the Edit screen since the Application Entity has already been added.


AVAYA
Avaya Aura™ System Manager 5.2
Welcome, admin Last Logged on at Nov. 20, 2009 3:02 PM
Help | Log off
Home / Applications / Application Management / Applications Details
Asset Management
Communication System Management
Monitoring
User Management
Network Routing Policy
Security
Applications
FPM
MSA
NMC
Session Manager 5.2
SMGR
SIP AS 8.0
Entities
Settings
Session Manager
Shortcuts
Change Password
Edit CM: AllanC-S8300-G350
Commit
Application | Port | Access Point | Attributes |
Expand All | Collapse All
Application
* Name AllanC-S8300-G350
* Type CM
Description For Survivability test
* Node 10.32.2.80
Port
Access Point
Attributes
* Login init
Password ••••••
Confirm Password ••••••
Is SSH Connection
* Port 5022

Step 2

Select **Session Manager → Application Configuration → Applications** on the left. Click on **New** (not shown). Enter the following fields, and use defaults for the remaining fields:

- **Name**: A descriptive name
- **SIP Entity**: Select the Communication Manager SIP Entity (see **Section 5.3**)

Click on **Commit**.

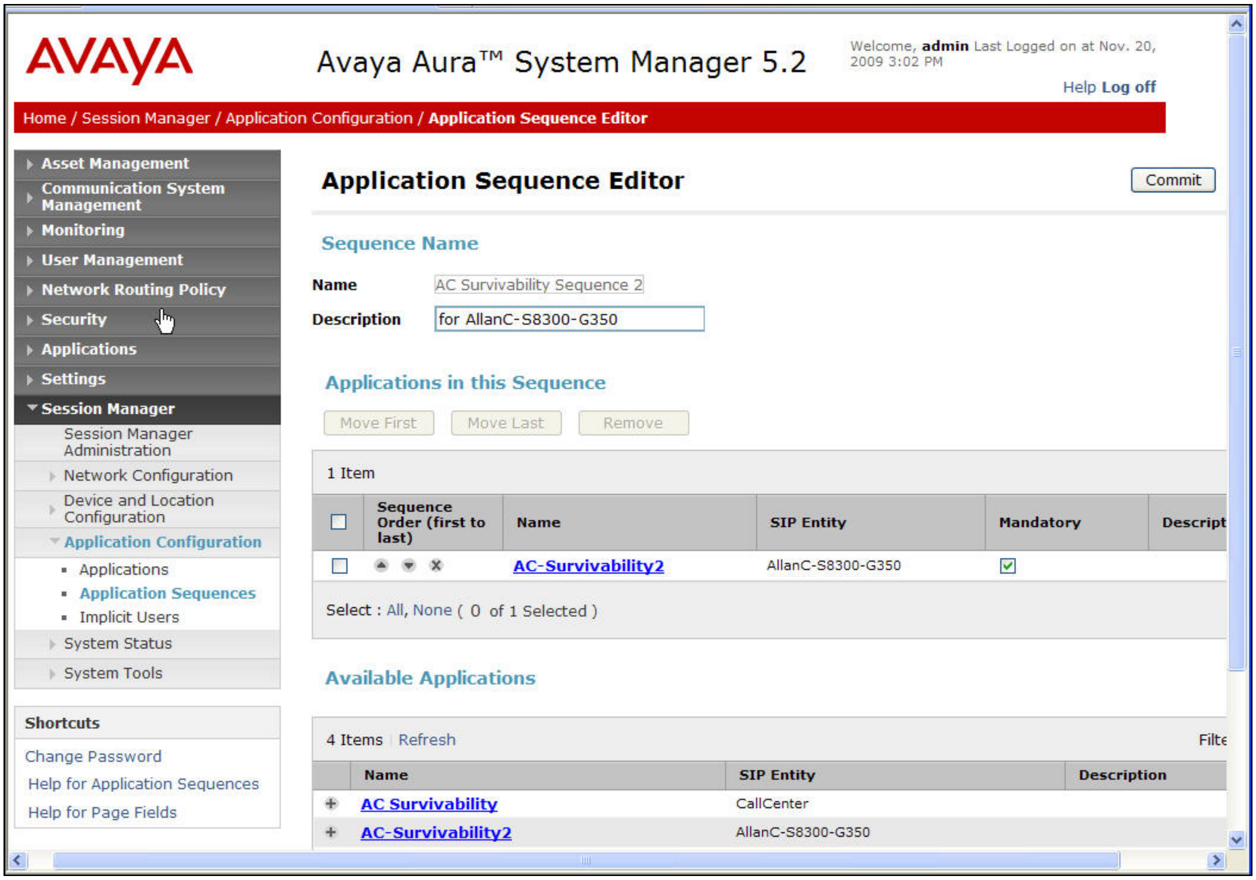
The screen shown below is the Edit screen since the Application has already been configured.

Step 3

Select **Session Manager → Application Configuration → Application Sequences** on the left. Click on **New** (not shown). Enter a descriptive Name. Click on the "+" sign next to the appropriate *Available Applications*, and the selected available application will be moved up to the *Applications in this Sequence* section. In this sample configuration, "AC-Survivability2" was selected, as shown in the screen below (which is the Edit screen since the Application Sequence has already been configured).

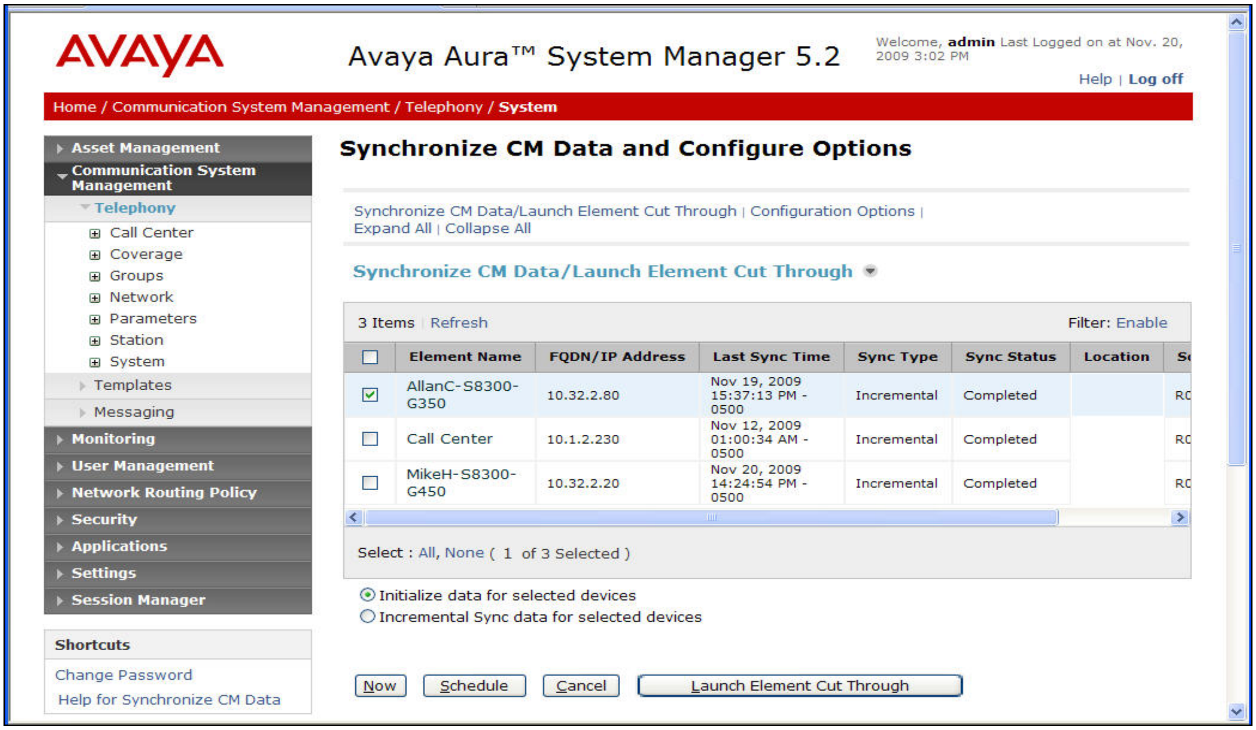
Click on **Commit**.

Note that the entry "AC-Survivability" listed in the screen was not used in the sample configuration. It was set up for other purposes.

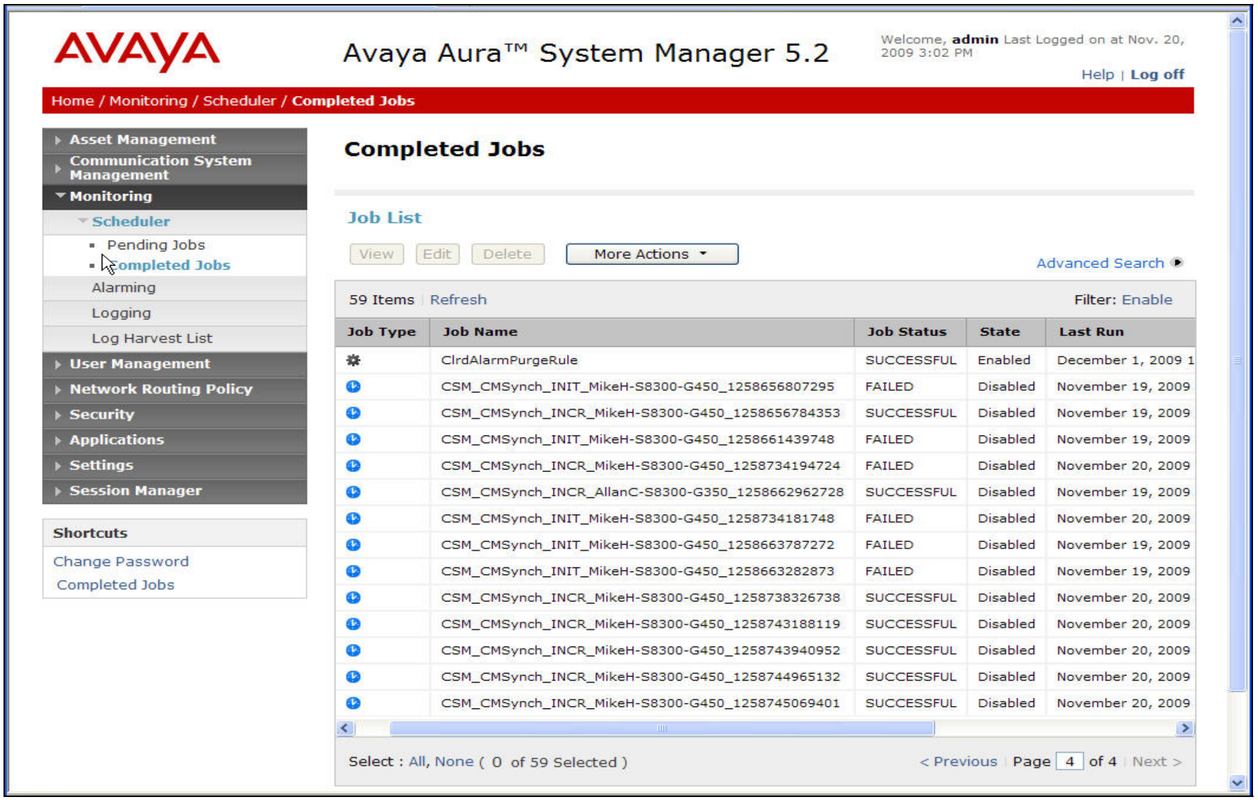


Step 4

Select **Communication System Management → Telephony** on the left. Select the appropriate Element Name (“AllanC-S8300-G350” in this case). Select **Initialize data for selected devices**. Then click on **Now**. This will cause a data synchronization task to start. This may take some time to complete.

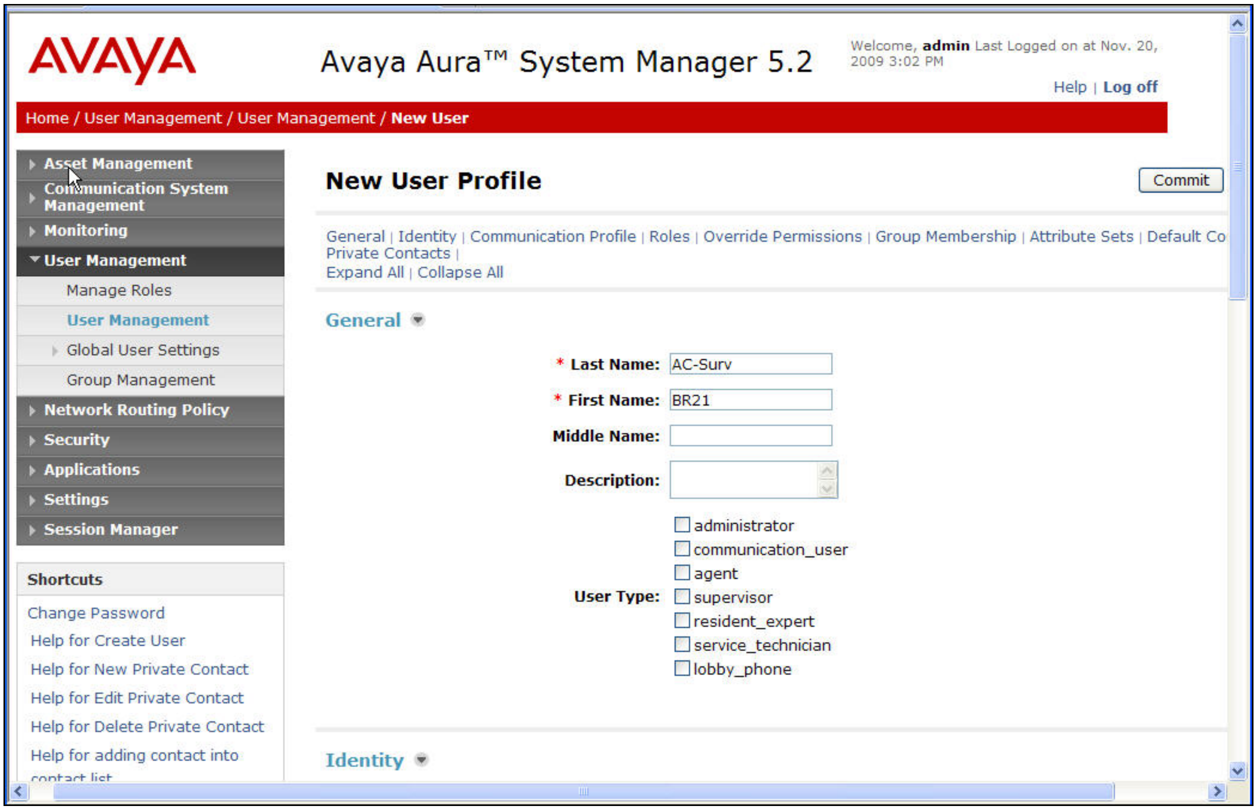


Use the menus on the left under **Monitoring → Scheduler → Completed Jobs** to determine when the task has completed, as shown below (see entry with embedded Communication Manager name "AllanC-S8300-G350" for the sample configuration).



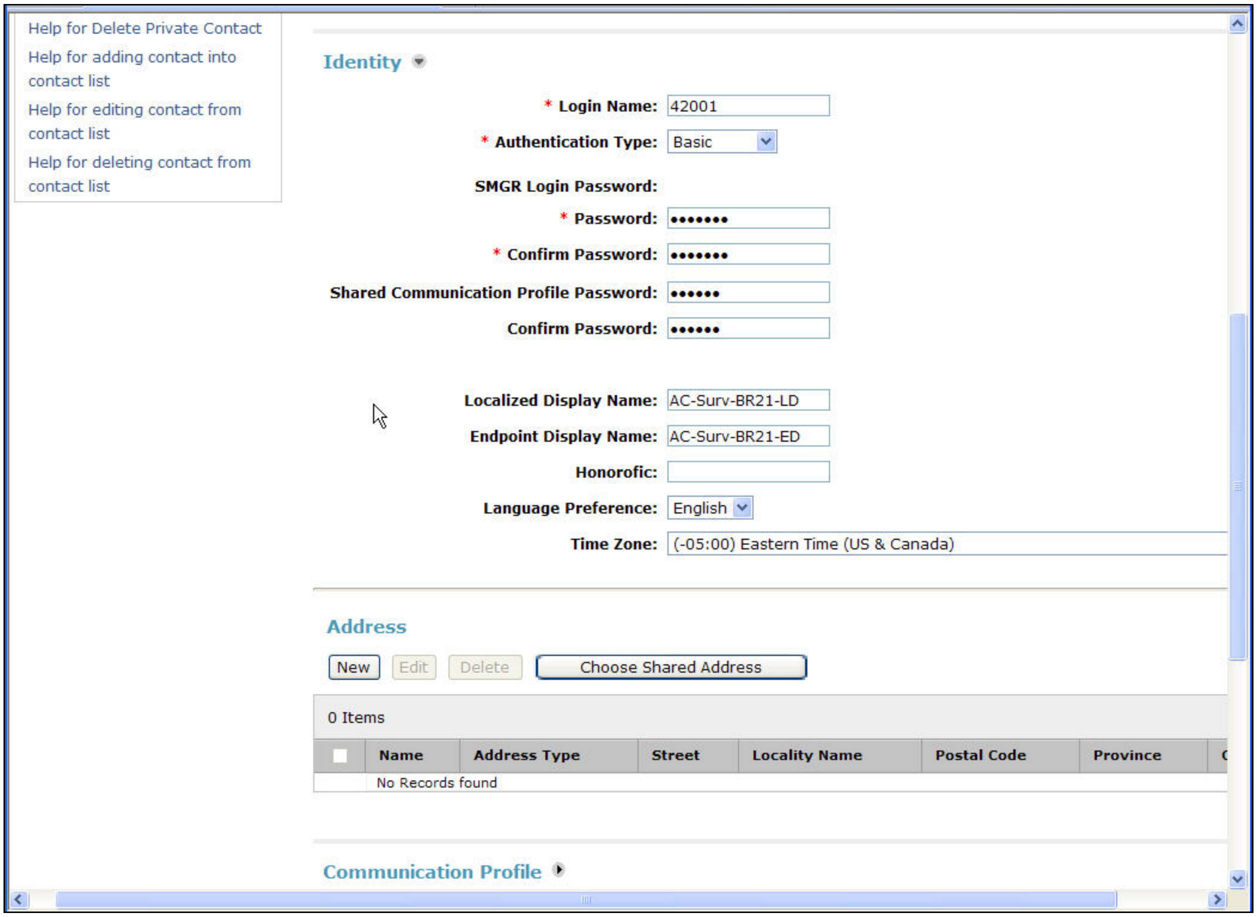
5.10. User Management for Adding SIP Telephone Users

Users must be added to Session Manager corresponding to the SIP stations added in Communication Manager (see **Section 4.6**). Select **User Management → User Management** on the left. Then click on **New** (not shown) to open the New User Profile page. Enter a **First Name** and **Last Name** for the user to add.



Click on *Identity* to expand that section. Enter the following fields, and use defaults for the remaining fields:

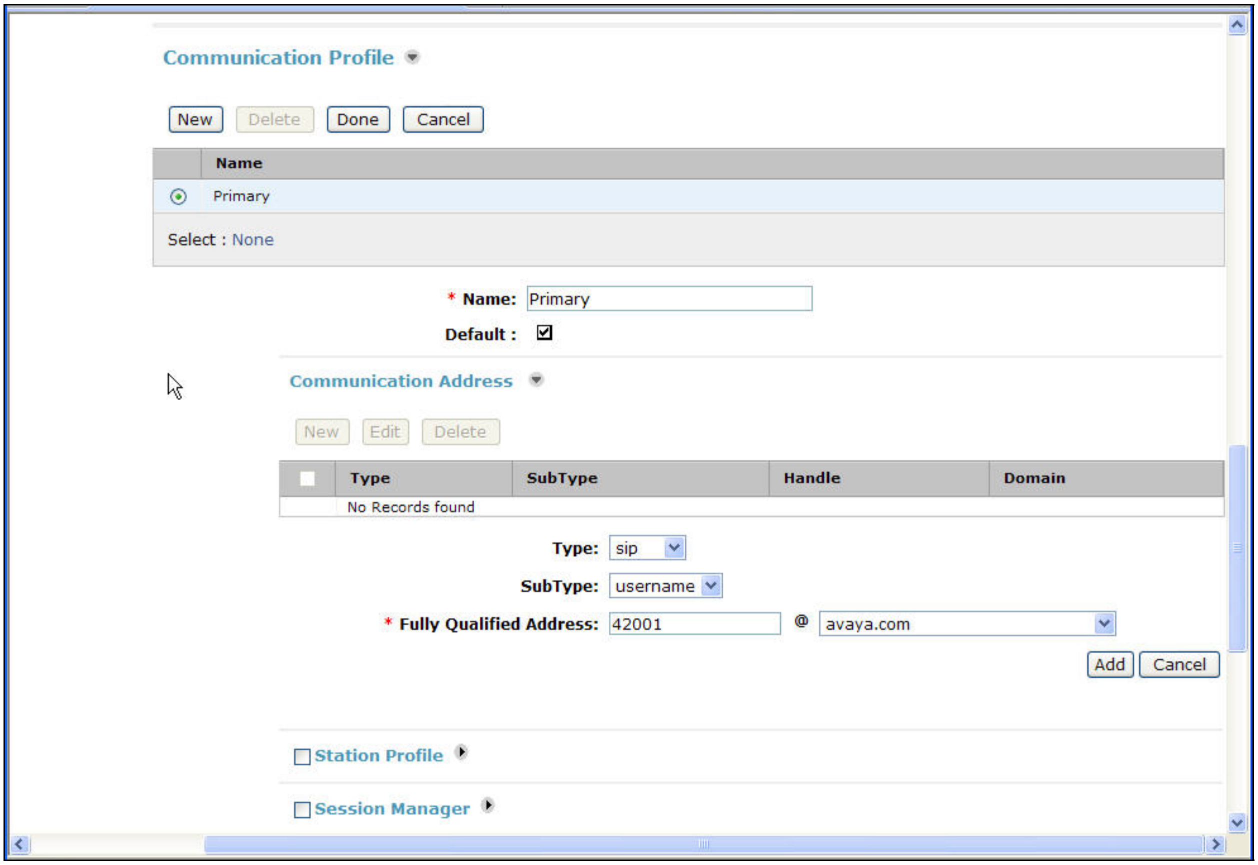
- **Login Name**: Telephone extension (see **Section 4.7**)
- **SMGR Login Password**:
  - **Password**: Password to log into System Manger
  - **Shared Communication Profile Password**: Password to be entered by the user when logging onto the telephone
- **Localized Display Name**: Name to be used as calling party
- **Endpoint Display Name**: Full name of user
- **Language Preference**: Select the appropriate language preference
- **Time Zone:** Select the appropriate time zone



Click on *Communication Profile* to expand that section in the above screen. Then click on *Communication Address* to expand that section. Enter the following fields and use defaults for the remaining fields:

- **Type**: Select "sip"
- **SubType**: Select "username"
- **Fully Qualified Address**: Enter the extension and select the domain as specified in **Section 5.1**

Click on **Add** to add the record with the above information.

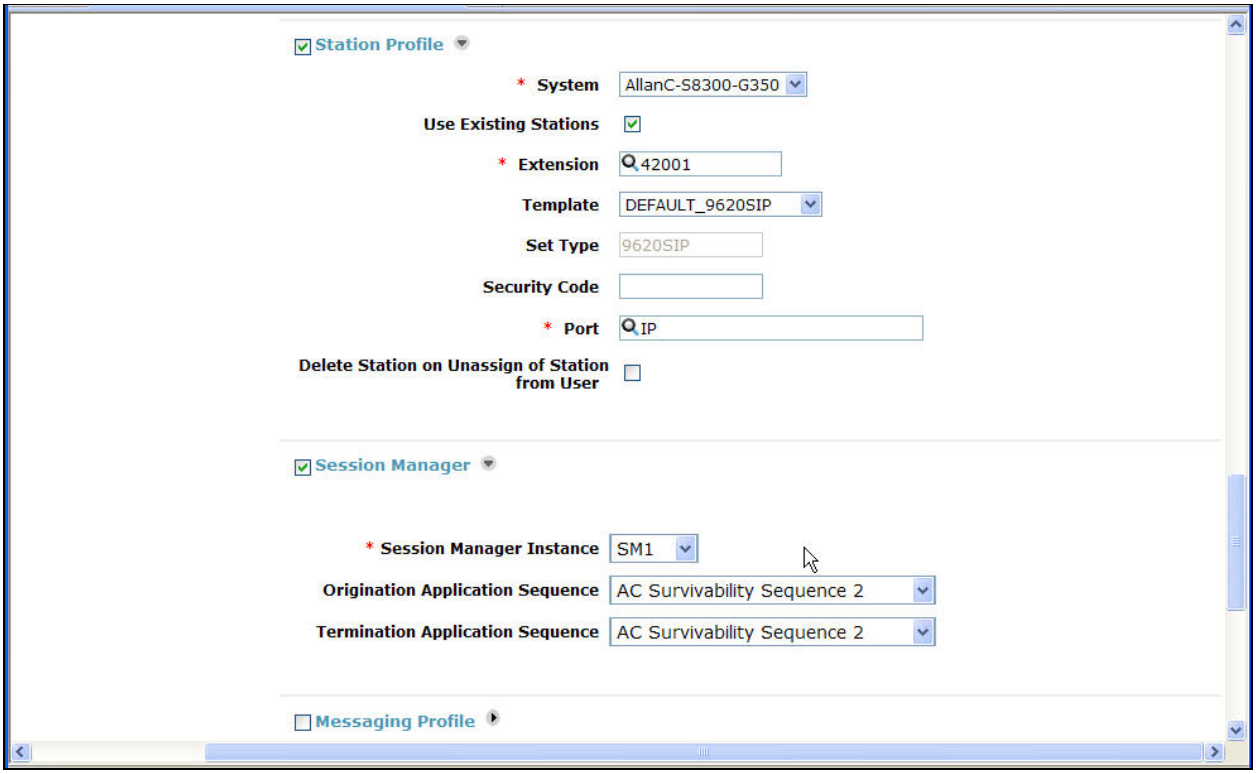


Click on *Station Profile* in the above screen to expand that section. Enter the following fields and use defaults for the remaining fields:

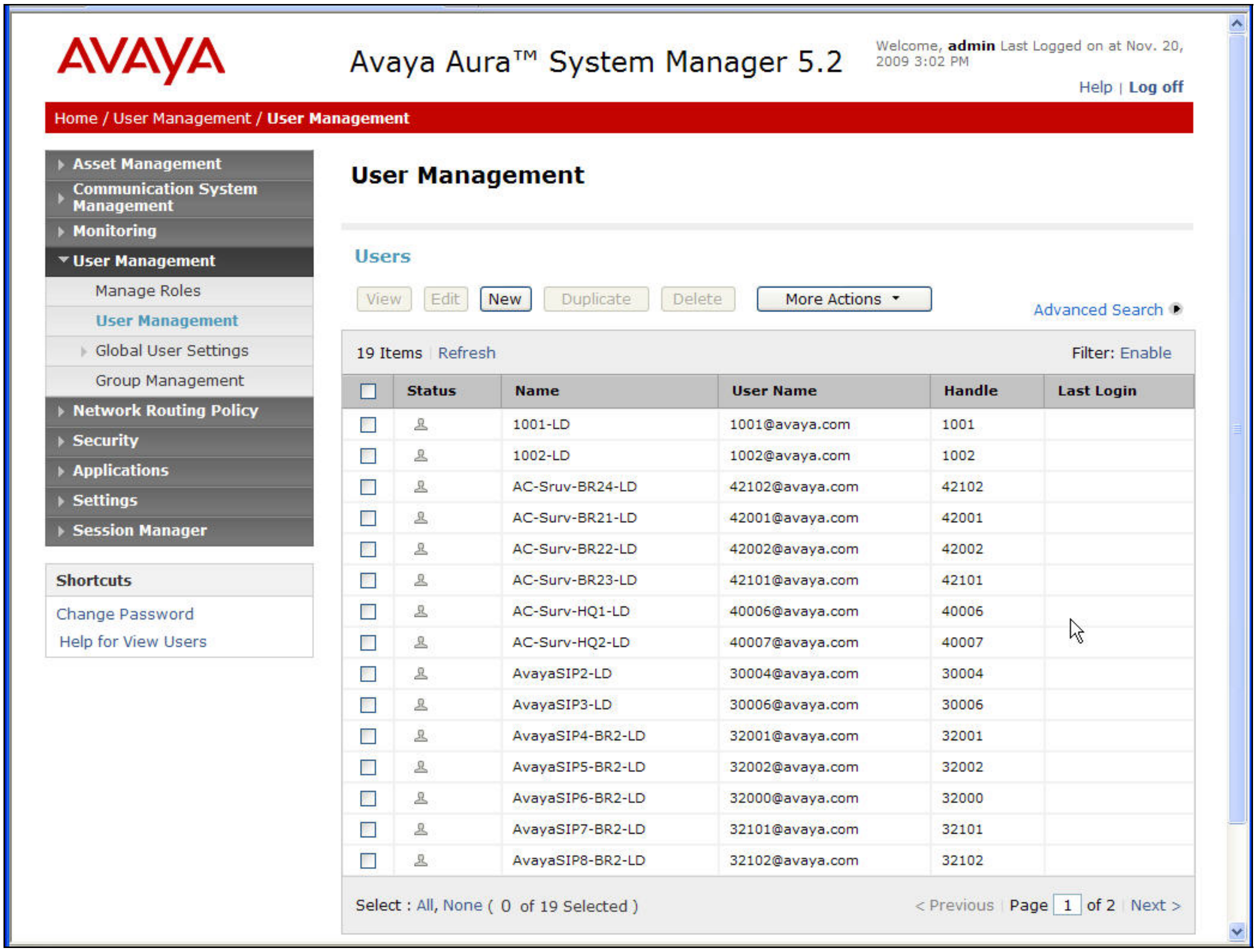
- **System**: Select the Communication Manager entity
- **Use Existing Stations**: Check this box
- **Extension**: Enter the extension
- **Template**: Select an appropriate template matching the telephone type as configured on Communication Manger (see **Section 4.7**)
- **Port**: Click on the Search icon to pick a port (in this case ("IP")

Click on *Session Manager* in the above screen to expand that section. Select the appropriate Session Manager server for **Session Manager Instance**. For **Origination Application Sequence** and **Termination Application Sequence**, select the Application Sequence configured in **Section 5.9 Step 3**.

Click on **Commit** (not shown).


Station Profile
* System
AllanC-S8300-G350
Use Existing Stations
* Extension
42001
Template
DEFAULT_9620SIP
Set Type
9620SIP
Security Code
* Port
IP
Delete Station on Unassign of Station from User
Session Manager
* Session Manager Instance
SM1
Origination Application Sequence
AC Survivability Sequence 2
Termination Application Sequence
AC Survivability Sequence 2
Messaging Profile

Repeat the above procedures to add each SIP telephone user for the Headquarters site as well as the branch site (including the analog phones connected to the FXS interface ports on the MP-118). The following User Management screen shows the SIP telephone users configured in the sample configuration for the Headquarters site and Branch 2 (40006 and 40007 are Headquarters Avaya 9600 SIP Phone users; 42001 and 42002 are Avaya 9600 SIP Phone users at Branch 2; 42101 and 42102 are analog phones connected to the MP-118 FXS ports at Branch 2).



6. Configure Avaya 9600 SIP Phones

The Avaya 9600 SIP Phones at all sites will use the Session Manager (10.1.2.170) as the SIP Proxy Server. The Avaya 9600 SIP Phones at the branch sites will also configure the on-site MP-118 (192.168.75.100 for Branch 2) as an additional call server for survivability. The table below shows an example of the SIP telephone configuration settings for the Headquarters and Branch 2.

	Headquarters	**Branch 2**
Extension	40006	42002
IP Address	10.32.1.105	192.168.75.50
Subnet Mask	255.255.255.0	255.255.255.0
Router	10.32.1.1	192.168.75.1
File Server	10.32.2.75	10.32.2.75
DNS Server	0.0.0.0	0.0.0.0
SIP Domain	avaya.com	avaya.com
SIP Proxy Server	10.1.2.170	10.1.2.170
Alternate SIP Proxy Server		192.168.75.100

Note that the alternate SIP Proxy Server can be configured manually on the Avaya 9600 SIP Phones or through the 46xxsetttings configuration file.

The configuration parameters of the Avaya 9600 SIP Phone specific to SIP Survivability in the 46xxsettings file are listed in the table below. See **Section 11** [7] for more details.

46xxsettings.txt Parameter Name	**Value Used in Sample Configuration**	**Description**
SIP_CONTROLLER_LIST	10.1.2.170:5060 ;transport=tcp, 192.168.75.100: 5060;transport= tcp	A priority list of SIP Servers for the phone to use for SIP services. The port and transport use the default values of 5061 and TLS when not specified. The setting used in the sample configuration shows the values used for this parameter for a phone in Branch 2. The Session Manager is the first priority SIP Server listed using port and transport of 5060 and TCP. Separated by a comma, the Branch 2 AudioCodes MP-118 is the next priority SIP Server using port 5060 and TCP transport.

		The SIP Server list for each branch would require different values for the SIP_CONTROLLER_LIST, e.g. the list for Branch 1 phones will include the Session Manager and the Branch 1 AudioCodes MP-118 while the list for Branch 2 phones will include the Session Manager and the Branch 2 AudioCodes MP-118. To accomplish this, the GROUP system value mechanism can be implemented as described in [7].
FAILBACK_POLICY	Auto	While in Survivable Mode, determines the mechanism to use to fail back to the centralized SIP Server. **Auto** = the phone periodically checks the availability of the primary controller and dynamically fails back.
FAST_RESPONSE_TIMEOUT	2	The timer terminates SIP INVITE transactions if no SIP response is received within the specified number of seconds after sending the request. Useful when a phone goes off-hook after connectivity to the centralized SIP Server is lost, but before the phone has detected the connectivity loss. The default value of 4 seconds may be retained if desired. After the SIP INVITE is terminated, the phone immediately transitions to Survivable Mode.
MSGNUM	5000	The number dialed when the Message button is pressed and the phone is in Normal Mode.
PSTN_VM_NUM	919081235000	The number dialed when the Message button is pressed and the phone is in Survivable Mode.
RECOVERYREGISTERWAIT	60	A Reactive Monitoring Interval. If no response to a "maintenance check" REGISTER request is received within the timeout period, the phone will retry the monitoring attempt after a randomly selected delay of 50% - 90%

		of this parameter.
DIALPLAN	40xxx|41xxx|42xxx|43xxx|911|9911|91xxxxxxxxxx|9011x.T	Enables the acceleration of dialing when the WAN is down and the AudioCodes SAS is active, by defining the dial plan used in the phone. In normal mode, the Avaya telephone does not require these settings to expedite dialing. The dialplan values used in the phone will generally match the values used by the AudioCodes MP-118 in **Section 7.6**. See [7] for additional format details on the DIALPLAN parameter.
DISCOVER_AVAYA_ENVIRONMENT	1	Automatically determines if the active SIP Server is an Avaya server or not.
SIPREGPROXYPOLICY	alternate	A policy to control how the phone treats a list of proxies in the SIP_CONTROLLER_LIST parameter **alternate** = remain registered with only the active controller **simultaneous** = remain registered with all available controllers
SIPDOMAIN	avaya.com	The enterprise SIP domain. Must be the same for all SIP controllers in the configuration. SIPDOMAIN is set to "avaya.com" in the sample configuration.

7. Configure AudioCodes MP-118

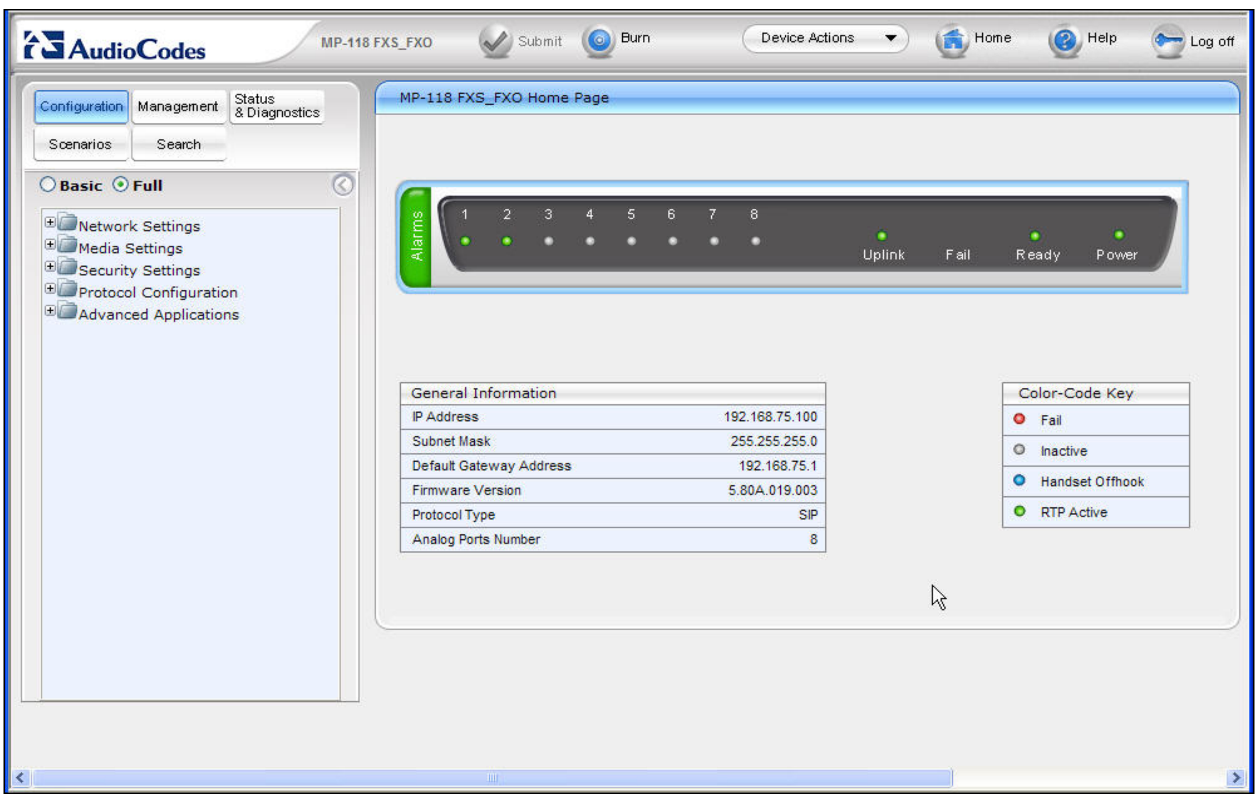
This section shows the necessary steps to configure the AudioCodes MP-118 Gateway to support the Avaya Session Manager Survivable SIP Gateway Solution in a Distributed Trunking scenario. It is assumed that the basic configuration of the AudioCodes MP-118 has already been administered. See [11] and [12] 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 Avaya Session Manager Survivable SIP Gateway Solution. In some cases, the parameter values used are specific to the sample configuration and may not apply to all environments.

7.1. 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.

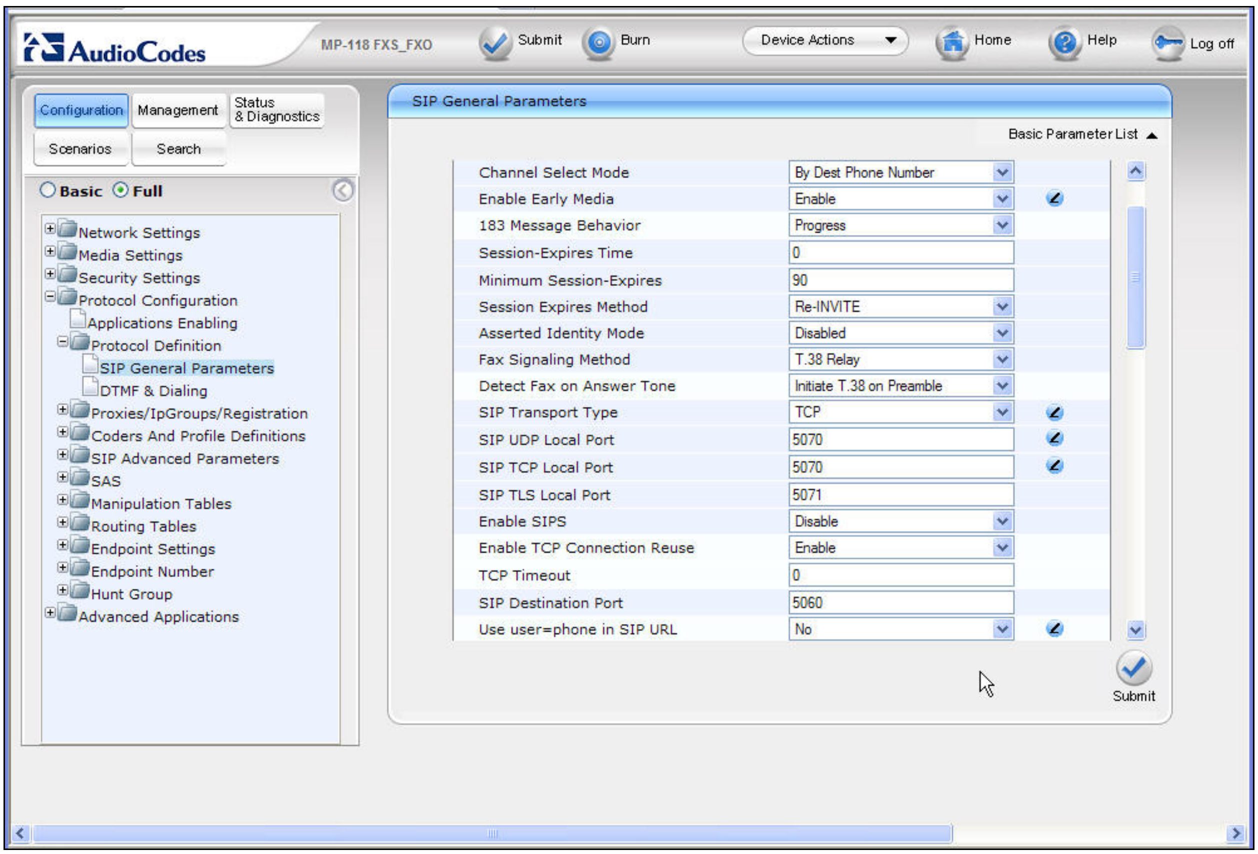
Once logged in, select the **Full** radio button and **Configuration** from the left navigation panel. The example screen below was captured when two calls were up. Each call was between an Avaya 9600 SIP Phone at the branch and an analog FXS port. This is the reason that ports 1 and 2 show green for "RTP Active". The FXO line on port 5 was idle. Other ports were not assigned/used in the sample configuration.



7.2. 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 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 5070 for SIP messages.

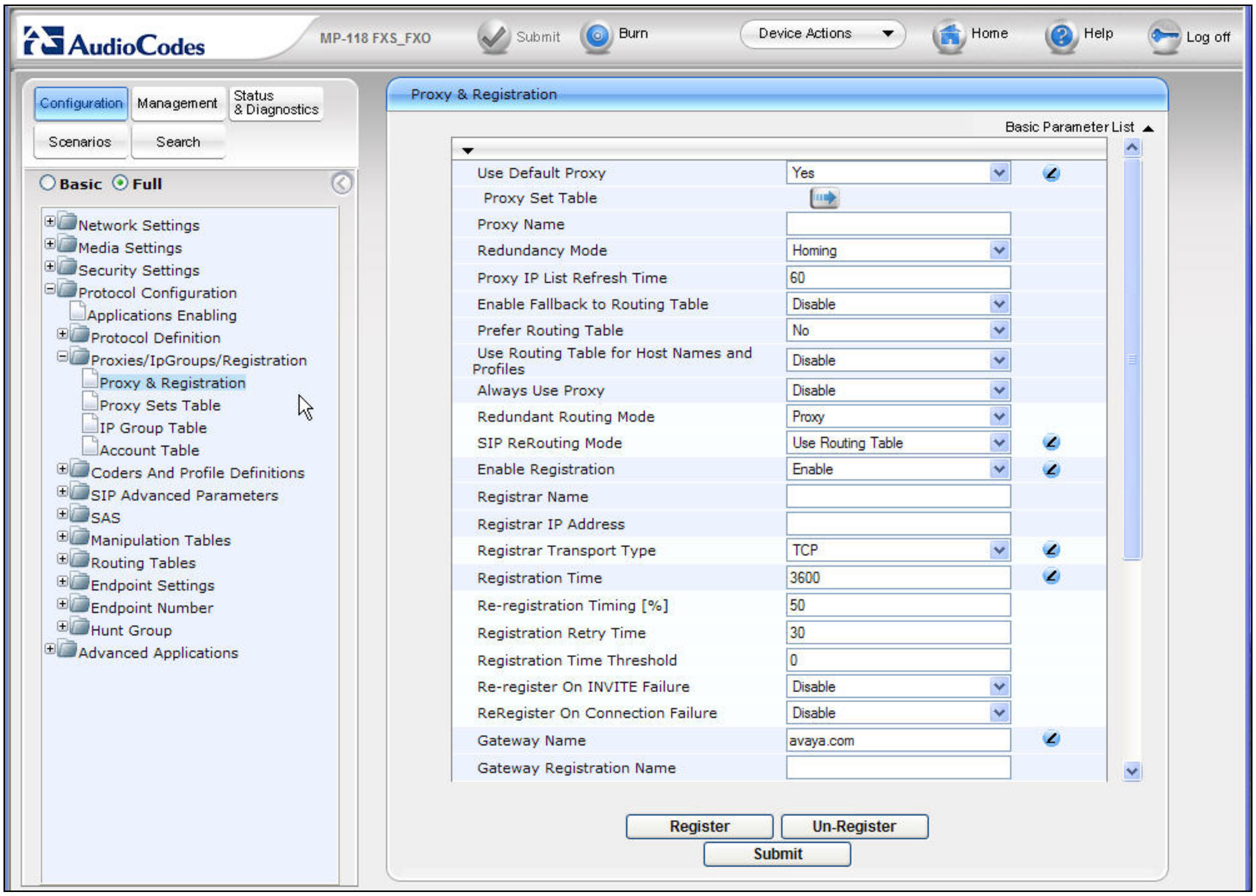


The remaining fields of the SIP General Parameters screens maintain the default values.

7.3. Proxy & Registration

From the left navigation panel, navigate to the Proxy & Registration screen by selecting **Protocol Configuration → Proxies/IpGroups/Registration → Proxy & Registration**. The values of the fields with an adjacent icon have 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 register each FXS station with the SIP registrar using TCP transport, refreshing every 3600 seconds.



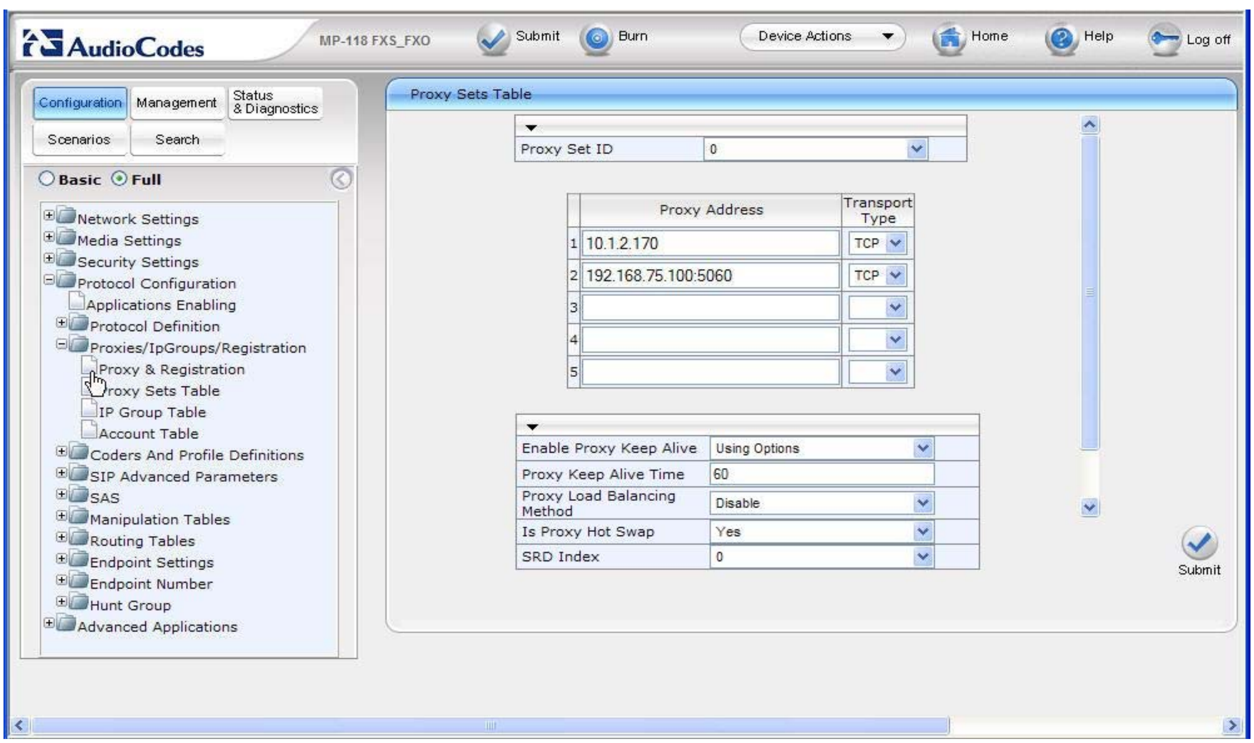
7.4. Proxy Sets Table

From the left navigation panel, navigate to the Proxy Sets Table screen by selecting **Protocol Configuration → Proxies/IpGroups/Registration → Proxy Sets Table**. The values of the fields with an adjacent  icon have changed from the default.

The Proxy Sets Table specifies the SIP Proxy server the AudioCodes MP-118 is going to monitor for connectivity health to determine when to become active as a Survivability Server. In this case, the SIP Proxy server is the Session Manager with IP 10.1.2.170. The Proxy Sets Table also contains an entry specifying the Survivability Server (the AudioCodes MP-118 itself) with IP 192.168.75.100.

The mechanism used to monitor the Session Manager is also specified. SIP Options is used in the sample configuration with the AudioCodes MP-118 default Proxy Keep Alive Time of 60 seconds. This results in the AudioCodes MP-118 sending SIP Options messages to the Session Manager and using the response as an acknowledgement that the Session Manager is accessible from the branch location. If a response to a SIP Options message is not received, the AudioCodes MP-118 will continue to attempt to contact the Session Manager for 60 seconds, the Proxy Keep Alive Time value, and then activate its SAS survivable SIP server feature.

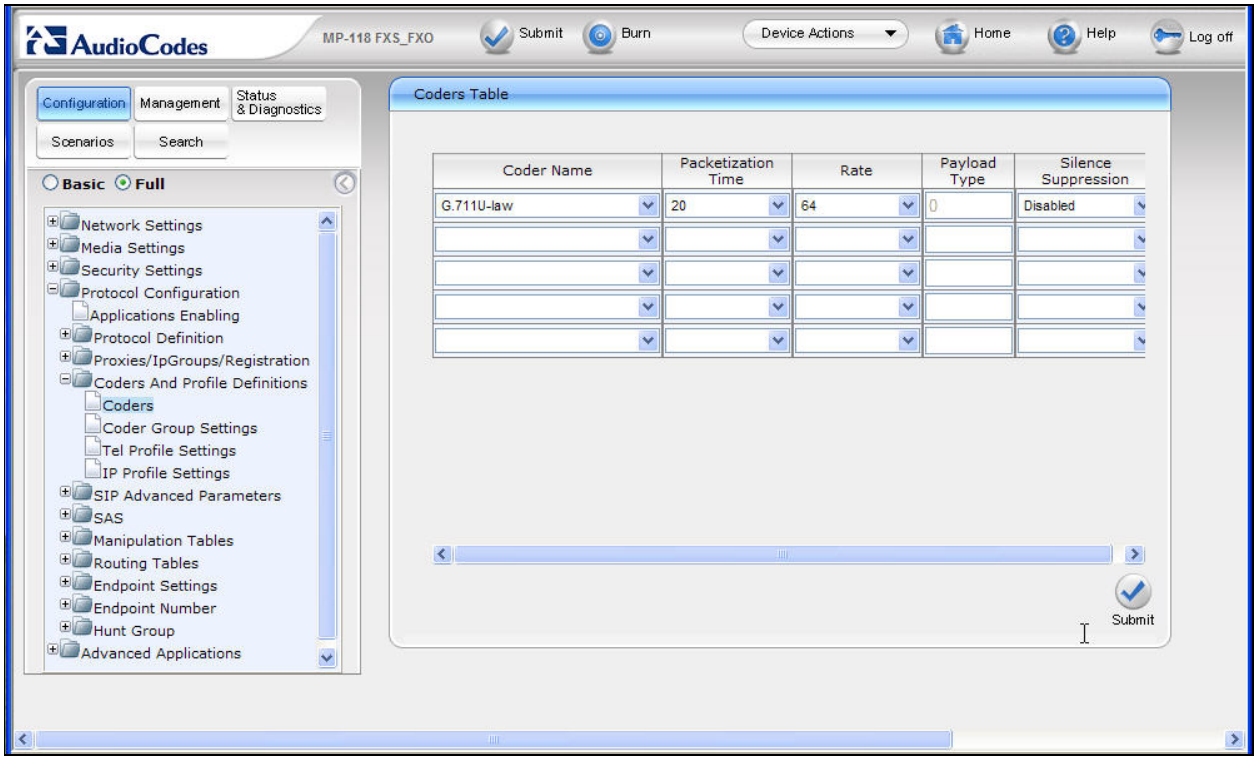
Enter the IP addresses of the Session Manager and the AudioCodes MP-118 in the **Proxy Address** table as shown below. Select TCP from the **Transport Type** drop-down list for both entries. For **Enable Proxy Keep Alive**, select “Using Options” from the drop-down list. Select “Yes” for **Is Proxy Hot Swap**.



7.5. Coders Table

From the left navigation panel, navigate to the Coders Table screen by selecting **Protocol Configuration → Coders And Profile Definitions → Coders**.

Select the codec from the drop-down list that matches the codec configured on Communication Manager (see **Section 4.4**).



7.6. DTMF & Dialing

From the left navigation panel, navigate to the DTMF & Dialing screen by selecting **Protocol Configuration → Protocol Definition → DTMF & Dialing**. The values of the fields with an adjacent  icon have changed from the default.

The value of the **RFC 2833 Payload Type** field must match the value configured for **Telephone Event Payload Type** for the Communication Manager SIP Trunks (see **Section 4.8.2**).

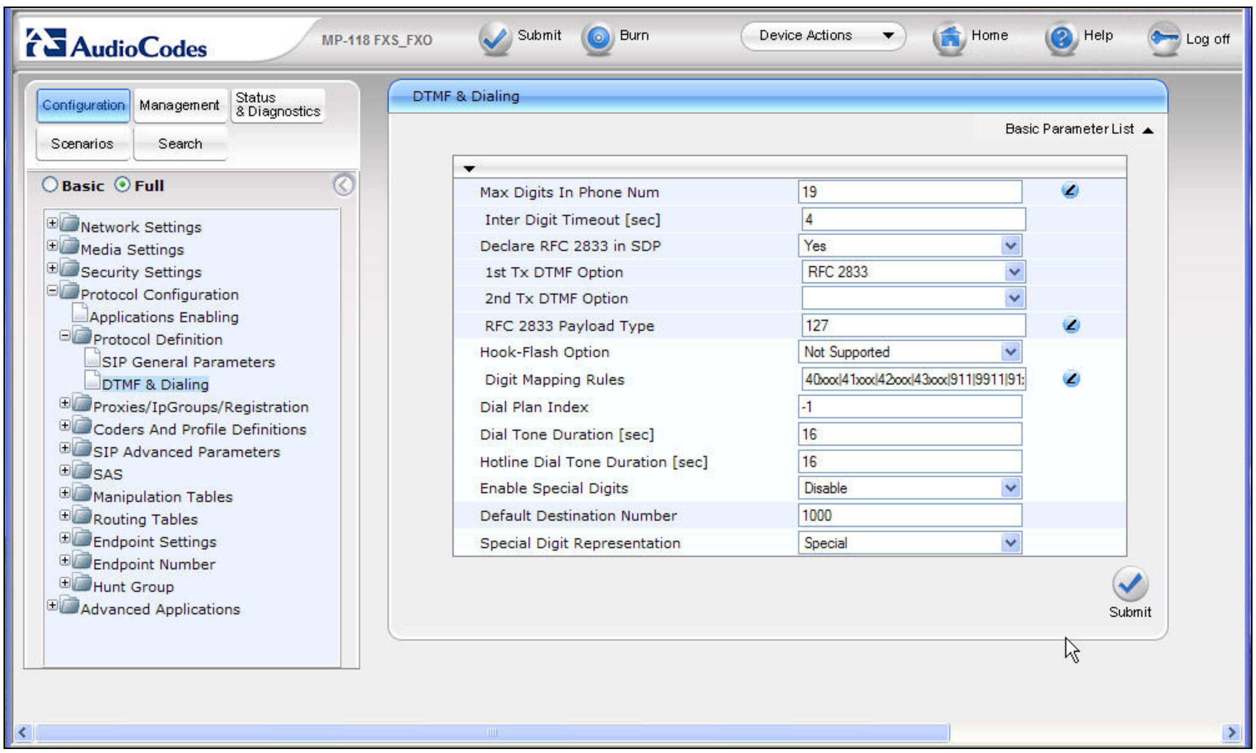
Because the full value of the **Digit Mapping Rules** field is not viewable in the screenshot, the full rule used in the sample configuration for Branch 2 is shown below:

40xxx|41xxx|42xxx|43xxx|911|9911|91xxxxxxxxxx|9011x.T

The details of the Digit Mapping Rule are captured in **Table 2** below. The Digit Mapping Rules setting configured on AudioCodes MP-118 should be consistent with the DIALPLAN setting configured for the Avaya 9600 SIP Phone (see **Section 6**). Refer to [12] for additional information on digit mapping rules.

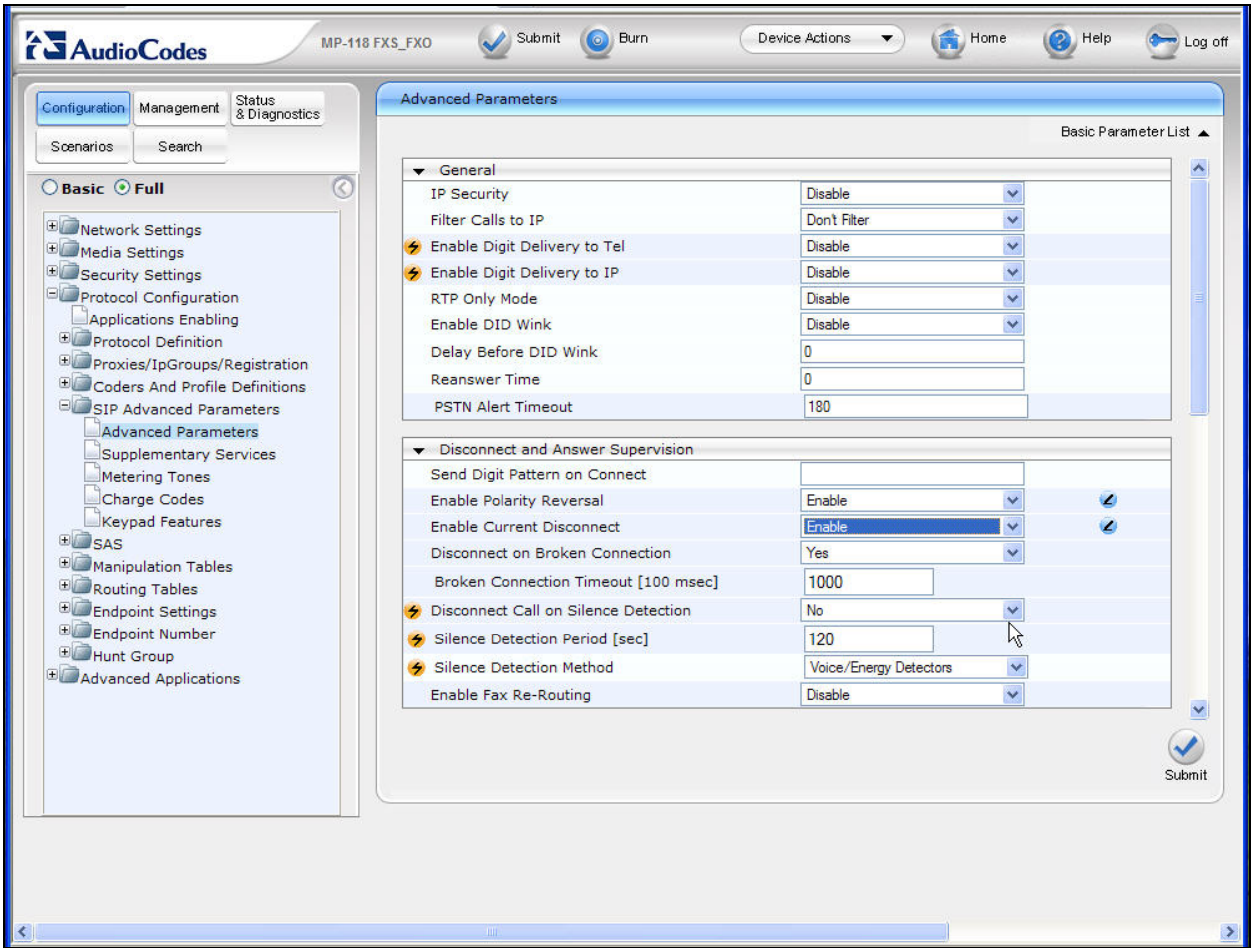
Digit String To Match	Sample Configuration Use
40xxx	HQ extensions
41xxx|42xxx|43xxx	Branch extensions (for Branches 1, 2, and 3)
911|9911	Emergency dialing
91xxxxxxxxxx	North American Numbering Plan
9011x.T	International dialing

Table 2 – Digit Mapping Rule used in Sample Configuration



7.7. Advanced Parameters

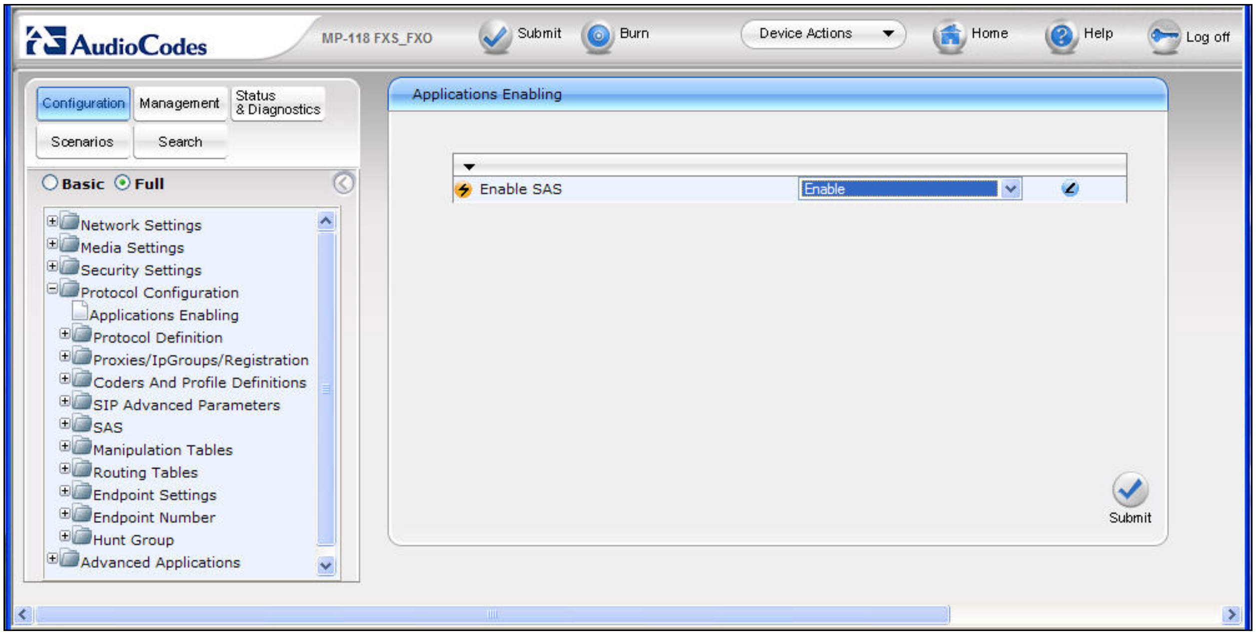
From the left navigation panel, navigate to the Advanced Parameters screen by selecting **Protocol Configuration → SIP Advanced Parameters → Advanced Parameters**. The values of the fields with an adjacent icon have changed from the default.



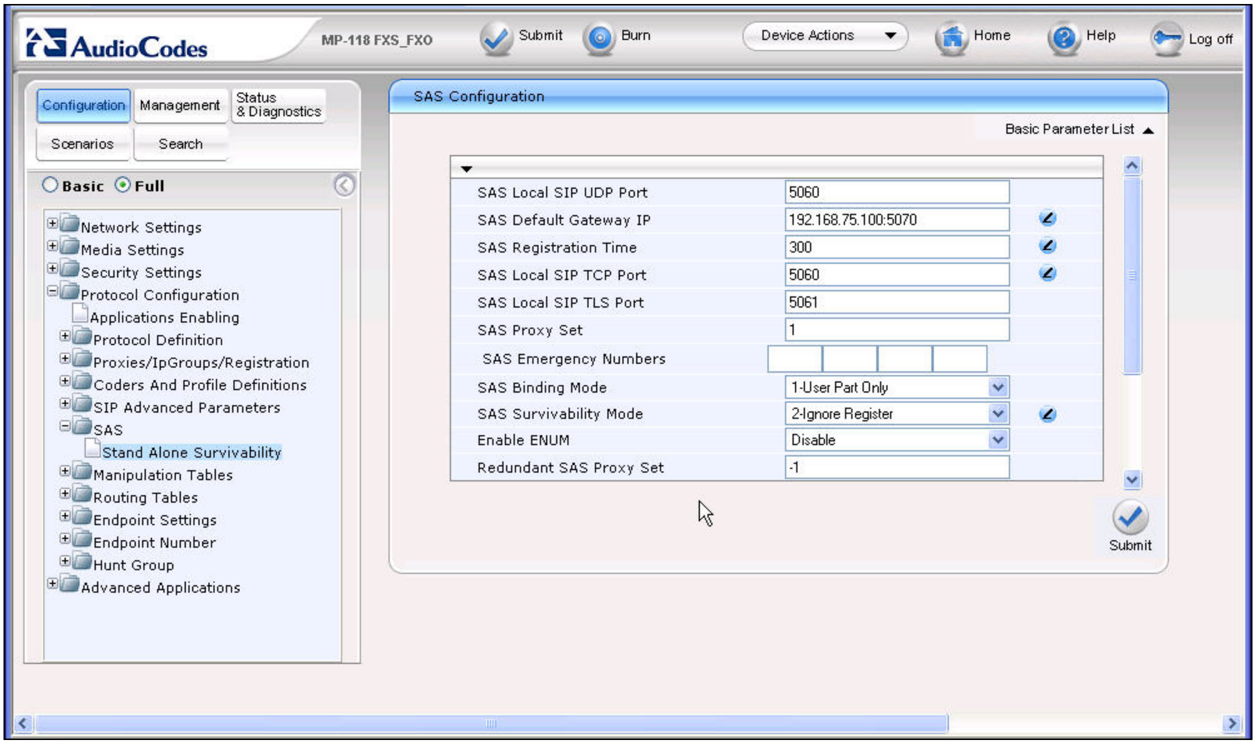
The remaining fields of the SIP General Parameters screens maintain the default values.

7.8. Stand-Alone Survivability

From the left navigation panel, navigate to the Application Enabling screen by selecting **Protocol Configuration → Application Enabling**. Select "Enable" for **Enable SAS**.



From the left navigation panel, navigate to the Stand-Alone Survivability screen by selecting **Protocol Configuration → SAS → Stand-Alone Survivability**. The values of the fields with an adjacent icon have changed from the default. Note the SAS SIP Proxy and Registrar IP address specified for the **SAS Default Gateway IP** field. Also note the selection for **SAS Survivability Mode** (see **Section 7.19.1** for details).

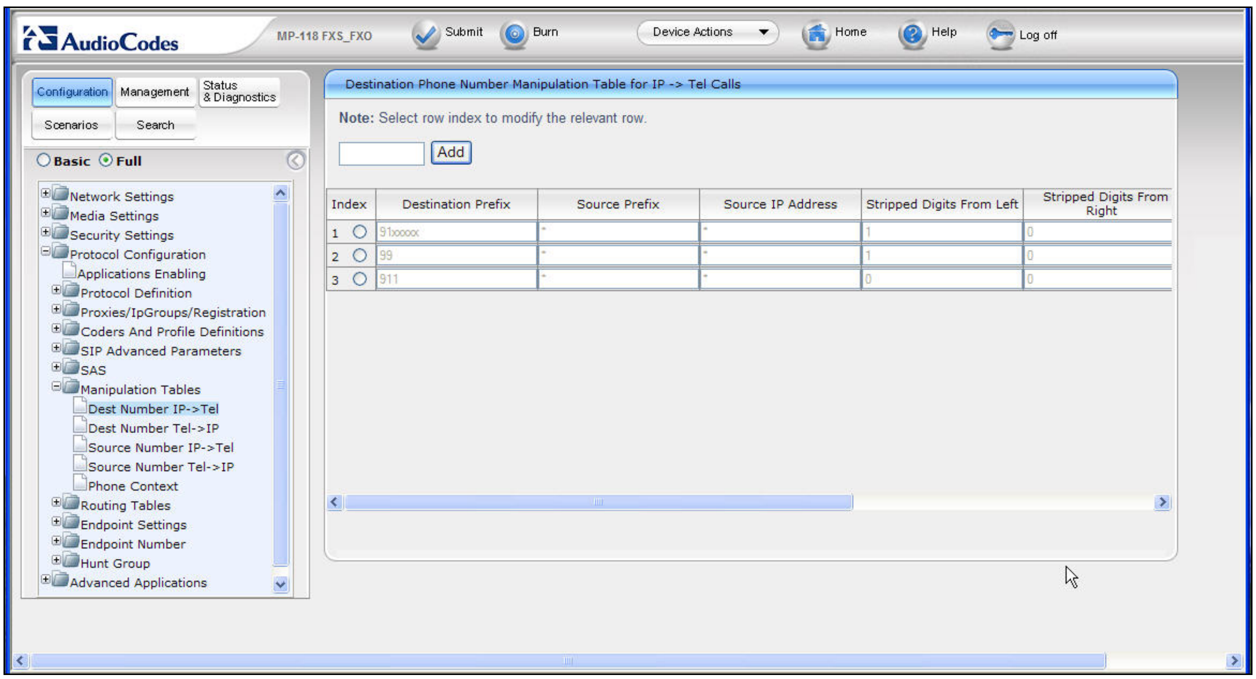


7.9. Dest Number IP → Tel

From the left navigation panel, navigate to **Protocol Configuration → Manipulation Tables → Dest Number IP->Tel**.

The entry in this table strips the leading 9 from the dialed digit strings (for numbers matching the **Destination Prefix**) for IP to PSTN calls while in Survivability Mode. In Normal Mode, this is done by Communication Manager.

As an example, the leading digit "9" would be stripped in the dialed number "9 1-732-555-1111" leaving "1-732-555-1111" presented to the PSTN via the AudioCodes MP-118 FXO interface. Similarly, the dialed emergency number "9 911" would be presented to the PSTN as "911". However, if the user simply dials "911", the AudioCodes MP-118 FXO interface will pass it along to the PSTN as is.



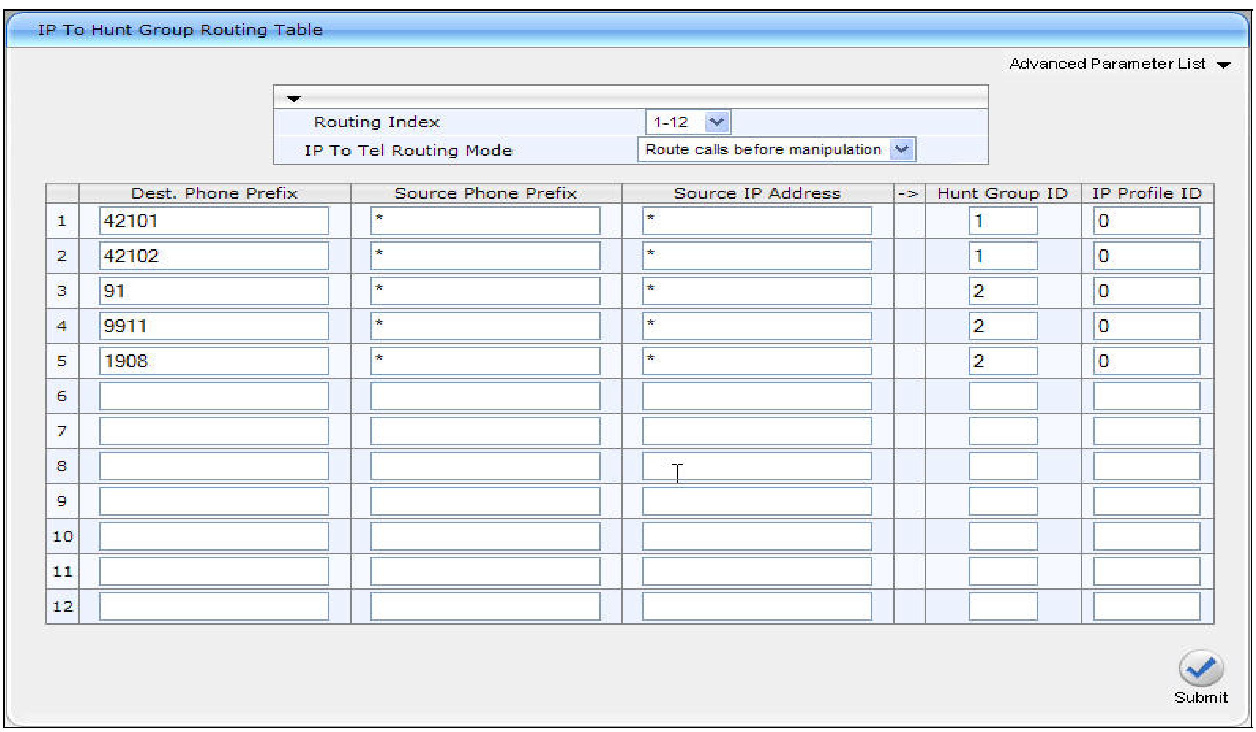
7.10. IP to Hunt Group Routing

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, the FXS analog phone numbers are entered explicitly and route to Hunt Group ID 1. Calls to PSTN starting with "91" (including 911 call and 91xxxxxxxxxx conforming to North American Numbering Plan) as well as 911 call with a PSTN access digit "9" will route to Hunt Group ID 2. Calls routed from Session Manager with the leading digits "1908" are local PSTN calls from branch phones, and therefore routed to Hunt Group ID 2..

Hunt Group ID 1 consists of two FXS interfaces and Hunt Group ID 2 consists of one FXO interface. Hunt Group to Channel assignments are configured in **Section 7.14.** The table below shows a summary of the Hunt Group assignments.

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

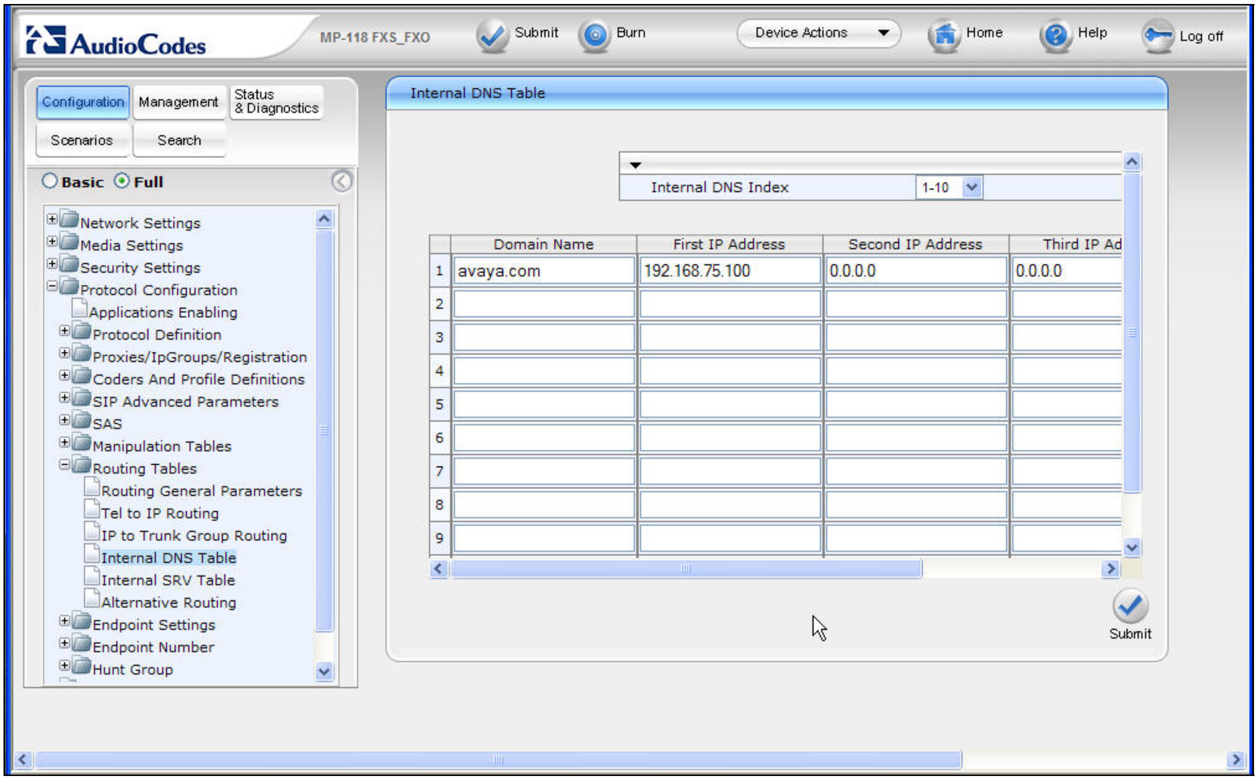


	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Hunt Group ID	IP Profile ID
1	42101	*	*		1	0
2	42102	*	*		1	0
3	91	*	*		2	0
4	9911	*	*		2	0
5	1908	*	*		2	0
6						
7						
8						
9						
10						
11						
12						

7.11. Internal DNS Table

From the left navigation panel, navigate to the Internal DNS Table screen by selecting **Protocol Configuration → Routing Tables → Internal DNS Table**.

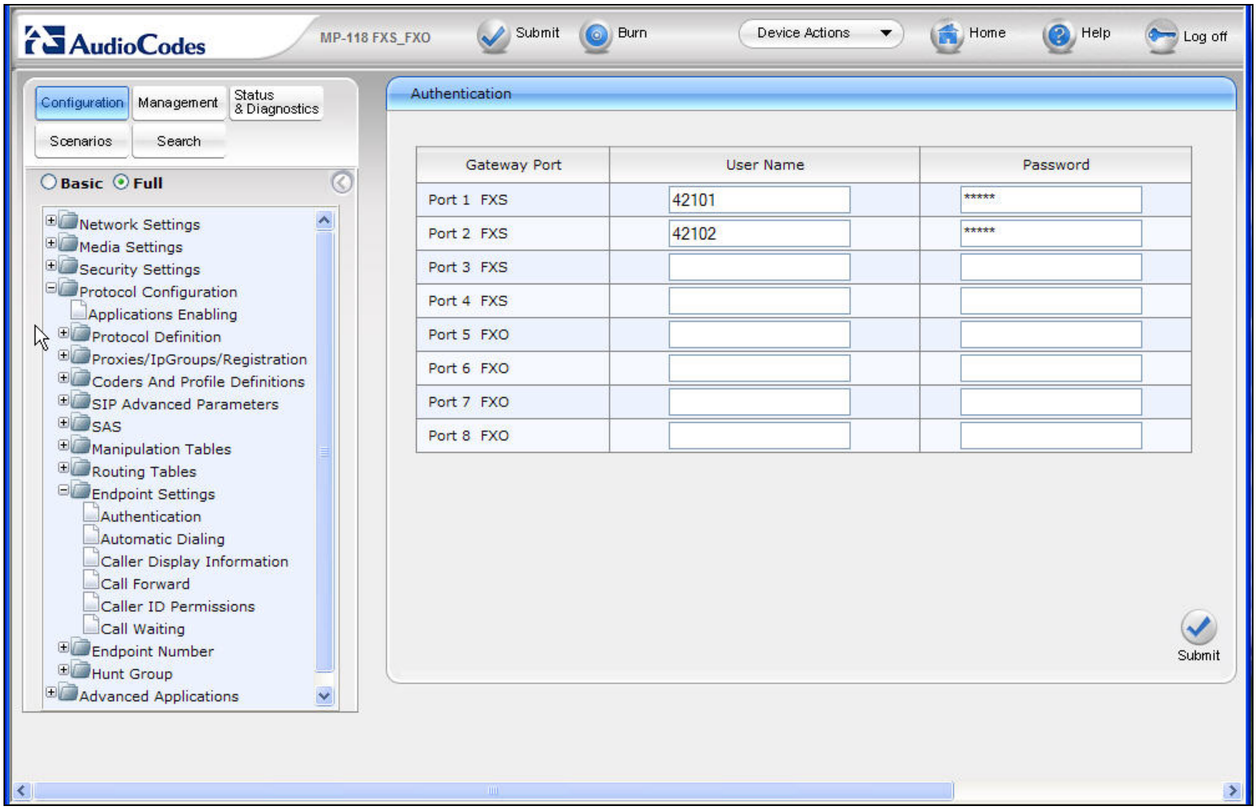
Enter the SIP domain and the IP address of the on-site branch AudioCodes MP-118 in the first table entry. Enter "0.0.0.0" for **Second IP Address**, **Third IP Address**, and **Fourth IP Address** (not shown)..



7.12. Authentication

From the left navigation panel, navigate to the Authentication screen by selecting **Protocol Configuration → Endpoint Settings → Authentication**.

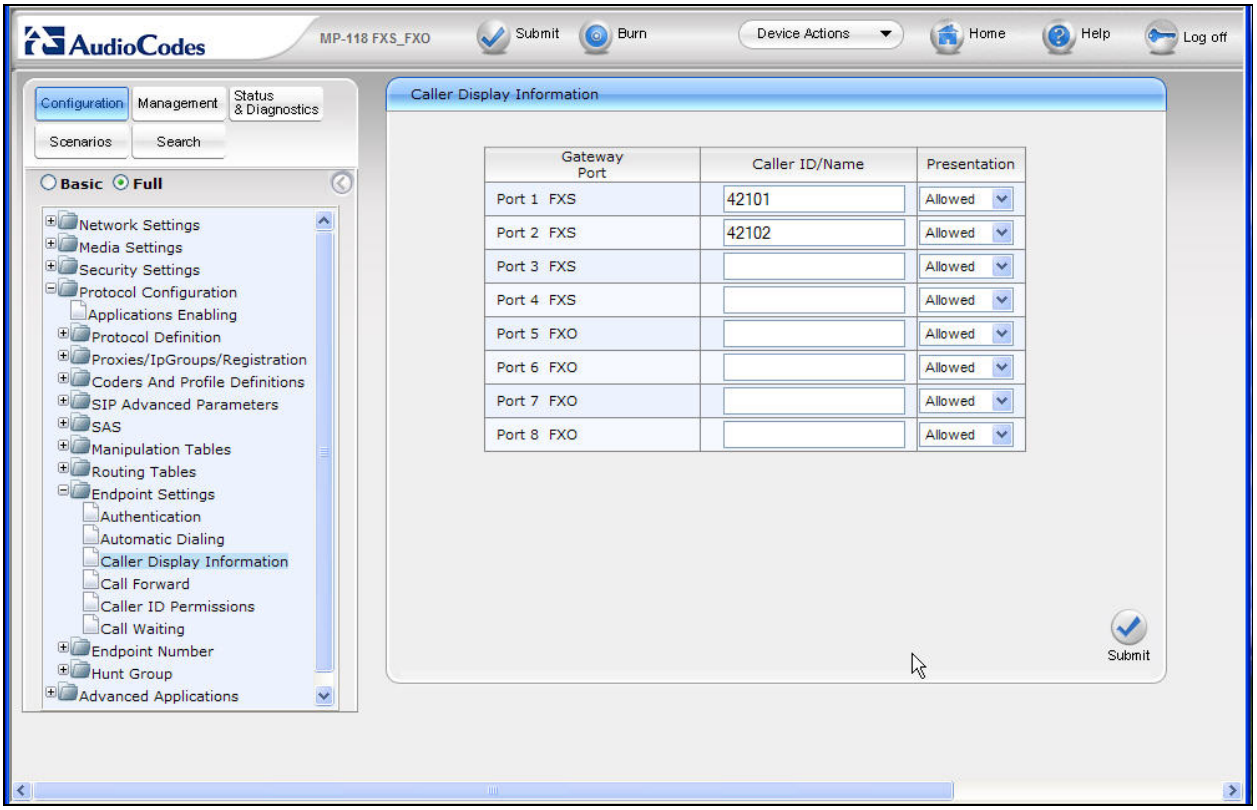
Enter the SIP user name and password that match the AudioCodes MP-118 FXS Analog Phone User Account created on Session Manager in **Section 5.10**.



7.13. Caller Display Information

From the left navigation panel, navigate to the Caller Display Information screen by selecting **Protocol Configuration → Endpoint Settings → Caller Display Information**.

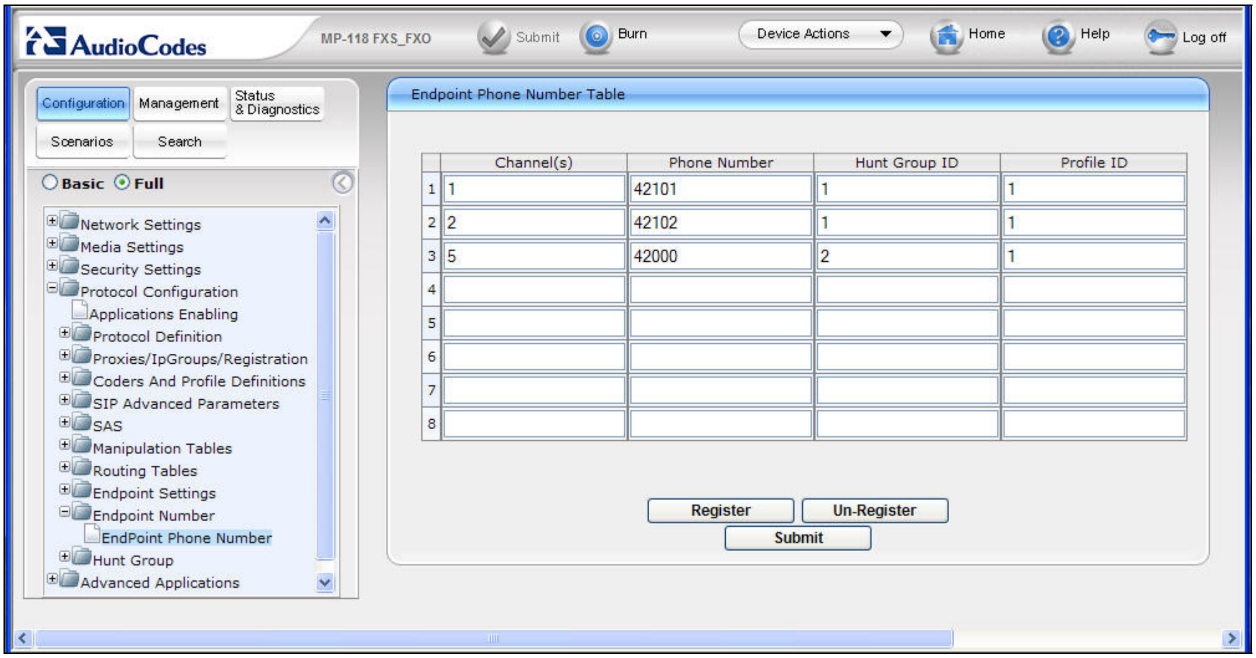
Enter the name/number to be displayed on the called station in Survivable Mode for each interface. The FXS extension numbers are used in the sample configuration. In Normal Mode, the display information is controlled by the name and number configuration in Communication Manager.



7.14. Endpoint Phone Number

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 4 are the FXS interfaces; Channels 5 through 8 are the FXO interfaces. The sample configuration used Channels 1, 2 (FXS) and 5 (FXO) only.



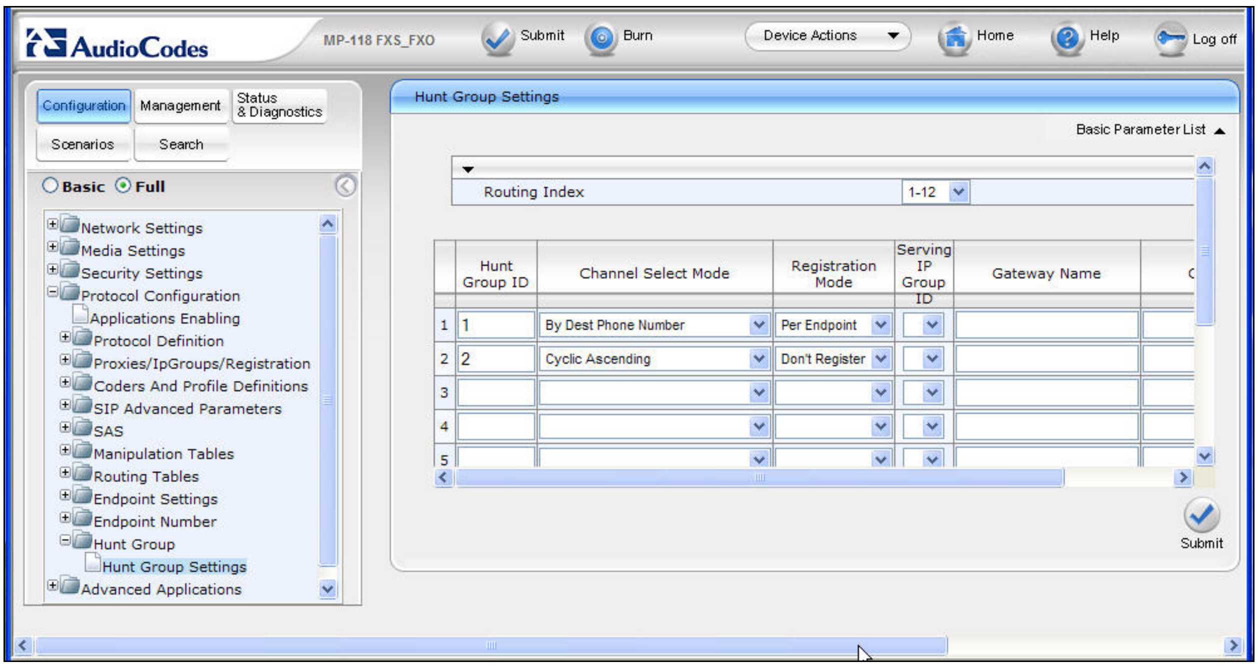
7.15. Hunt Group Settings

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.

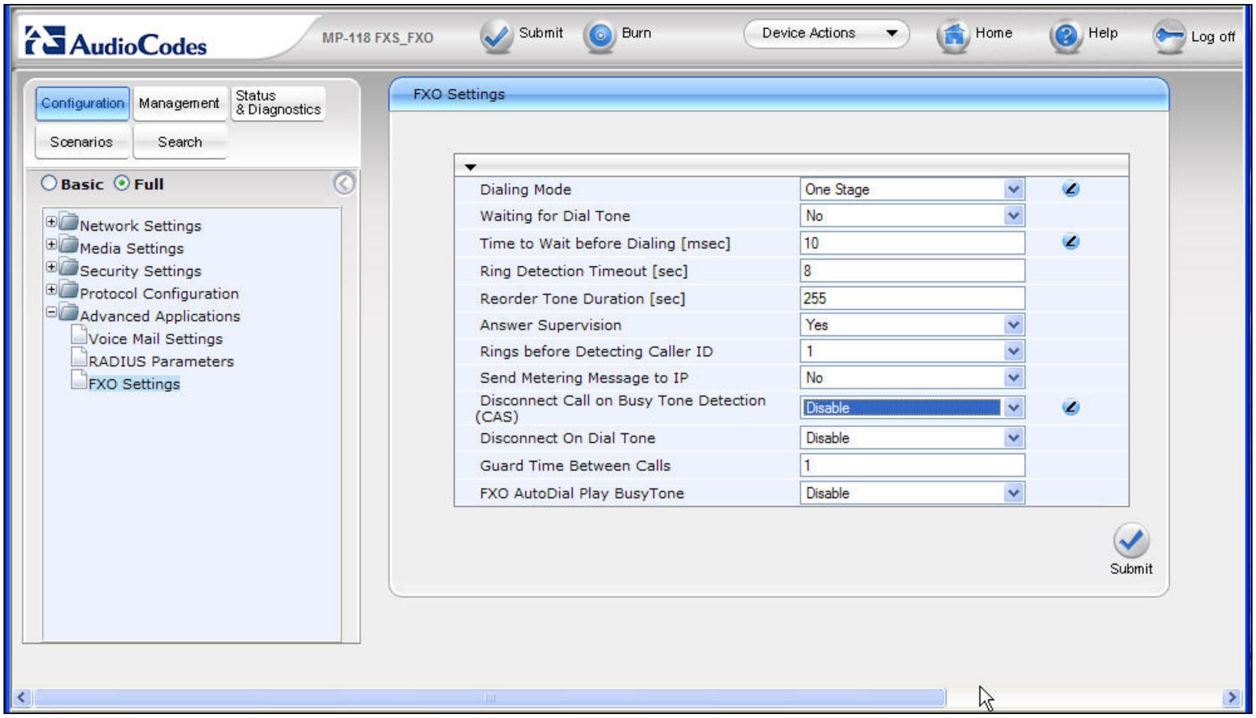
Hunt Group 1, containing 2 FXS interfaces for analog phones, is configured to select the proper FXS interface to terminate calls based on the destination phone number.

Hunt Group 2, containing 1 FXO interface to the PSTN, is configured to select any interface in this Hunt Group in a Cyclic Ascending order. Cyclic Ascending is the default. Since only one FXO interface is configured for Hunt Group 2 in the sample configuration, no channel cycling is occurring.



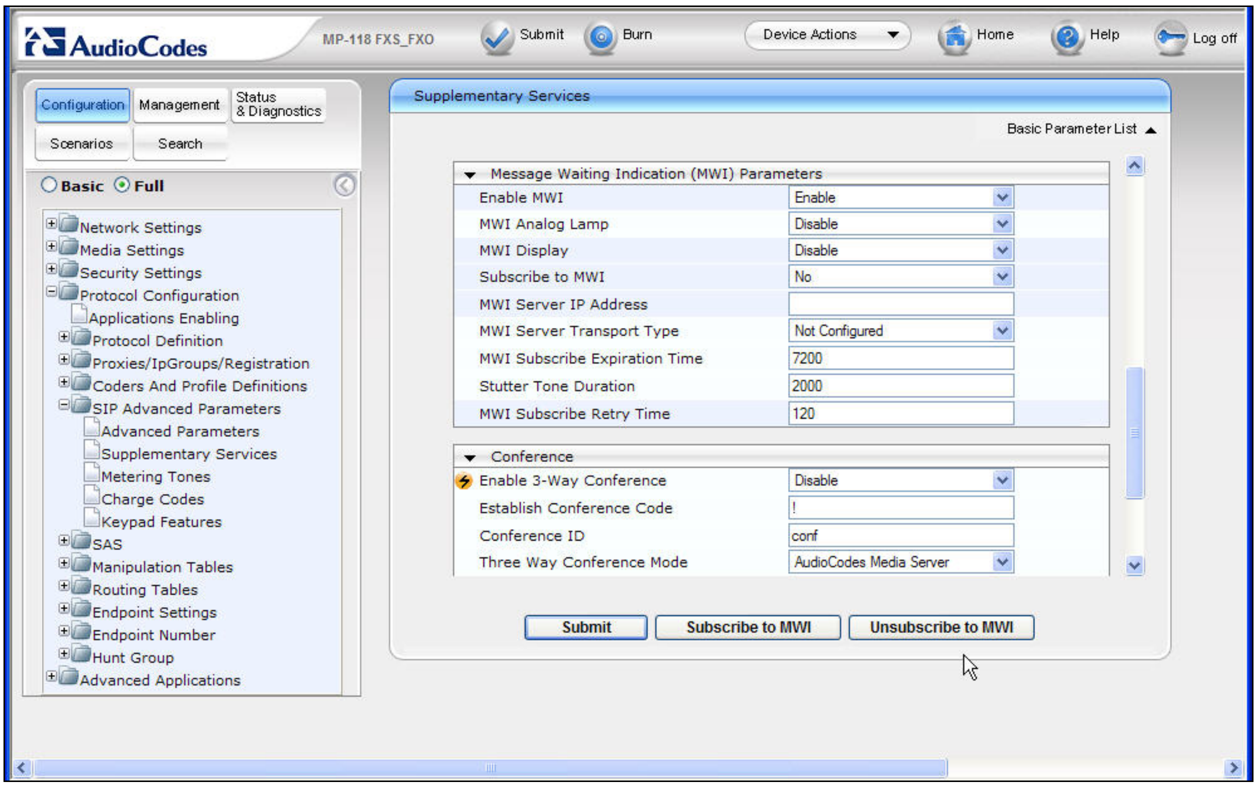
7.16. Advanced Applications → FXO Settings

From the left navigation panel, navigate to the FXO Settings screen by selecting **Advanced Applications → FXO Settings**. The values of the fields with an adjacent  icon have changed from the default.



7.17. Message Waiting Indication via Stutter Dial Tone for Analog FXS

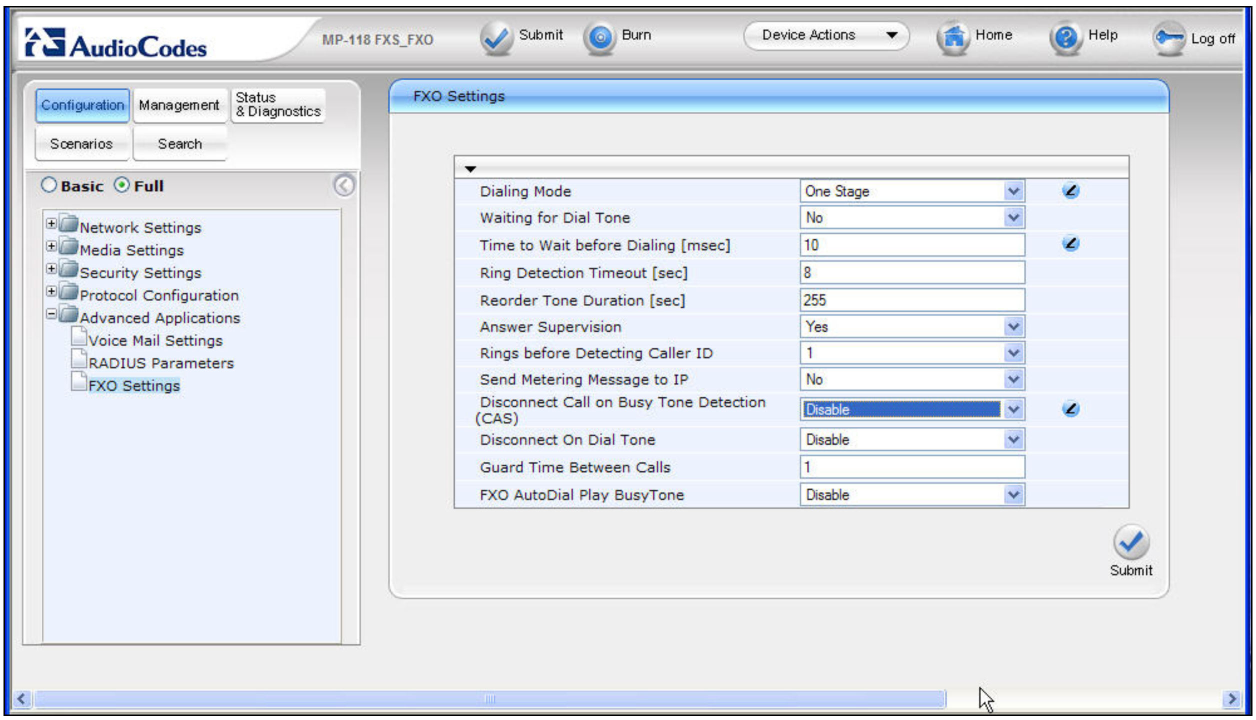
To enable analog stations connected to the FXS ports to receive stutter dial tone for audible message waiting notification, navigate to **Protocol Configuration → SIP Advanced Parameters → Supplementary Services**. Verify that "Enable" from the **Enable MWI** drop-down is selected, as shown in the following screen. When a SIP user registers, or the message waiting status of a registered user changes, Session Manager will send SIP NOTIFY messages to update the message waiting status. The AudioCodes Gateway can process these NOTIFY messages, and provide normal dial tone to the FXS ports when there is no message waiting, and stutter dial tone when there is a message waiting (e.g., a new message in a Communication Manager Messaging or Avaya Modular Messaging mailbox). It is not necessary that the AudioCodes Gateway subscribe to MWI, but this option (**Subscribe to MWI**) is available. Observe that **Stutter Tone Duration** can also be configured.



7.18. Disable FXO Disconnect on Busy Tone Detection (Optional)

The AudioCodes Gateway can automatically detect when a call is connected to busy tone from the PSTN on an FXO line, and disconnect the call if desired. For the sample configuration, it is recommended that this feature be disabled. If the feature remains enabled, and an Avaya SIP Telephone in the branch makes a call to a PSTN number (in Survivable Mode) that is busy (e.g., a standard home telephone that is in use with no call waiting and no voice mail), the Avaya SIP Telephone will hear busy tone for a few seconds, and then the call appearance will be cleared. Although this frees the FXO more quickly, it may be perceived by the telephone user as a problem with the system. With the feature disabled as shown below, the Avaya SIP Telephone would simply hear busy tone until hanging up the telephone.

Navigate to **Advanced Applications → FXO Settings**. Use the drop-down menu to select "Disable" for the **Disconnect Call on Busy Tone Detection (CAS)** parameter.



7.19. .ini File

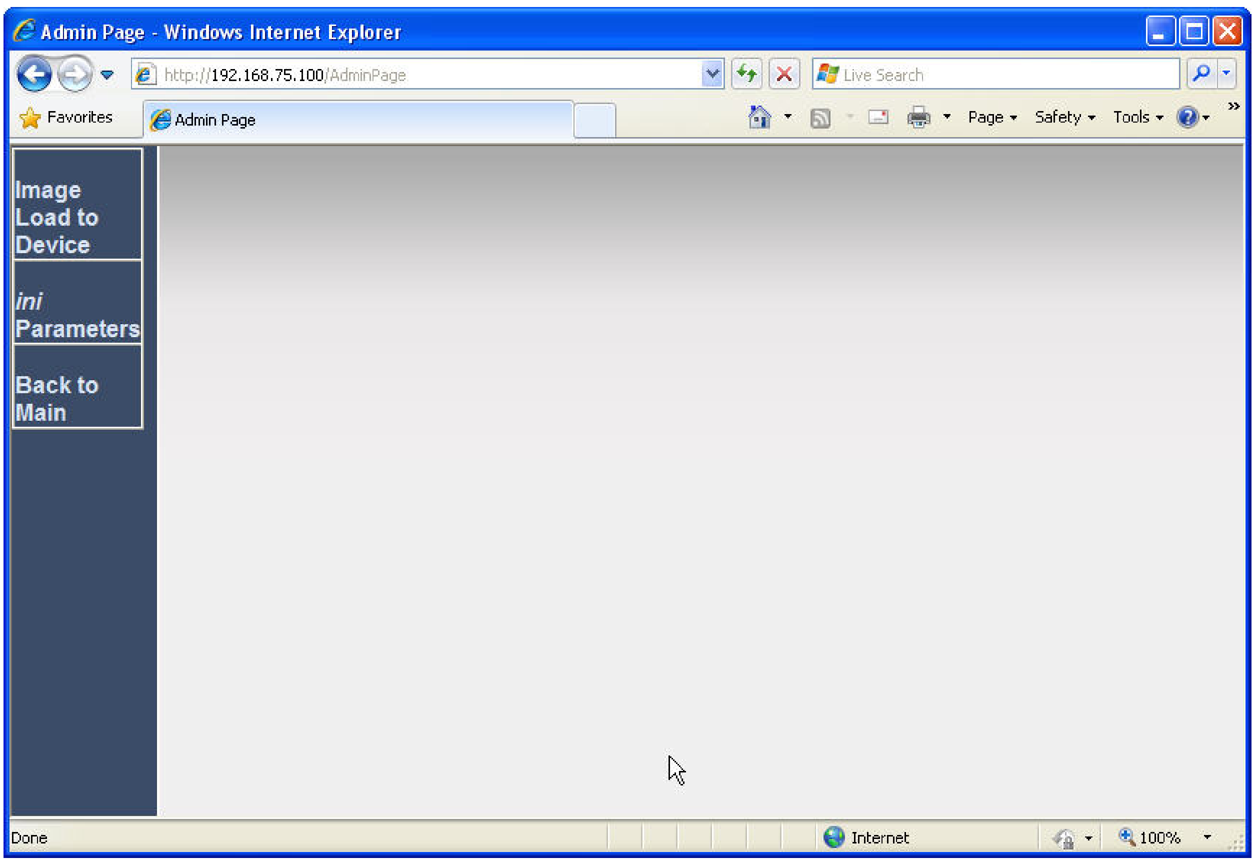
The AudioCodes MP-118 utilizes an initialization text file with a .ini extension. The .ini file contains MP-118 parameters that have been set by the WebUI, such as the parameters described in the previous sections. See [12] for additional information about the ini configuration file.

For the AudioCodes MP-118 firmware version listed in **Table 1**, the following parameters are not configurable from the WebUI and must be modified directly in the .ini file.

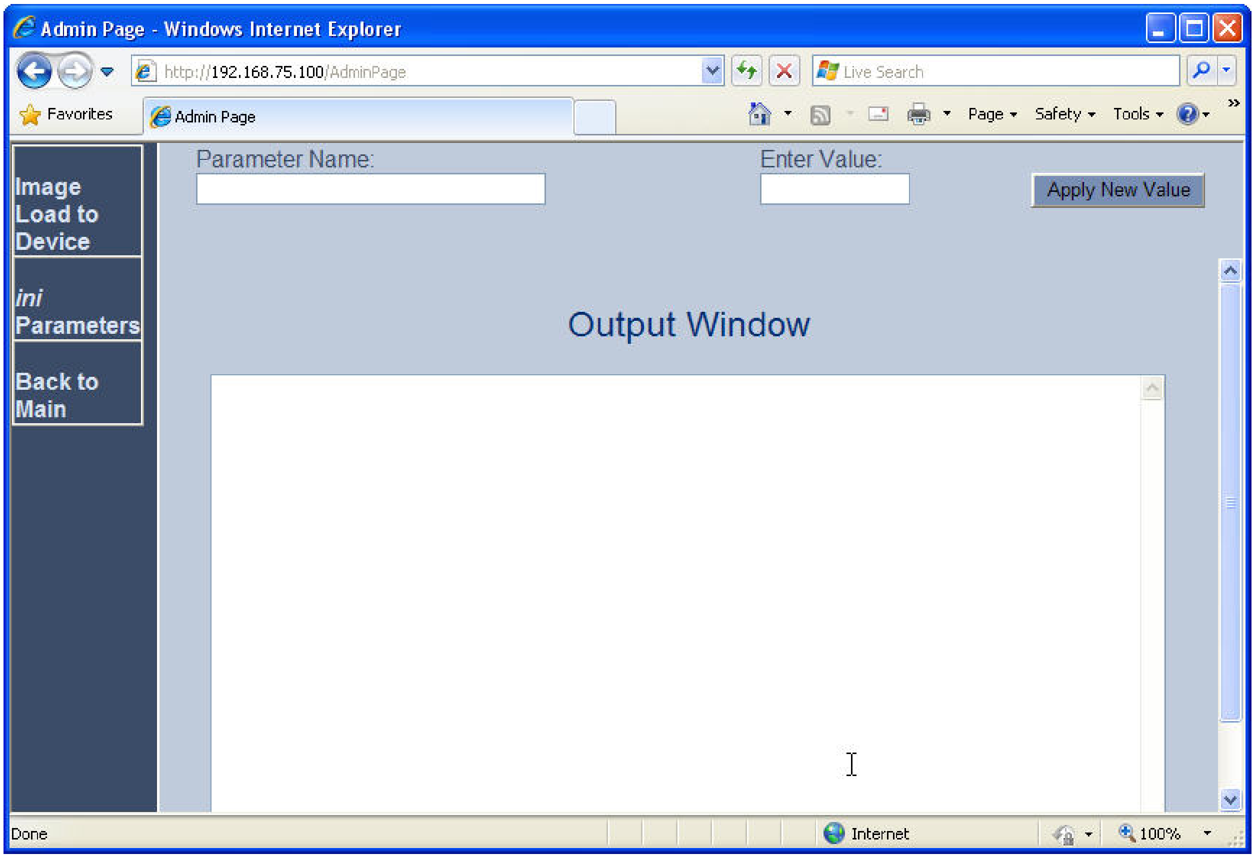
- ReliableConnectionPersistentMode
- CurrentDisconnectDuration

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.



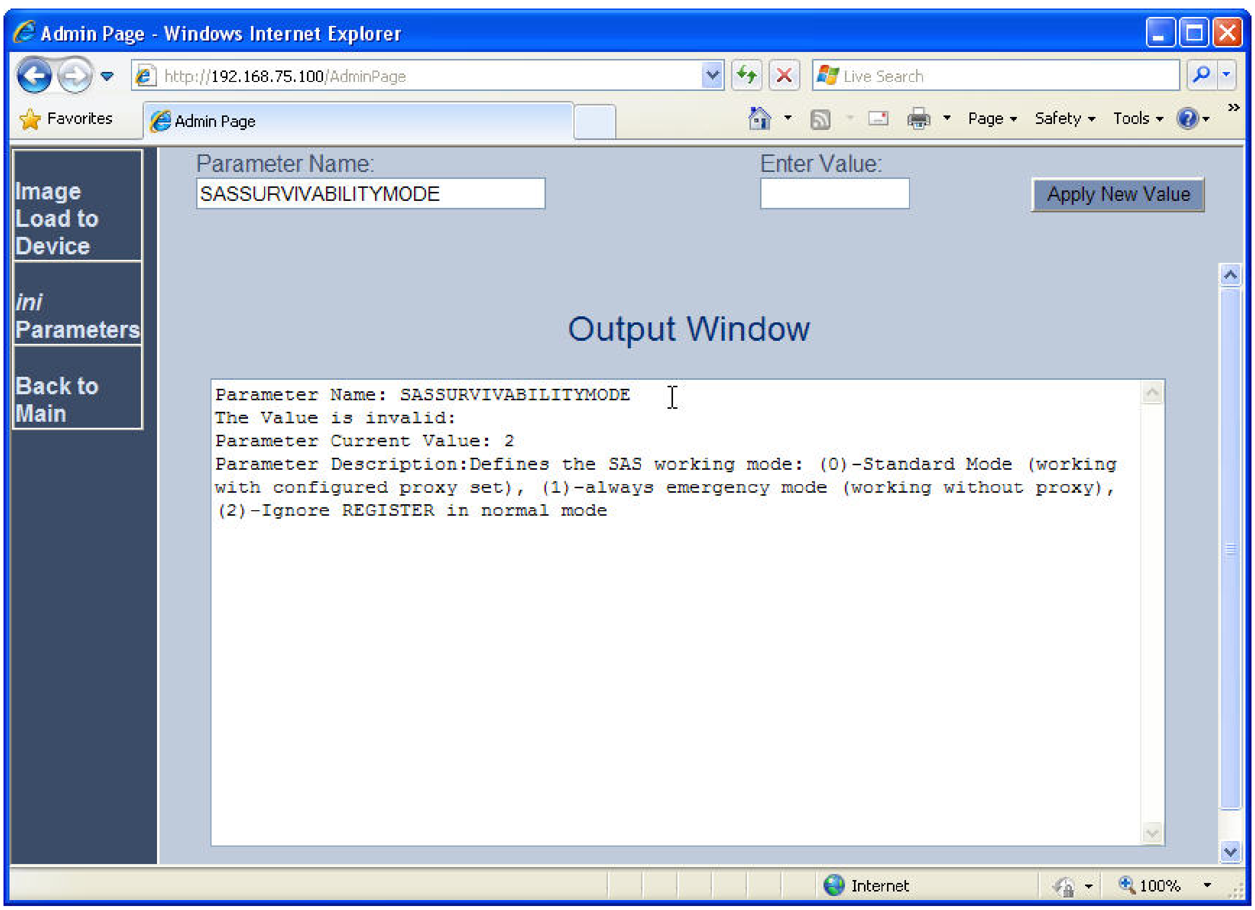
7.19.1. SASSurvivabilityMode

The **SASSurvivabilityMode** parameter is accessible from **Configuration→ Protocol Configuration→ SAS→ Stand Alone Survivability** of the MP-118 web administrative interface. This important setting is included here as a verification point.

The **SASSurvivabilityMode** parameter determines how the SAS feature of the AudioCodes MP-118 will operate. By default, **SASSurvivabilityMode** is set to a value of 0 which enables SAS to be able to accept SIP Registrations while the AudioCodes MP-118 can simultaneously communicate with Session Manager.

SASSurvivabilityMode *must be changed from the default value of 0 to a value of 2.* This sets SAS to become active and only accept SIP Registrations when it is not able to communicate with Session Manager.

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. The screen below shows that the **SASSurvivabilityMode** parameter is currently set to the required value of 2 as previously administered.



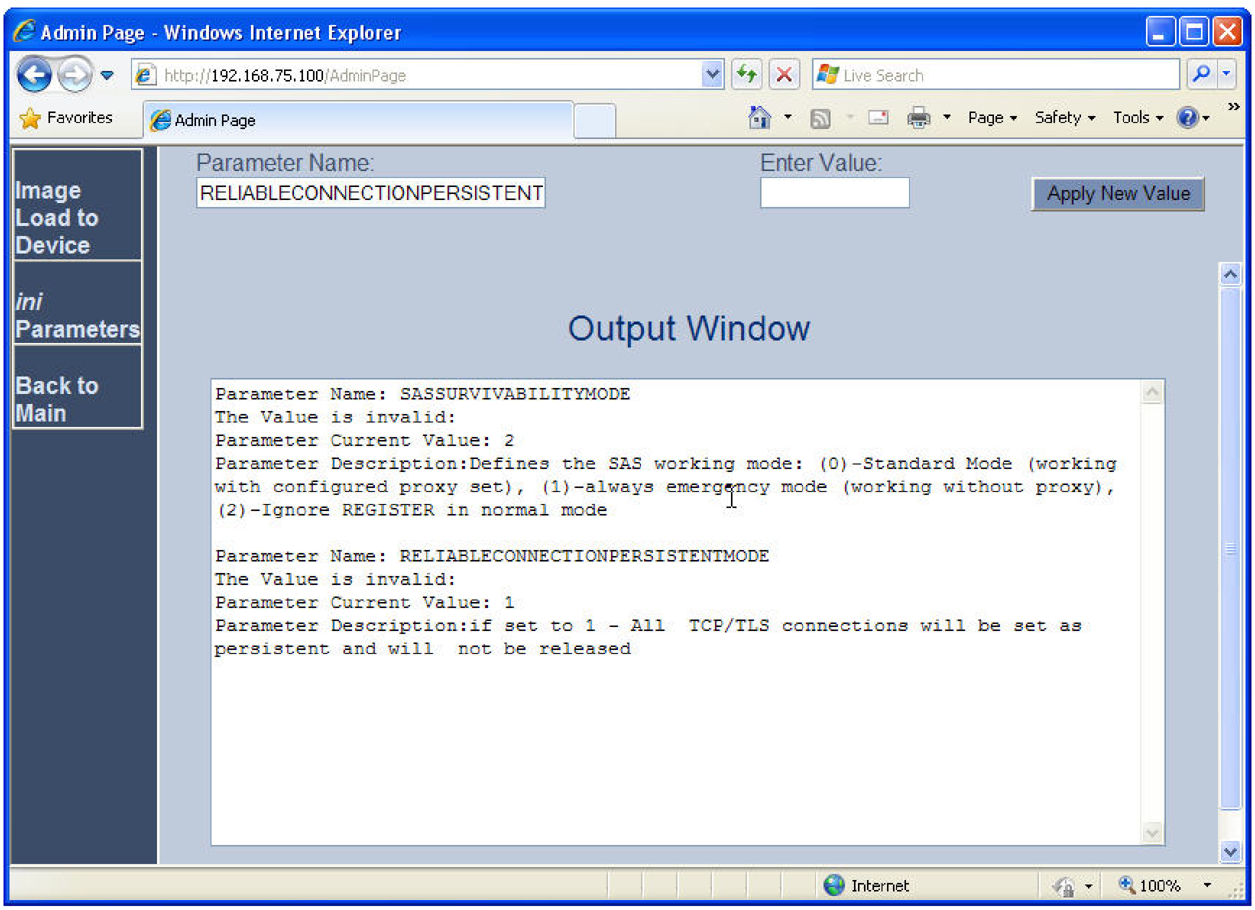
To change the value of a parameter, enter the new parameter value in the “Enter Value” field, then click the adjacent “Apply New Value” button. The resulting screen will show both the old and new settings.

7.19.2. ReliableConnectionPersistentMode

The **ReliableConnectionPersistentMode** parameter determines how the AudioCodes MP-118 establishes TCP connections. When **ReliableConnectionPersistentMode** is set to the default value of 0, all TCP/TLS 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 value of the **ReliableConnectionPersistentMode** parameter is currently set to the required value of 1 as previously administered.



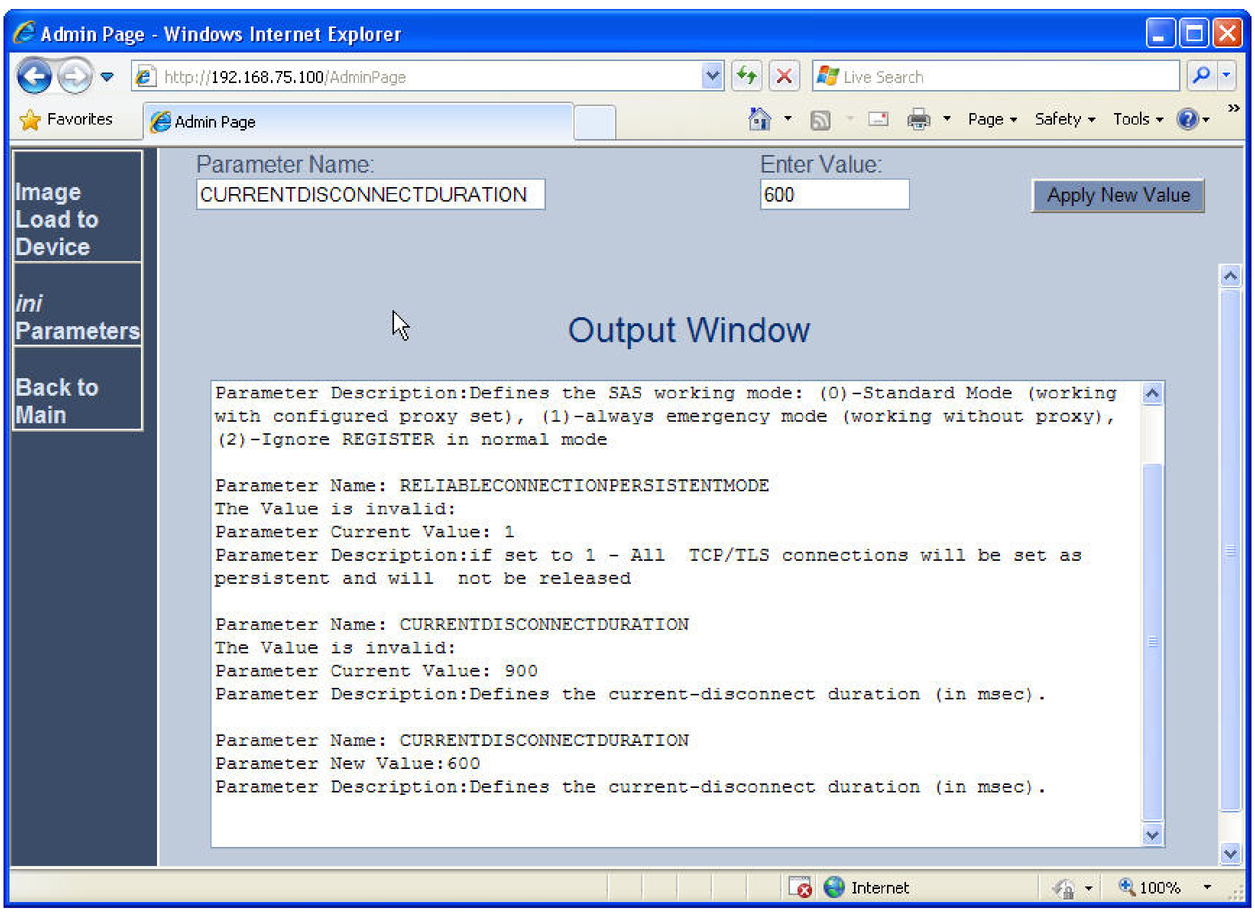
7.19.3. CurrentDisconnectDuration

The **CurrentDisconnectDuration** parameter determines the duration of time in milliseconds the analog line current is dropped indicating a disconnect pulse to the AudioCodes MP-118 FXO interfaces. For the sample configuration, this parameter was changed from the default value of 900ms to 600ms. This was required to obtain a proper disconnect on the AudioCodes MP-118 FXO Analog Trunk from the PSTN service provider.

Note: The need to change **CurrentDisconnectDuration** may not apply to all environments and will be determined by the PSTN service provider configuration of the analog trunk.

Also, the parameters **EnableReversalPolarity** and **EnableCurrentDisconnect** must both be enabled for **CurrentDisconnectDuration** to be active. The **EnableReversalPolarity** and **EnableCurrentDisconnect** parameters are both configured on the Advanced Parameters screen as shown in **Section 7.7**.

The following screen shows the value of the **CurrentDisconnectDuration** parameter was successfully set to a value of 600.



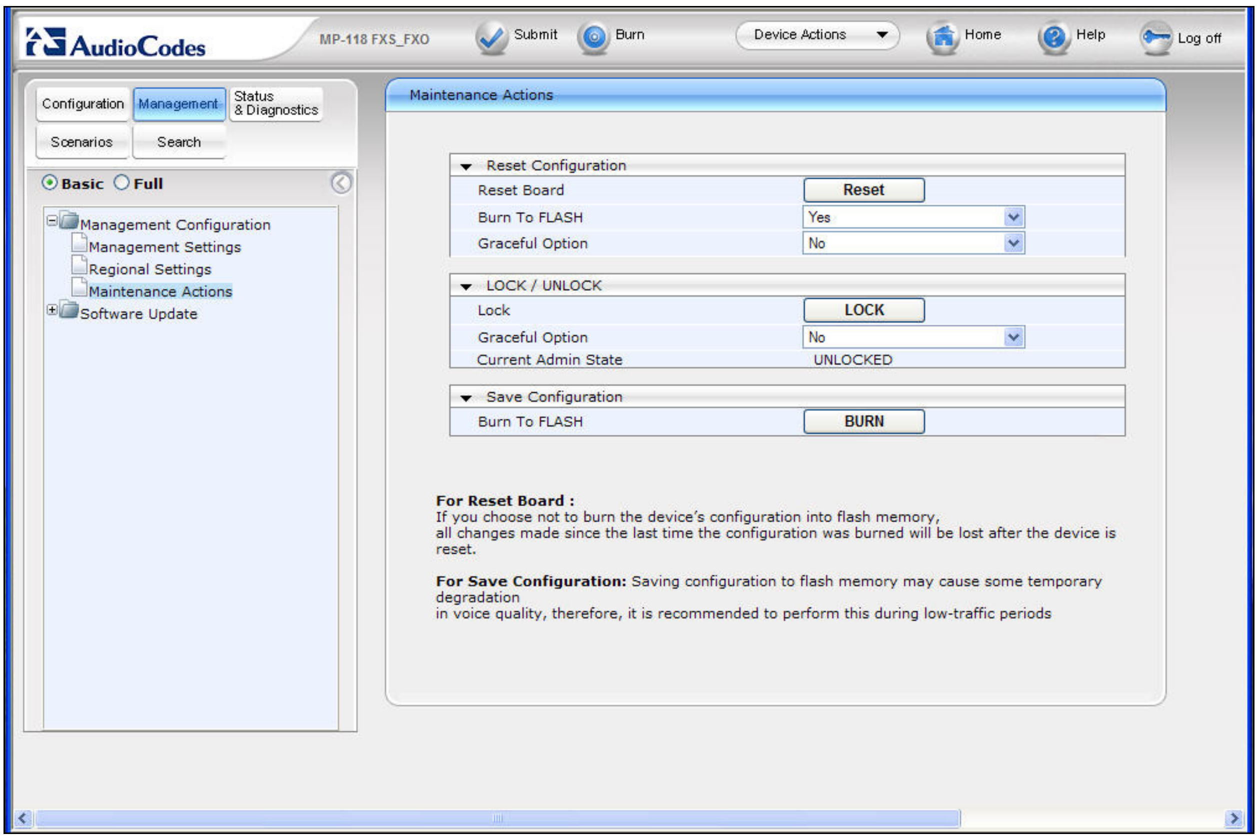
7.20. Saving Changes to the AudioCodes Gateway

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.



8. General Test Approach and Test Results

This section describes the testing used to verify the sample configuration for the Avaya Session Manager Survivable SIP Gateway Solution using the AudioCodes MP-118 Media Gateway in a Distributed Trunking scenario. This section covers the general test approach and the test results.

8.1. General Test Approach

The general test approach was to break and restore network connectivity from the branch site to the headquarters location to verify that

- When network connectivity is broken, the branch AudioCodes MP-118 gateway automatically assumes the SIP proxy and SIP registrar functions. In this Survivable Mode, the branch phones can still call each other and reach PSTN through the AudioCodes MP-118 FXO trunk interface.
- When network connectivity is restored, SIP proxy and registrar functions are automatically switched back to the Session Manager at the headquarters location for providing centralized SIP call control. In this Normal Mode, PSTN access by phones at both the headquarters and branch sites are through the T1/E1 connection on the Avaya Media Gateway at the central location with the exception that local non-toll calls from the branch phones are routed to the PSTN through the branch AudioCodes MP-118.

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:

- In Normal Mode, branch phones register to the Session Manager located at the central site; in Survivable Mode, branch phones register to the AudioCodes MP-118 located at the branch location.
- Switching between the Normal and the Survivable Modes is automatic and within a reasonable time span (within one to 2 minutes).
- In Normal Mode, calls can be placed between phones at the main site and the branch site, and among phones within the site.
- In Normal Mode, local non-toll calls from the branch phones are routed to the PSTN through the branch AudioCodes MP-118; long-distance toll calls from the branch phones are routed to the PSTN through the T1/E1 connection on the Avaya Media Gateway at the central location.
- In Survivable Mode, calls can be placed among phones within the branch. In addition, branch phones can still place calls to the PSTN (and to the phones at headquarters via PSTN) using the FXO interface on the AudioCodes MP-118 located at the branch site.
- PBX features including Hold, Transfer, Call Waiting, Call Forwarding and Conference on Avaya 9600 SIP Phones in both Normal and Survivable Modes.
- Analog phones connected to the FXS ports on the AudioCodes MP-118 are properly adapted as SIP phones in both Normal and Survivable Modes.

- Messaging system access by branch phones (through internal access number in Normal Mode and PSTN call in Survivable Mode) and proper function of MWI (Messaging Waiting Indicator) on Avaya 9600 IP Phones.
- Proper system recovery after AudioCodes MP-118 restart and loss/restoration of IP connection.

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

- **Call Waiting on branch analog phones do not work in Survivable Mode after initial Flash button press**: When a new call arrives at the analog phone already on call with an Avaya 9600 SIP IP Phone, the first Flash button press correctly switches to the new call while placing the existing call on hold. However, subsequent Flash button presses do not switch between the two calls. Traces on SIP messages in this call scenario seemed to indicate the problem was with the Avaya 9600 SIP IP Phone. On second Flash button press to switch back to the original call with the Avaya 9600 SIP IP Phone, the IP phone sends the 200 OK message which contains SDP contents with an indication that the phone status is *inactive*.

- **Delayed ring-back for PSTN calls in Survivable Mode**: When branch phones call into PSTN through the FXO interface on the AudioCodes MP-118, there is a pause of about 3 to 4 seconds between end of dialing and start of ring-back. AudioCodes support and development engineers investigated and determined that this behavior is due to the interface between the MP-118 FXO and the specific Service Provider analog trunk used in the testing to verify the sample configuration.

- **In Survivable Mode, no secondary dial-tone for branch phones after dialing PSTN access digit**: Currently there is no configuration on AudioCodes MP-118 that will enable a secondary dial-tone after a PSTN access digit is dialed for both IP and analog phones in the branch. Some specific configuration can enable the secondary dial-tone for the analog phones but not for IP phones.

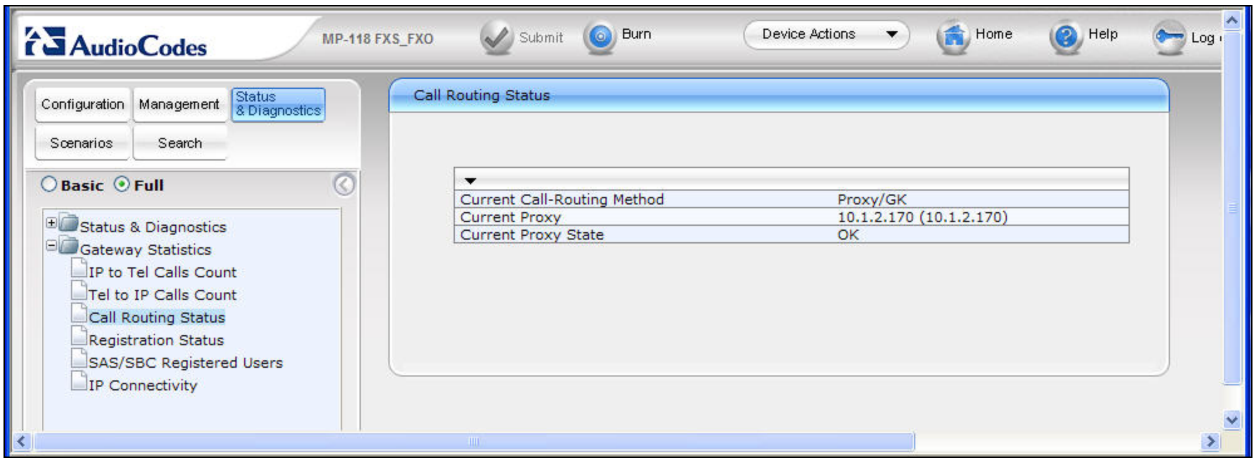
9. Verification Steps

9.1. AudioCodes MP-118 Call Routing Status

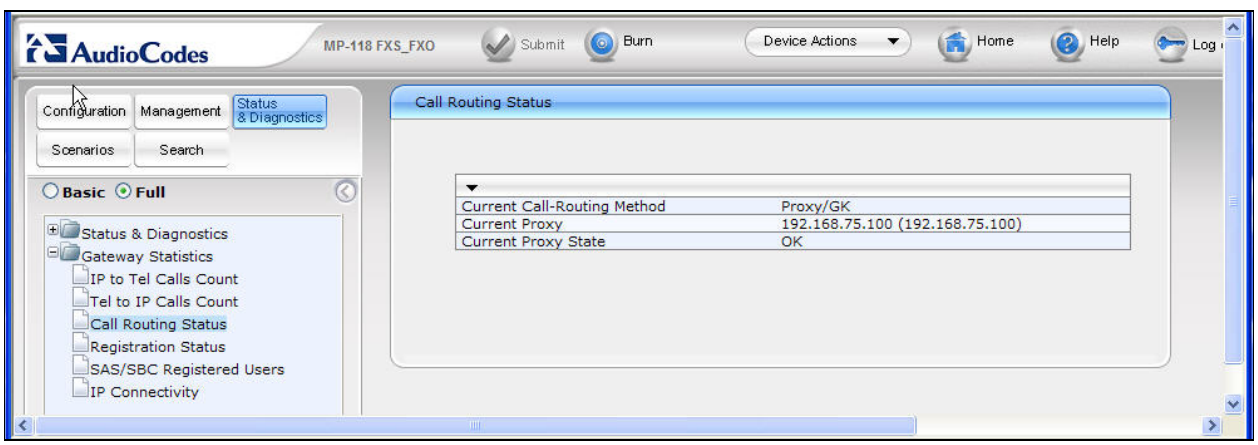
From the left navigation panel, select the **Status & Diagnostics** tab, then navigate to the Call Routing Status screen by selecting **Gateway Statistics → Call Routing Status**.

The Call Routing Status screens from the Branch 2 AudioCodes MP-118 while in Normal Mode and Survivable Mode are shown below:

Normal Mode:
The status shows all call routing is using the centralized Session Manager IP address which is in an "OK" state.



Survivable Mode:
The status shows all call routing is using the internal AudioCodes SAS Proxy IP address which is in an "OK" state.

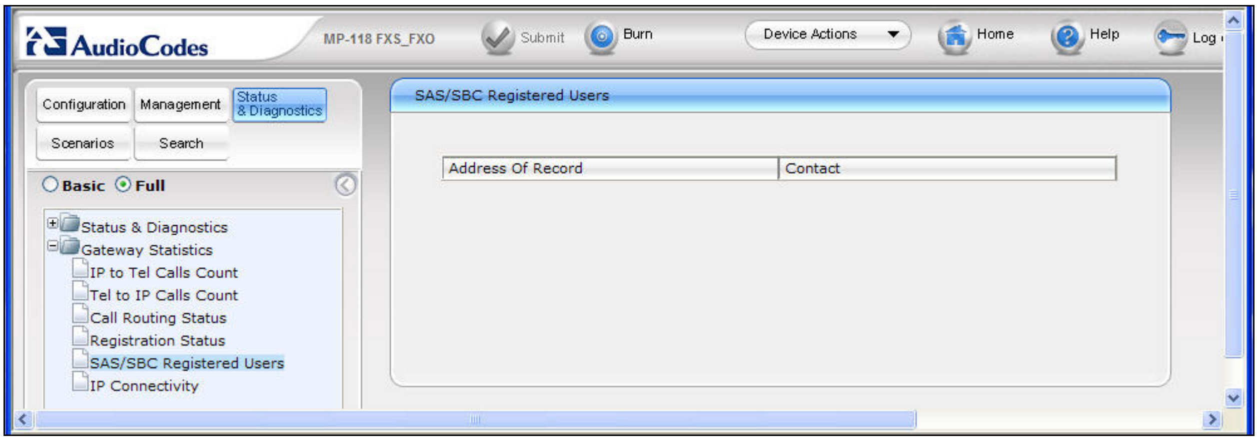


9.2. SAS/SBC Registered Users

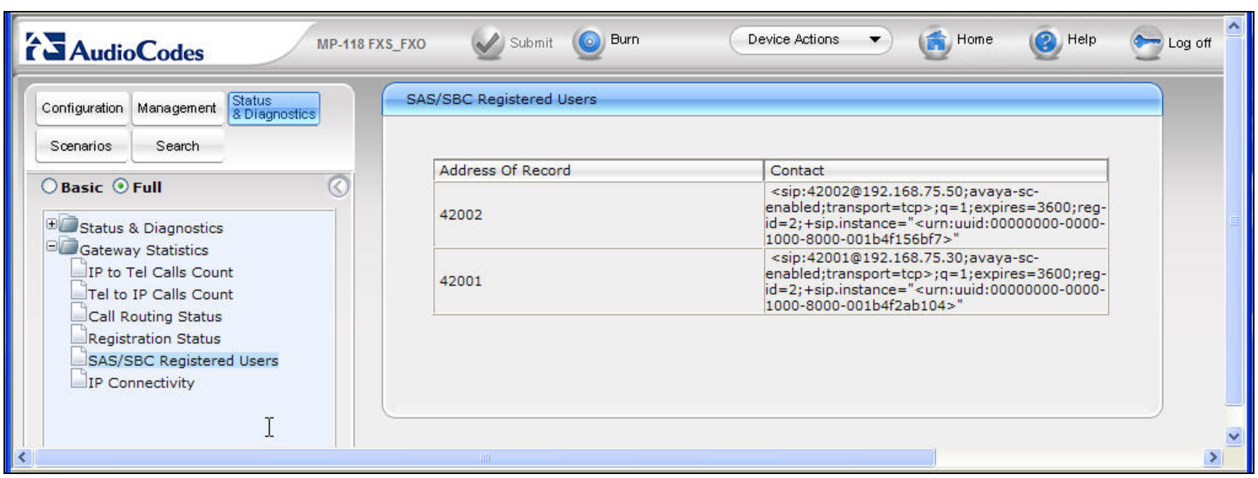
From the left navigation panel, select **Status & Diagnostics** then navigate to the SAS/SBC Registered Users screen by selecting **Gateway Statistics → SAS/SBC Registered Users**.

The SAS Registered Users screens from the Branch 2 AudioCodes MP-118 while in Normal Mode and Survivable Mode are shown below:

Normal Mode:
The screen shows no active SAS users.



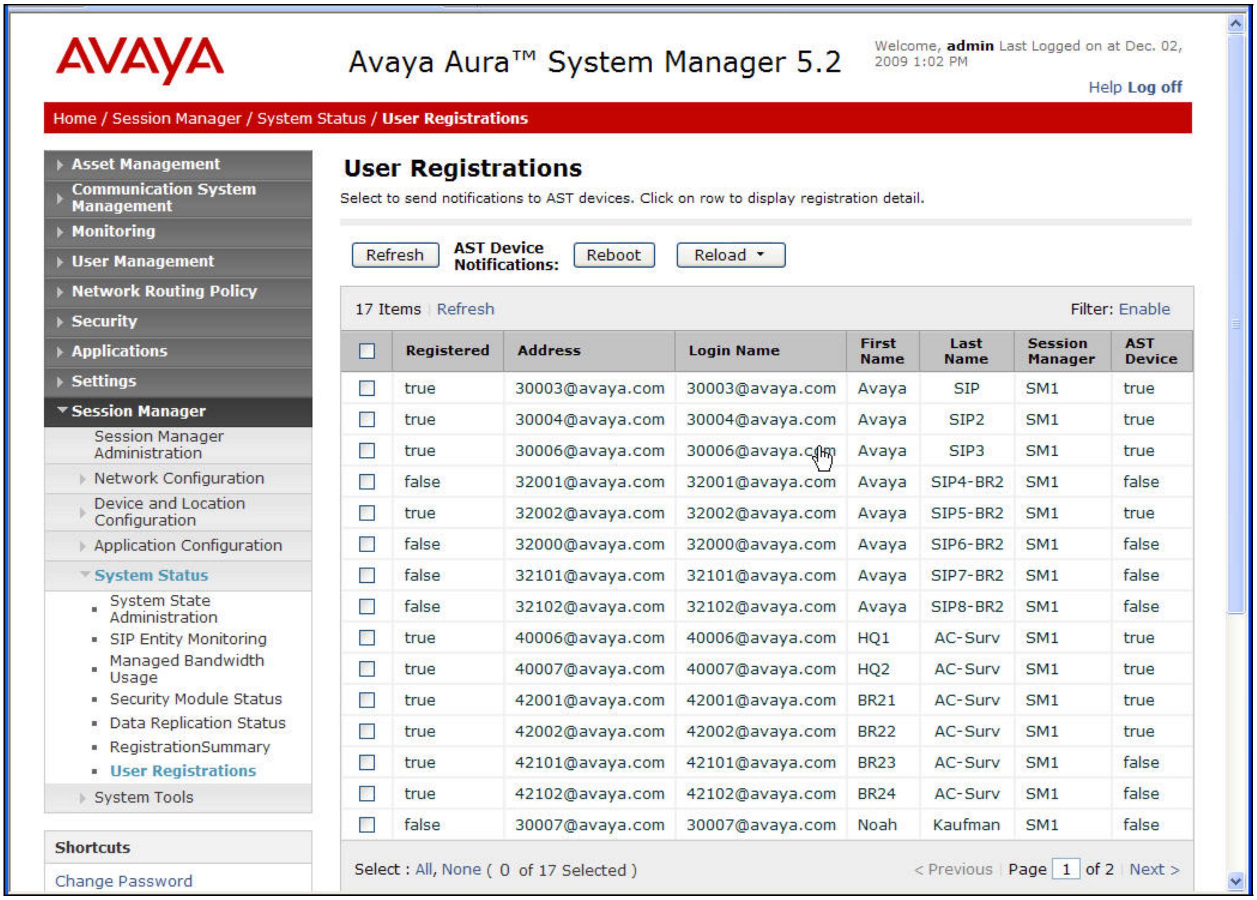
Survivable Mode:
The screen shows two Branch 2 Avaya 9600 SIP Phones actively registered to the AudioCodes MP-118 SAS.



9.3. Session Manager Registered Users

The following screen shows Session Manager registered users in Normal Mode. This screen can be accessed from the left navigation menu **Session Manager → System Status → User Registrations** on System Manger.

Note the user registrations for the 2 Avaya 9600 SIP Phones (42001 and 42002) and the two FXS stations (42101 and 42102) at the Branch 2 location. Also note the user registrations for the main site Avaya 9600 SIP Phones (40006 and 40007). The **AST Device** field indicates whether the registered phone is an Avaya SIP Telephone set.



9.4. Timing Expectations for Fail-over to AudioCodes SAS Mode

This section is intended to set *approximate* expectations for the length of time before Avaya 9600 SIP Telephones in the branch will acquire service from the AudioCodes Gateway, when a failure occurs such that the branch is unable to communicate with the central Session Manager. In practice, failover timing will depend on a variety of factors. Using the configuration described in these Application Notes, when the IP WAN is disconnected, idle Avaya SIP Telephones in the branch will typically display the "Acquiring Service…" screen in approximately 45 seconds. With multiple identical idle phones in the same branch, it would not be unusual for some phones to register to the AudioCodes Gateway for SAS service before others, with the earliest registering in approximately one minute and the latest registering in approximately two minutes. In other words, the Avaya SIP Telephones in the branch can typically place and receive calls processed by the AudioCodes Gateway approximately two minutes after the branch is isolated by a WAN failure.

9.5. Timing Expectations for Fail-back to Normal Mode

This section is intended to set *approximate* expectations for the length of time before Avaya 9600 SIP Telephones registered to the AudioCodes Gateway in SAS mode will re-acquire service from the Session Manager for normal service, once the branch communications with the central Session Manager is restored. In practice, failover timing will depend on a variety of factors. Using the configuration described in these Application Notes, when the IP WAN is restored such that the branch telephones can again reach the Session Manager, idle Avaya SIP Telephones in the branch will typically be registered with the Session in one minute or less. With multiple identical idle phones in the same branch, it would not be unusual for some phones to register back with the Session Manager before others. For example, some may register within 30 seconds, others within 45 seconds, with others registering in approximately one minute.

10. Conclusion

SIP endpoints deployed at remote branch locations risk a loss of service if a break in connectivity to the centralized SIP call control platform occurs. Connectivity loss can be caused by WAN access problems being experienced at the branch or network problems at the centralized site blocking access to the Avaya SIP call control platform. These Application Notes present the configuration steps to implement the Session Manager Survivable SIP Gateway Solution to minimize service disruption impact to these remote branch SIP endpoints.

11. Additional References

Avaya Aura™ Session Manager:
[1] *Avaya Aura™ Session Manager Overview*, Doc ID 03-603473, available at http://support.avaya.com.
[2] *Installing 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 one-X Deskphone Edition 9600 Series SIP IP Telephones:
[7] *Avaya one-X Deskphone Edition for 9600 SIP IP Telephones Administrator Guide*, Doc ID 16-601944, December 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 with Avaya Aura™ Communication Manager 5.2 as an Access Element – Issue 1.1*, 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. Appendix – Example Approach to 911

These Application Notes have illustrated a "Distributed Trunking" configuration, where calls from branch users can egress to the PSTN via an AudioCodes Gateway FXO port, both in normal mode and in survivable mode. In the sample configuration, when a branch user dials a PSTN number local to the branch where the call originates, Communication Manager uses ARS location-based routing to route the call back to Session Manager which is configured with a Dial Pattern that matches on the leading digits of the PSTN number (e.g.., an area code), and direct the call to the proper AudioCodes Gateway at the branch. The branch AudioCodes Gateway in turn routes the call to an FXO port.

Branch calls to 911 can be handled similarly. However, since the number "911" is common to all branches, Communication Manager can insert a branch prefix code so that the Dial Patterns configured on Session Manager can distinguish the proper AudioCodes Gateway based on the branch prefix. This approach uses the Communication Manager "route-pattern" to insert the branch prefix, and therefore this approach uses one additional "911 route-pattern" for each branch. Each unique "911 route-pattern" can direct the call to a common SIP trunk group to Session Manager. This Appendix shows the additions to the configuration to enable this approach to 911.

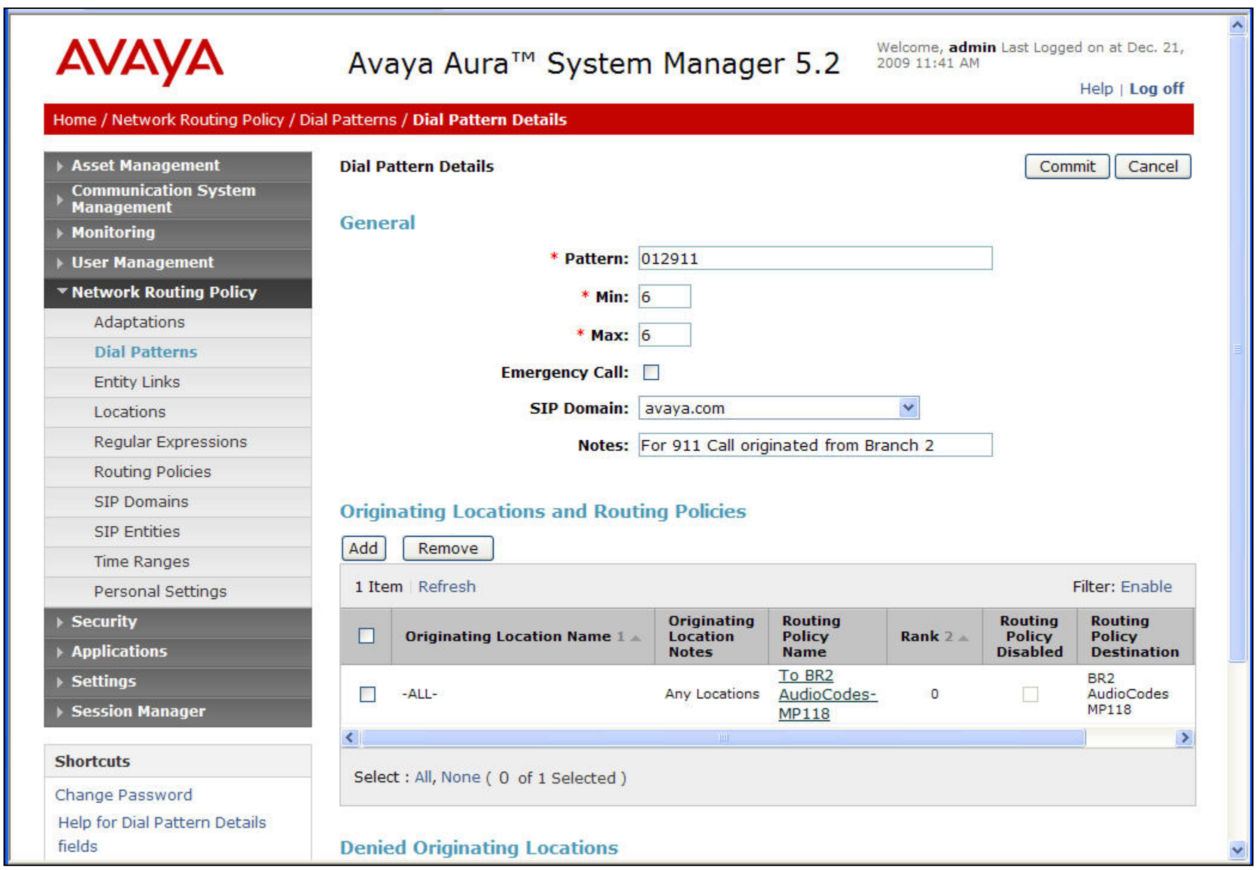
In Communication Manager, add a 911 entry to the ARS table for the location of each branch. An example is shown in bold for branch 2, which uses location 12 in the sample configuration. For 911 calls originated by branch 2 in Normal Mode, the bold entry will direct the call to route-pattern 129.

```
change ars analysis 1908 location 12                           Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  12           Percent Full:    2

          Dialed           Total     Route     Call   Node   ANI
          String          Min  Max  Pattern    Type   Num    Reqd
    1908                  11   11    12        natl          n
    911                   3    3     129       emer          n
                                                             n
                                                             n
```

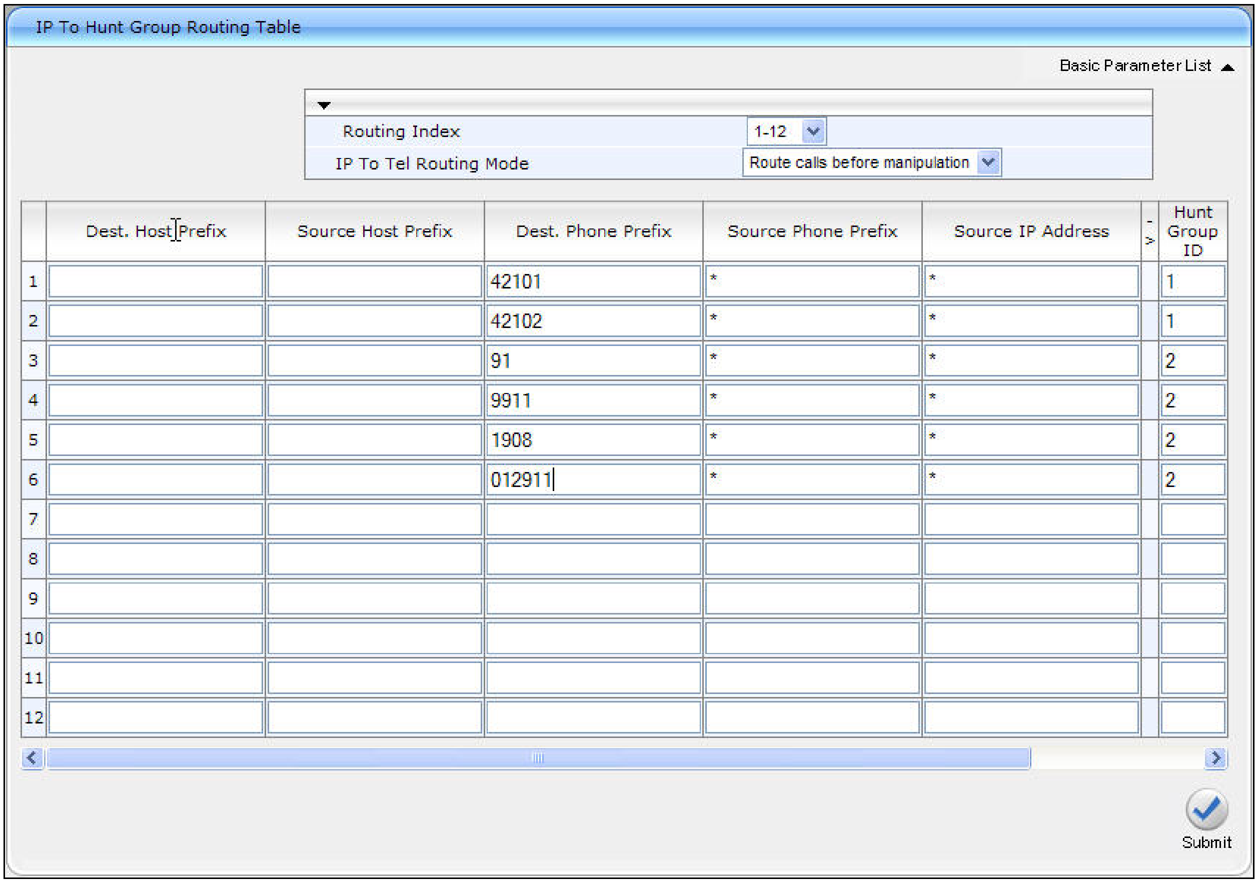
In route pattern 129, insert a prefix to uniquely identify the branch. In the sample below, the number "012" is chosen to match the location number used for ARS location-based routing. It is not necessary to match the location number. Trunk group 32 is a SIP trunk previously configured to connect Communication Manager to Session Manager.

```
change route-pattern 129                                        Page   1 of   3

                    Pattern Number: 129   Pattern Name: 911-Branch2
                             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                         012                                   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
```

In Session Manager, configure a Dial Pattern matching the number “012911”. Note the selection for the previously configured Routing Policy (“To BR2 AudioCodes-MP118”).



The sample configuration of the AudioCodes Gateway in these Application Notes requires an entry to be added to the IP To Hunt Group Routing Table (**Protocol Configuration → Routing Tables → IP to Trunk Group Routing**) to allow the AudioCodes Gateway to route the location-based 911 call out an FXO port. The 911 call will be directed to Hunt Group 2, and FXO port 5.



	Dest. Host Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Hunt Group ID
1			42101	*	*		1
2			42102	*	*		1
3			91	*	*		2
4			9911	*	*		2
5			1908	*	*		2
6			012911	*	*		2
7							
8							
9							
10							
11							
12							

The leading digits of the called numbers are used to determine the selected AudioCodes MP-118 Hunt Group. In the sample configuration, the FXS analog phone numbers are entered explicitly and route to Hunt Group ID 1. Calls to PSTN starting with "91" (including 911 call and 91xxxxxxxxxx conforming to North American Numbering Plan) as well as 911 call with a PSTN access digit "9" will route to Hunt Group ID 2. These two numbers are configured for calls originated from branch phones in Survivable Mode. Calls routed to the branch MP-118 from Session Manager with leading digits "1908" are local PSTN calls originated from branch phones in Normal Mode. Calls routed to the branch MP-118 from Session Manager with the number "012911" are 911 calls originated from branch phones in Normal Mode.

After these changes are completed, if 9-911 is dialed from an Avaya SIP Telephone at the branch while in Normal Mode, the call will egress FXO port 5 of the branch 2 MP-118 to the PSTN, and the call can be answered by a 911 operator. If it is desirable for 911 to be reachable without the user dialing the ARS access code 9, the ARS location based routing tables can include matching on “11” also. The “9” would be interpreted as the ARS access code, and the “11” with length 2 would be interpreted as another type of call intended to reach 911. A Session Manager Dial Pattern would also need to account for the alternate matching pattern.

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