Avaya Solution & Interoperability Test Lab

Avaya Communication Manager Survivable SIP Gateway Solution using the AudioCodes MP-114 in a Distributed Trunking Configuration – Issue 1.1

Abstract

These Application Notes present a sample configuration of the Avaya Communication Manager Survivable SIP Gateway Solution using the AudioCodes MP-114 Media Gateway in a Distributed Trunking configuration.

The Avaya Communication Manager Survivable SIP Gateway 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 is lost. 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.

The Avaya Communication 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 Gateway dynamically switch to survivability mode, restoring telephony services at 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 Solutions and Marketing Team.

1. Introduction

These Application Notes present a sample configuration of the Avaya Communication Manager Survivable SIP Gateway Solution using the AudioCodes MP-114 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 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. The Avaya Communication Manager 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 endpoint and SIP gateway components within the branch dynamically switch to survivability mode to provide basic telephony services at the branch. When connectivity from the branch to the centralized Avaya SIP call control platform is restored, SIP components can dynamically switch back to normal operation.

The primary components of the Avaya Communication Manager Survivable SIP Gateway Solution are the Avaya one-X Deskphone SIP 9600 Series IP Telephones and the AudioCodes SIP Media Gateways models MP-114 and MP-118. The sample configuration presented in these Application Notes utilizes the AudioCodes SIP Media Gateway model MP-114. These configuration steps can also be applied to the AudioCodes SIP Media Gateway model MP-118 using the AudioCodes firmware version specified in **Table 2**.

2. Overview

2.1. AudioCodes SIP Media Gateway

The AudioCodes SIP Media Gateway, referred to as AudioCodes MP-114 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 Avaya SES)

Note: AudioCodes labels the Survivable SIP Registrar and Proxy functionality of the MP-114 as Stand-Alone Survivability (SAS). SAS will be used throughout these Application Notes.

2.2. 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 Avaya Communication Manager Survivable SIP Gateway Solution. The 2.4 firmware release of the Avaya 9600 SIP Phone includes new feature capabilities specific to SIP survivability enabling the phone to monitor connectivity to the Avaya SES and dynamically failover to the local AudioCodes MP-114 as an alternate or survivable SIP server. See reference [1] for additional information on the Avaya 9600 SIP Phone.

2.3. PSTN Trunking Configurations

The Avaya Communication 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 by Avaya Communication Manager 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 the Avaya Communication Manager Survivable SIP Gateway Solution:

Centralized Trunking: All PSTN calls, inbound to the enterprise and outbound from the enterprise, are routed to/from PSTN media gateways centrally located at the Headquarters/Datacenter location.

Distributed Trunking: PSTN call routing can be determined by the originating source location using Avaya 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-114 branch gateway. 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 Distributed Trunking call flows presented in **Section 2.5** provide additional details of how calls are routed based on the location of the caller and the number being called.

The sample configuration presented in these Application Notes implements a Distributed Trunking configuration. For a sample configuration of the Avaya Communication Manager Survivable SIP Gateway Solution in a Centralized Trunking configuration, see the Application Notes titled "Avaya Communication Manager Survivable SIP Gateway Solution using the AudioCodes MP-114 in a Centralized Trunking Configuration" [4].

2.4. Network Modes

PSTN call routing is further determined within each of the trunking configurations based on the network status of each branch.

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, and the local branch AudioCodes MP-114 SAS SIP call control is being used for all calls at that branch. Note: if the Avaya SES loses connectivity to the WAN, all branches will go into survivable mode simultaneously.

2.5. Call Flows

The section presents the primary call flows for the Avaya Communication Manager Survivable SIP Gateway Solution in a Distributed Trunking configuration for both Normal Mode and Survivability Mode. The components included in these call flows are based on the components used in the sample configuration presented in these Application Notes.

2.5.1. Distributed Trunking – Normal Mode

Overview:

- **SIP Call Control**: All SIP call control and call routing is provided by the centralized Avaya SES and Avaya Communication Manager.

- **Branch PSTN Outbound Local**: Avaya Communication Manager Location Based Routing and Avaya SES Host Address Maps are used to route these calls to the local branch AudioCodes MP-114 FXO interface.

- **Branch PSTN Outbound Non-Local**: PSTN outbound calls from the branch to non-local numbers are routed to a centralized Avaya G650 Media Gateway.

- **Branch PSTN Inbound**: Calls from the PSTN to a branch Listed Directory Number (LDN) enter the network at the local branch AudioCodes MP-114 FXO interface then route to the Avaya SES/Avaya Communication Manager for call treatment.

- **Headquarters PSTN Inbound**: Calls to Headquarters endpoints enter the network at the Headquarters Avaya G650 Media Gateway.

Call Flows:

1. **Avaya 9600 SIP Phone at branch to H.323 IP phone at Headquarters.**

   Avaya 9600 SIP → SES → Avaya Communication Manager → H.323 IP phone

2. **Avaya 9600 SIP Phone at branch to Digital/Analog phone at Headquarters.**

   Avaya 9600 SIP → SES → Avaya Communication Manager → Avaya Media Gateway → Digital/Analog phone

3. **Avaya 9600 SIP Phone at branch to PSTN endpoint – Local Number**

   Avaya 9600 SIP→ SES → Avaya Communication Manager → SES → AudioCodes MP-114 FXO → PSTN phone

4. **Avaya 9600 SIP Phone at branch to PSTN endpoint – Long Distance Number**

   Avaya 9600 SIP → SES → Avaya Communication Manager → Avaya Media Gateway → PSTN phone

5. **Avaya 9600 SIP Phone at branch to Avaya 9600 SIP phone at same branch.**

   Avaya 9600 SIP → SES → Avaya Communication Manager → SES → Avaya 9600 SIP

6. **PSTN phone to Branch LDN assigned to Avaya 9600 SIP phone.**

   PSTN phone → AudioCodes MP-114 FXO → SES → Avaya Communication Manager → SES → Avaya 9600 SIP

Figure 1 presents a high level network view of the Distributed Trunking Normal Mode call flows.

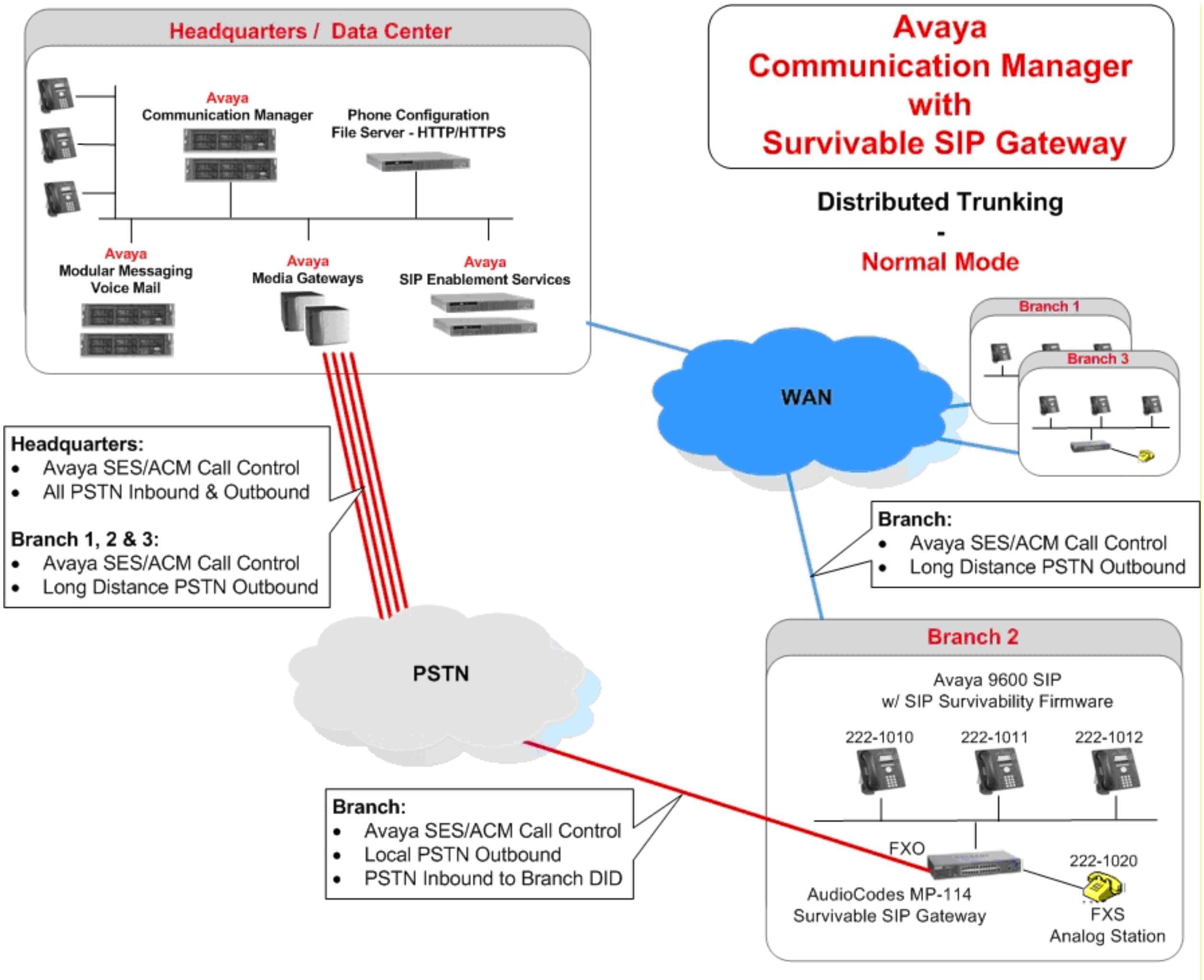


Figure 1

2.5.2. Distributed Trunking – Survivability Mode

Overview:

- **SIP Call Control**: All SIP call control and call routing is provided by the local branch AudioCodes MP-114 SAS.

- **SIP Registration**: All branch Avaya 9600 SIP Phones are transitioned and registered to the AudioCodes MP-114 SAS.

- **All Branch PSTN Outbound**: Local and Non-Local: Routed to the AudioCodes MP-114 FXO interface.

- **Branch PSTN Inbound**: Calls from the PSTN to a branch LDN or DID enter the network at the local branch AudioCodes MP-114 FXO interface. The AudioCodes MP-114 SAS routes the call to a phone assigned to the FXO interface.

Call Flows:

1. **Avaya 9600 SIP Phone at branch to PSTN endpoint – Local & Non-Local**

   Avaya 9600 SIP → AudioCodes MP-114 FXO → PSTN phone

2. **PSTN phone to Branch LDN or DID assigned to Avaya 9600 SIP phone.**

   PSTN phone → AudioCodes MP-114 FXO → Avaya 9600 SIP

3. **Avaya 9600 SIP Phone at branch to Avaya 9600 SIP phone at same branch.**

   Avaya 9600 SIP → AudioCodes MP-114 → Avaya 9600 SIP

Figure 2 presents a high level view of the Distributed Trunking Survivable Mode call flows.

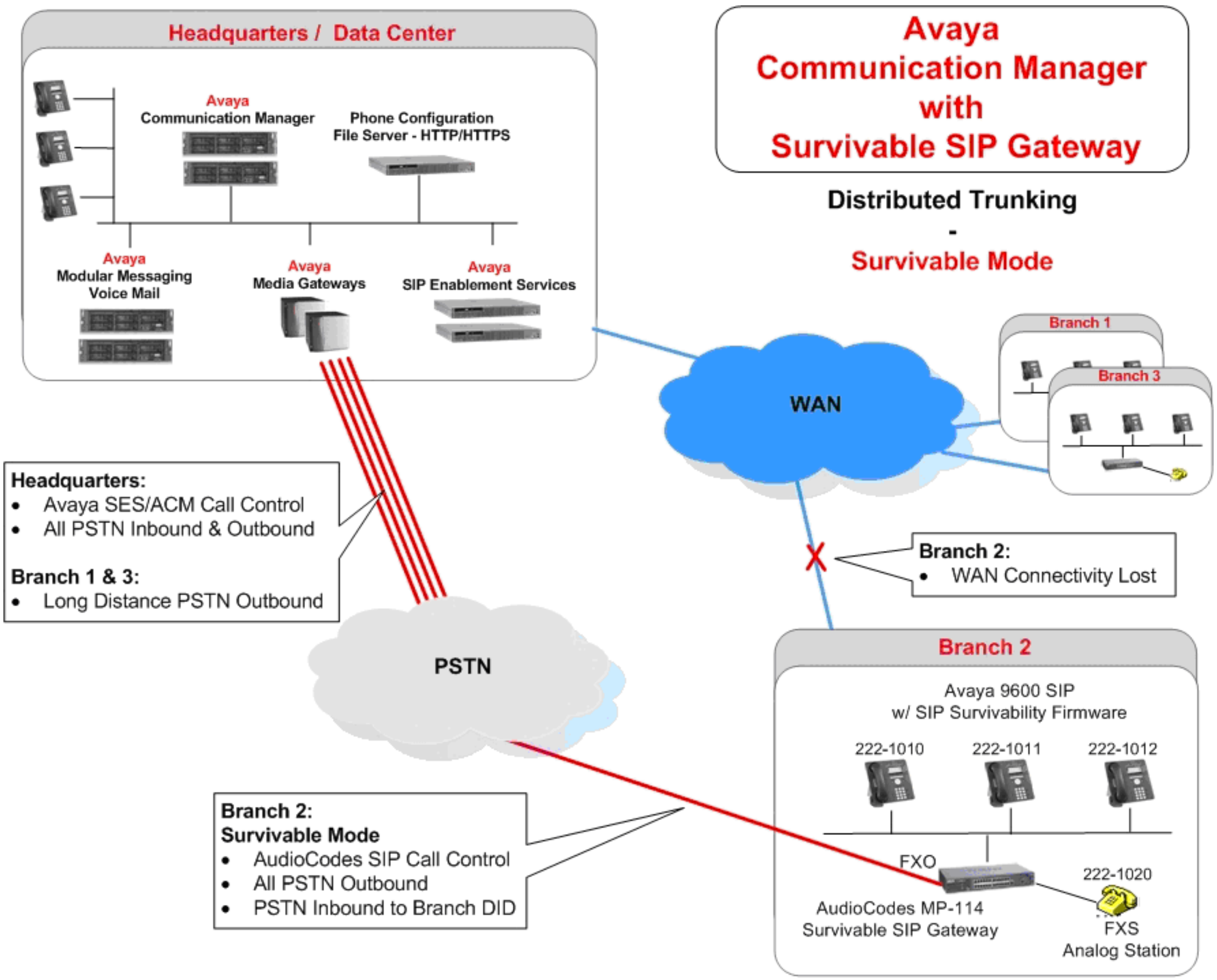


Figure 2

2.5.3. Detailed Call Flow: Branch PSTN Outbound Local – Normal Mode

Many of the Avaya SES and Avaya Communication 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 to better understand the linkage of the various configuration steps.

Branch PSTN Outbound Local – Normal Mode:

Branch 2 Avaya 9600 SIP Phone user dials the following local PSTN number:

9 1-732-555-1111.

1. Branch 2 Avaya 9600 SIP Phone sends SIP INVITE to Avaya SES with dialed digit string of 917325551111.
2. Avaya SES receives the SIP INVITE and identifies the Avaya 9600 SIP Phone user has an assigned Communication Manager Extension. Avaya SES forwards the SIP INVITE to Avaya Communication Manager.
3. Avaya Communication Manger receives the SIP INVITE from Avaya SES on SIP Trunk Group Number 7.
4. Avaya 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. Avaya Communication Manager now knows the source of the call is Location 12.
5. The leading 9 in the dialed digit string is identified by Avaya 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 17325551111.
7. A match on 1732 is found and Route Pattern 8 is chosen.
8. Route Pattern 8 strips the leading digit, 1, and routes the call to SIP Trunk Group Number 8.
9. Avaya Communication Manager sends a new SIP INVITE to Avaya SES over SIP Trunk Group Number 8 with the dialed digits of 7325551111.
10. Avaya SES Host Address Map with the name “Branch2-MP114” matches on the digits 732 of the dialed number and identifies the next hop as the Branch 2 AudioCodes MP-114 with IP address 22.1.1.100 using TCP port 5070.
11. Avaya SES forwards the SIP INVITE with dialed digits string 7325551111 to the Branch 2 AudioCodes MP-114.
12. The Branch 2 AudioCodes MP-114 internally routes the call to an FXO interface for termination on the PSTN.

2.6. Network Topology

The network implemented for the sample configuration shown in **Figure 3** 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.

The Headquarters location hosts an Avaya SES and Avaya Communication Manager providing enterprise wide SIP call control and advanced feature capabilities. The Avaya SES consists of separate Home and Edge servers. Avaya Communication Manager is running on Avaya S8710 redundant servers. A flat network of 30.1.1.0/24 is implemented at Headquarters. The Headquarters network is mapped to IP Network Region 1 which is assigned to Location 1 within Avaya Communications Manager. The Headquarters location also hosts the following centralized components: an Avaya G650 Media Gateway with PSTN trunks, an Avaya Modular Messaging voice mail platform, and an Avaya IP Phone Configuration File Server. The configuration details of these components are considered out of scope of these Application Notes and are 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 7** includes the parameters of the 46xxsettings.txt file used by the Avaya 9600 SIP Phone for survivability. The Avaya Modular Messaging voice mail platform can be reached by dialing the internal extension configured as the voice mail access numbers or pilot number, or by dialing a PSTN number that also terminates to Modular Messaging. The internal or private 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 phone is in Survivable Mode. This enables branch users to continue to access the centralized voice mail platform while in Survivable Mode via the PSTN using the Message button. Traditional Message Waiting Indication via the telephone is not available while the phone is in Survivable Mode. The messaging system, such as Avaya Modular Messaging, may enable other methods of notification that a message has been delivered.

The branch locations consist of several Avaya 9600 SIP Phones, an AudioCodes MP-114 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. The branch IP network addressing, IP Network Region numbers and Location ID's all use a numbering scheme associated with the branch's number.

The Distributed Trunking capabilities of the solution utilize the source based call routing feature of Avaya 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-114 IP Address
30.1.1.0/24	1	1	201	
11.1.1.0/24	11	11	609	11.1.1.100
22.1.1.0/24	12	12	732	22.1.1.100
33.1.1.0/24	13	13	908	33.1.1.100

Table 1 – Network Information

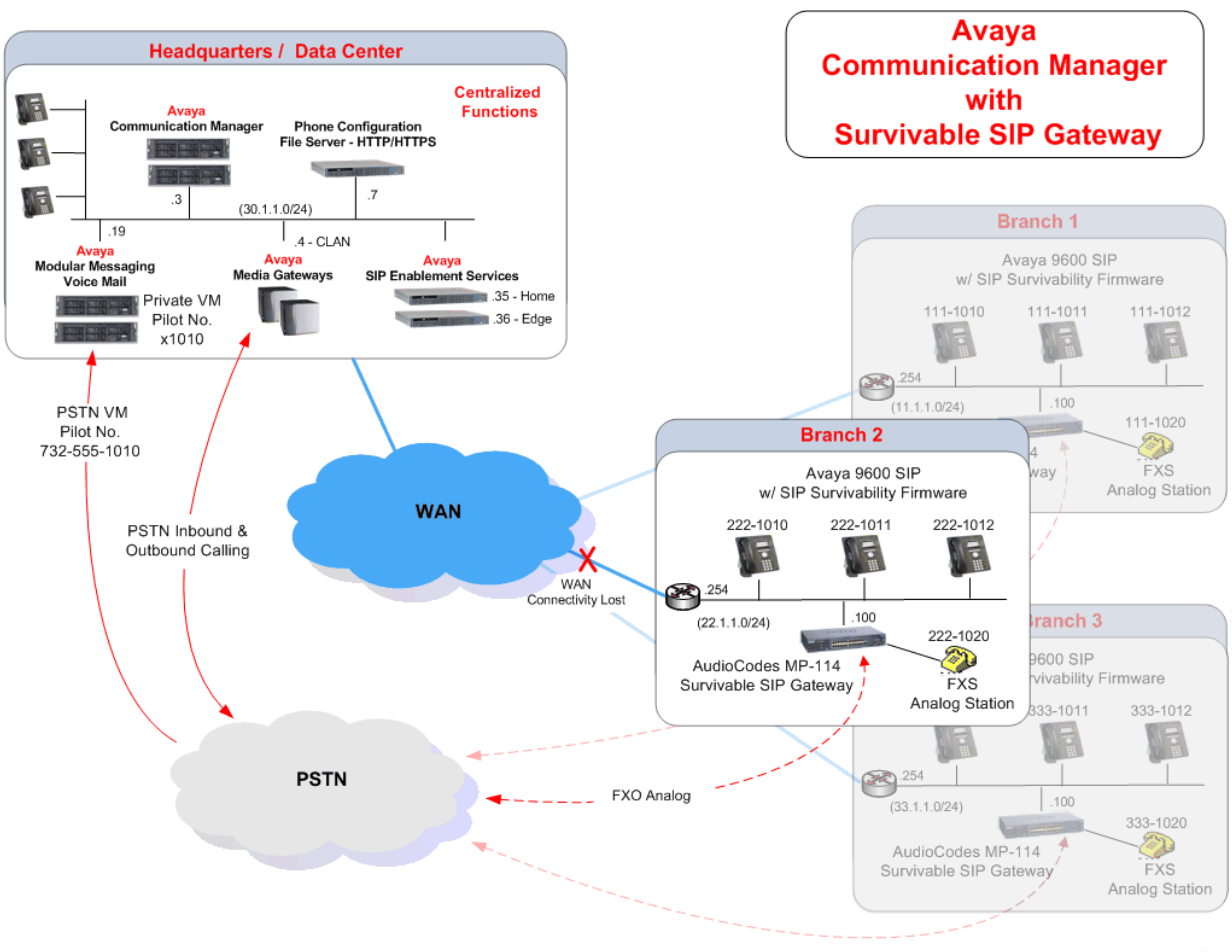


Figure 3 – Network Diagram

3. Equipment and Software Versions

The information in these Application Notes is based on the software and hardware versions listed in **Table 2**.

Device Description	Version
Avaya Communication Manager - S8710 Servers	Release 5.1 (R015x.00.1.414.3)
Avaya SIP Enablement Services	Release 5.1.1 (5.1.1.415.1)
Avaya G650 Media Gateway - IPSI (TN2312BP) - CLAN 1 (TN799DP) - MedPro (TN2302AP)	 - HW10 FW042 - HW01 FW026 - HW20 FW117
Avaya one-X Deskphone SIP 9600 Series Models: 9620 and 9630	R2.4.1.3
AudioCodes MP-114	FW version: 5.60A.010.005
List of Layer 2 Switches Tested at Branch: Avaya C363T / C364T – PWR CISCO Catalyst 3750 (or 3750G) PoE 24 CISCO Catalyst Express 500 D-Link DES-1526 PoE Switch (Discontinued. Replacement DES-1228P) Extreme Networks Summit X450-24t 3com Switch 4400 PWR / 3C17205 SuperStack 3	 SW 4.5.14 SW 12.2.25-SEB4 SW 12.2.25-SEG3 FW 1.00.04 SW 11.5.1.4 SW 3.12
H.323 Fax Adaptor - MultiTech MVP130-AV-FXS	- HW MVP130-AV-FXS-rev.A [B7b8] - FW 2.06.FQ
Fax Devices - OKI Okifax 5300 plus - Sharp UX510 - Brother IntelliFax 1360 - HP LaserJet 3050 - HP LaserJet 4345mfp (Avaya Building Facility) - Ricoh Aficio MP2000	 Model FX-050BVP (as is) (as is) FW 20060117 FW 09.131.1 FW 02.00.00 B2765522B

Table 2 – Software/Hardware Version Information

3.1. Layer 2 Switch

In lab testing, the Avaya 96xx series SIP phones can not acquire the SIP services from the AudioCodes MP-114/118 SIP Gateway under the survivability mode when NETGEAR FS-116P or FS-108P switches are positioned at the branch for the physical connectivity of AudioCodes MP114/118 and Avaya 96xx series SIP phones and interface to the enterprise WAN network. If the interface the WAN network is disconnected or out of service, the address resolution to locate the IP address the AudioCodes MP114/118 fails at the NETGEAR layer 2 switches. Avaya 96xx series SIP phones can not successfully establish IP connections to the AudioCodes MP114/118. So, the NETGEAR layer 2 switches – FS-116P and FS-108P should be avoided at the branch network setup.

The list of Layer 2 switches as documented in **Table 2** have been verified in Avaya SIL lab. They provide the adequate functions for SIP Survivability Solutions.

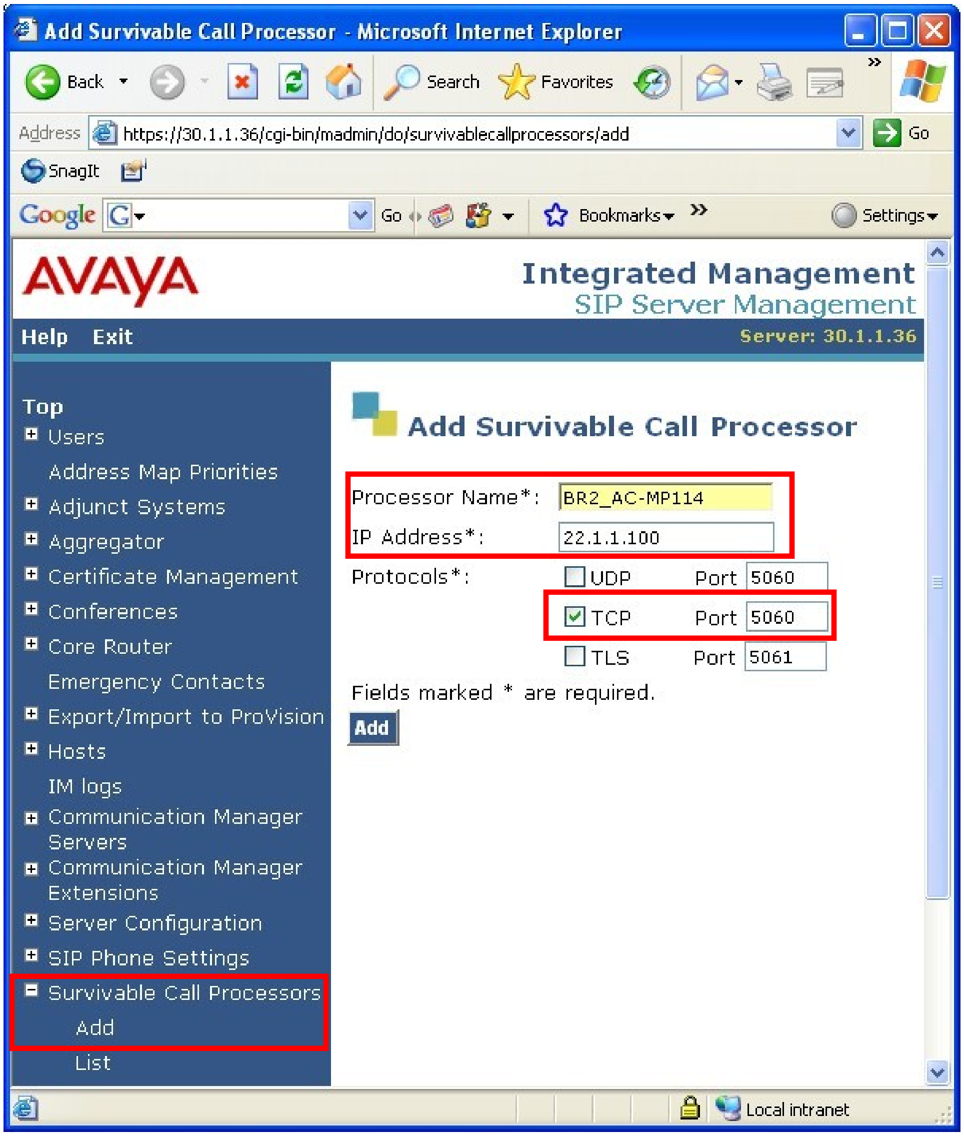
4. Avaya SES

This section describes the configuration steps for the Avaya SES.

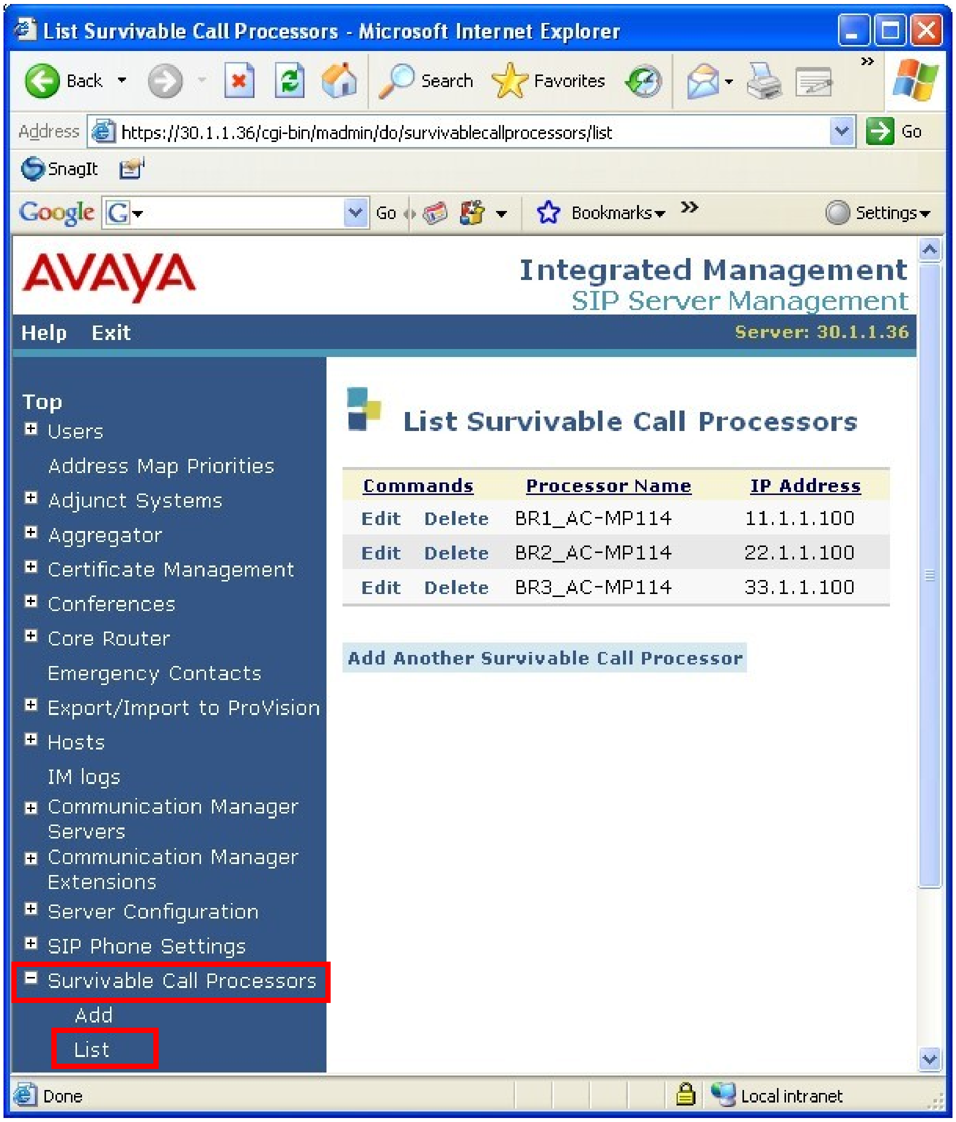
4.1. Survivable Call Processors

The Survivable Call Processors feature of the Avaya SES allows survivable SIP servers to be defined and then assigned to individual SES user accounts. The AudioCodes MP-114 at each branch is configured as a Survivable Call Processor within SES. Each individual SES user account is assigned to the AudioCodes MP-114 Survivable Call Processor that matches the branch office that the user is located. The 9600 SIP Phone will download the Survivable Call Processor information from Avaya SES when a user with an assigned Survivable Call Processor logs in.

The following screen illustrates the AudioCodes MP-114 for Branch 2 being added as a Survivable Call Processor. Note the Survivable Call Processor configuration for the AudioCodes MP-114 is set to use the TCP transport protocol on port 5060. The Avaya 9600 SIP Phone will use these same parameters when registering with the AudioCodes MP-114 in survivable mode.



The following screen illustrates the list of Survivable Call Processor for all three branches in the sample configuration.

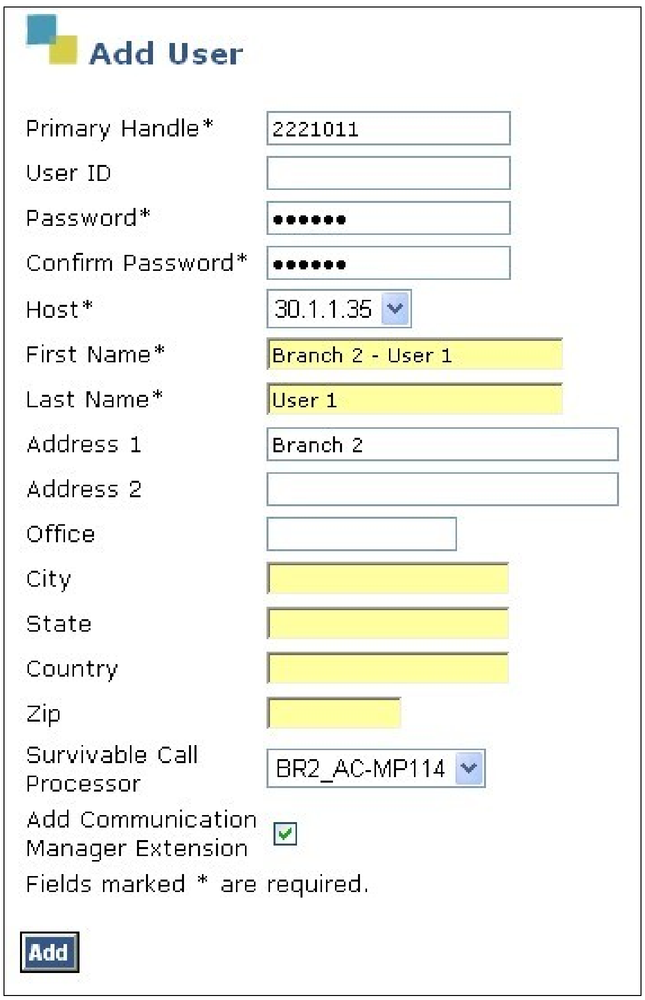


4.2. SIP User Accounts

4.2.1. Avaya 9600 SIP Phone Accounts

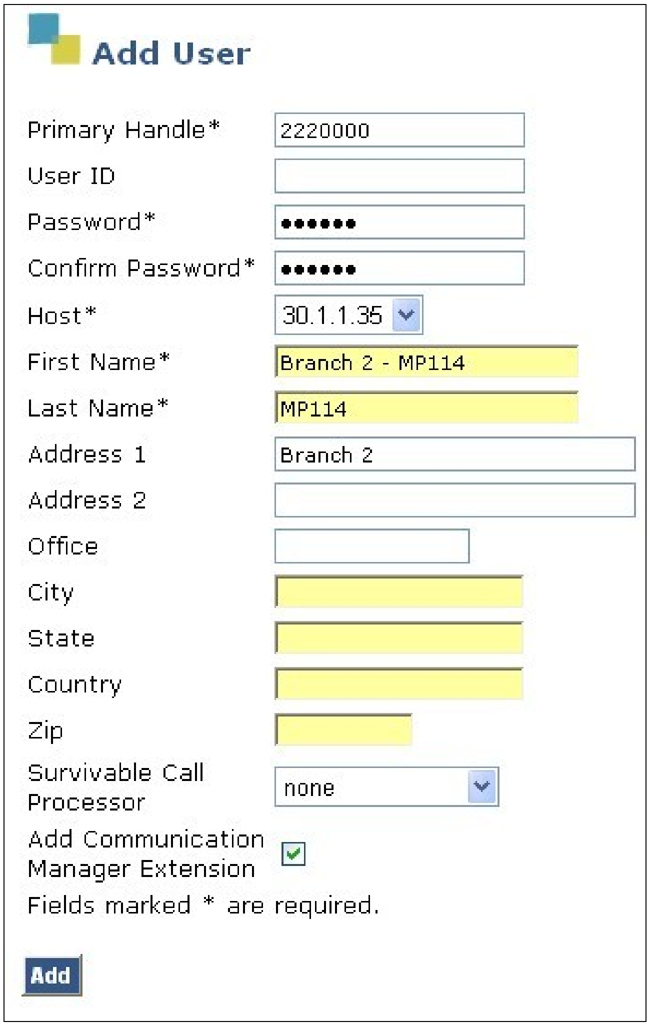
An account must be created for each Avaya 9600 SIP Phone user by selecting **User → Add** from the Avaya SES left navigation panel. The account must be configured with the Survivable Call Processor for the branch location that the user is located. Each user account must also be configured with a Communication Manager Extension. The screen below, left, illustrates the creation a user account for Branch 2 of the sample configuration. Note that the BR2_AC-MP114 Survivable Call Processor was selected for this Branch 2 user.

After adding the user account, the Add Communication Manager Extension screen appears similar to the one shown below, right. Enter the appropriate extension, typically the same extension as the Primary Handle of the user account. This Communication Manager Extension must also be created on Avaya Communication Manager as described in **Section 5.9**.



4.2.2. AudioCodes MP-114 SIP User Account

Each AudioCodes MP-114 is configured with a SIP user account on Avaya SES and Extension on Avaya Communication Manager. The following screens illustrate the creation of an SES user account with Communication Manager Extension for the Branch 2 AudioCodes MP-114 of the sample configuration. Note the AudioCodes MP-114 is itself a Survivable Call Processor for Branch 2 resulting in the selection of none for the Survivable Call Processor field. This Communication Manager Extension must also be created on Avaya Communication Manager as described in **Section 5.9**.



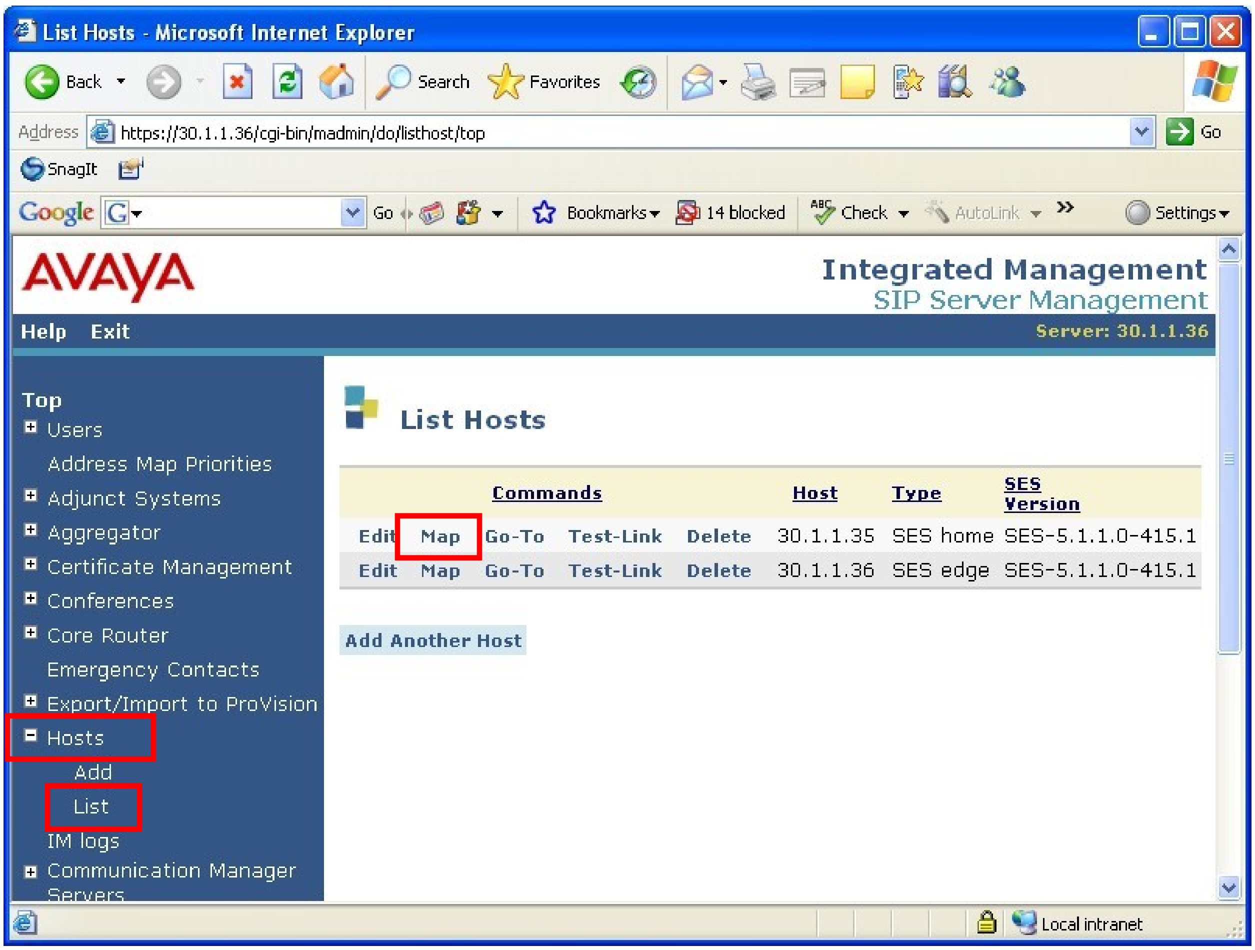
4.2.3. AudioCodes MP-114 FXS Analog Phone SIP User Account

Each AudioCodes MP-114 FXS Analog Phone must be configured with a SIP user account on Avaya SES and Extension on Avaya Communication Manager. The following screens illustrate the creation of an SES user account with Communication Manager Extension for one of the FXS Analog Phone on the Branch 2 AudioCodes MP-114 of the sample configuration. Note the AudioCodes MP-114 is itself a Survivable Call Processor for Branch 2 resulting in the selection of none for the Survivable Call Processor field. This Communication Manager Extension must also be created on Avaya Communication Manager as described in **Section 5.9**.

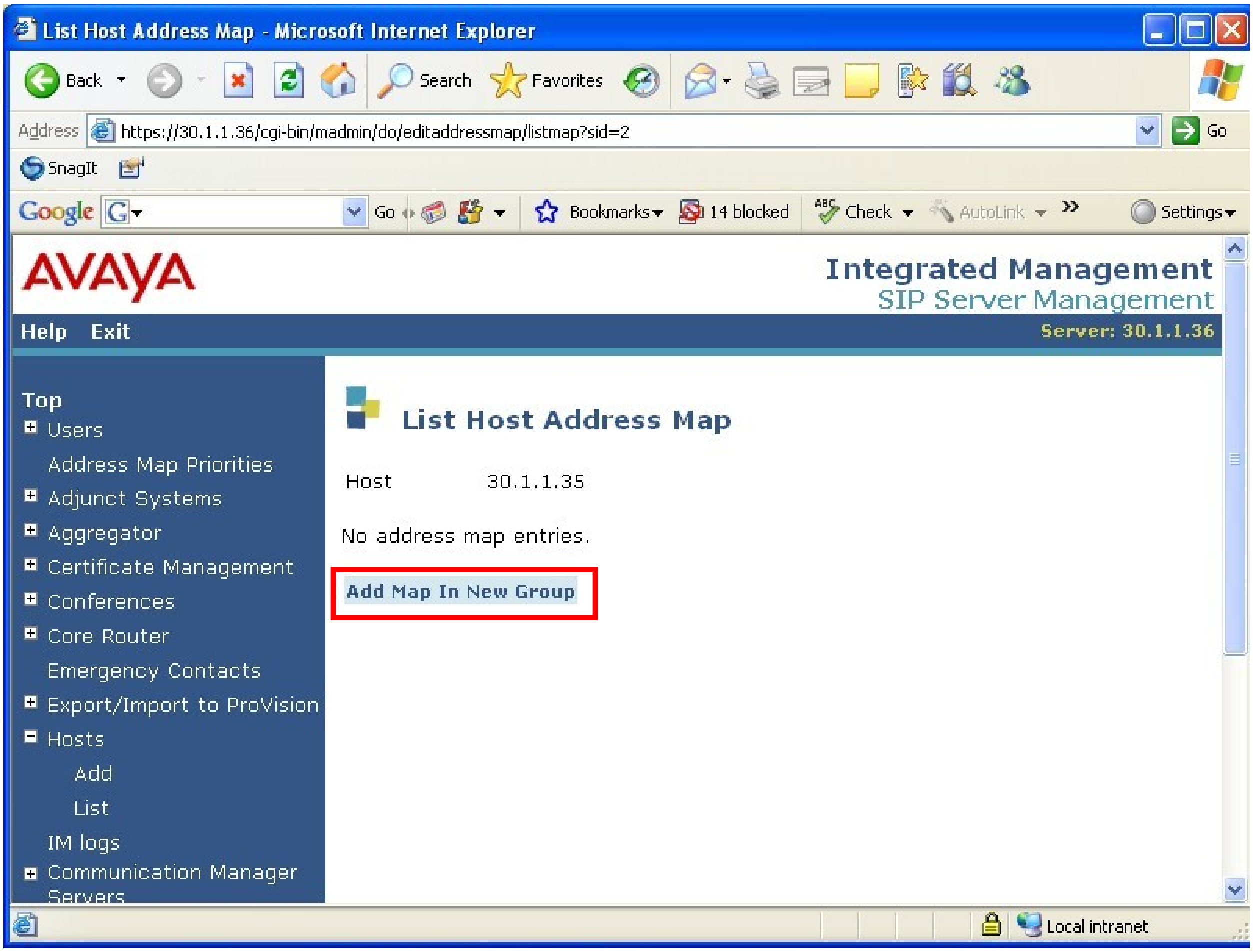
4.3. Host Address Maps

For calls to local PSTN numbers originated at a branch, Avaya Communication Manager will use location based routing to direct the call back to the Avaya SES. The Avaya SES must route the call to the proper branch AudioCodes MP-114 gateway. To accomplish this, Host Address Maps are created by selecting **Hosts → List → Map** from the Avaya SES left navigation panel. The Map must be added to the Avaya SES Home server to which the branch AudioCodes MP-114 is registered.

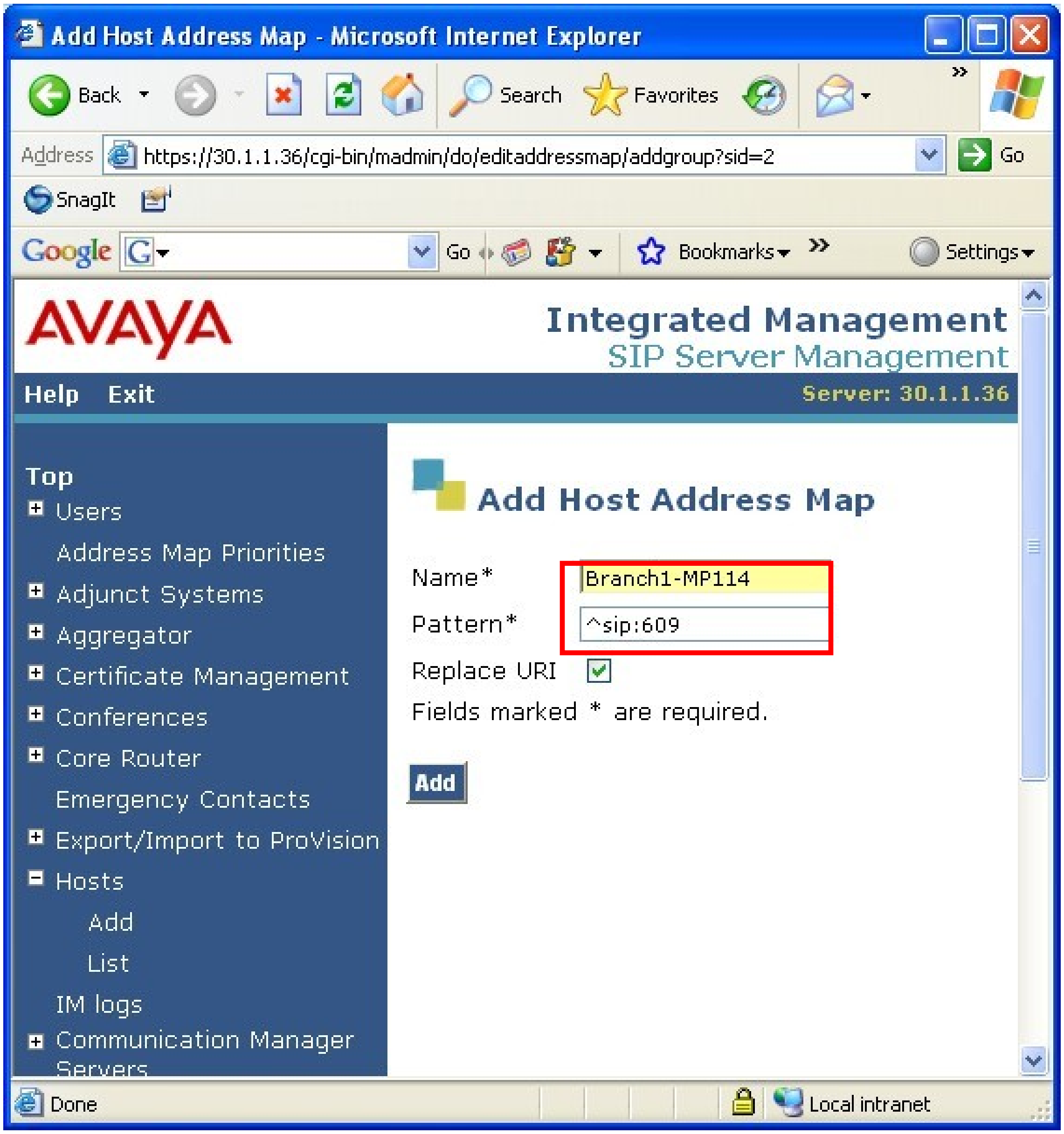
In the sample configuration, Host Address Maps match on the area code associated with each branch as shown in **Table 1** for the SIP call leg coming from Avaya Communication Manager. Each of these Host Address Maps then specify the IP Address, transport and port number to use to signal to the AudioCodes MP-114 Gateway at the proper branch.



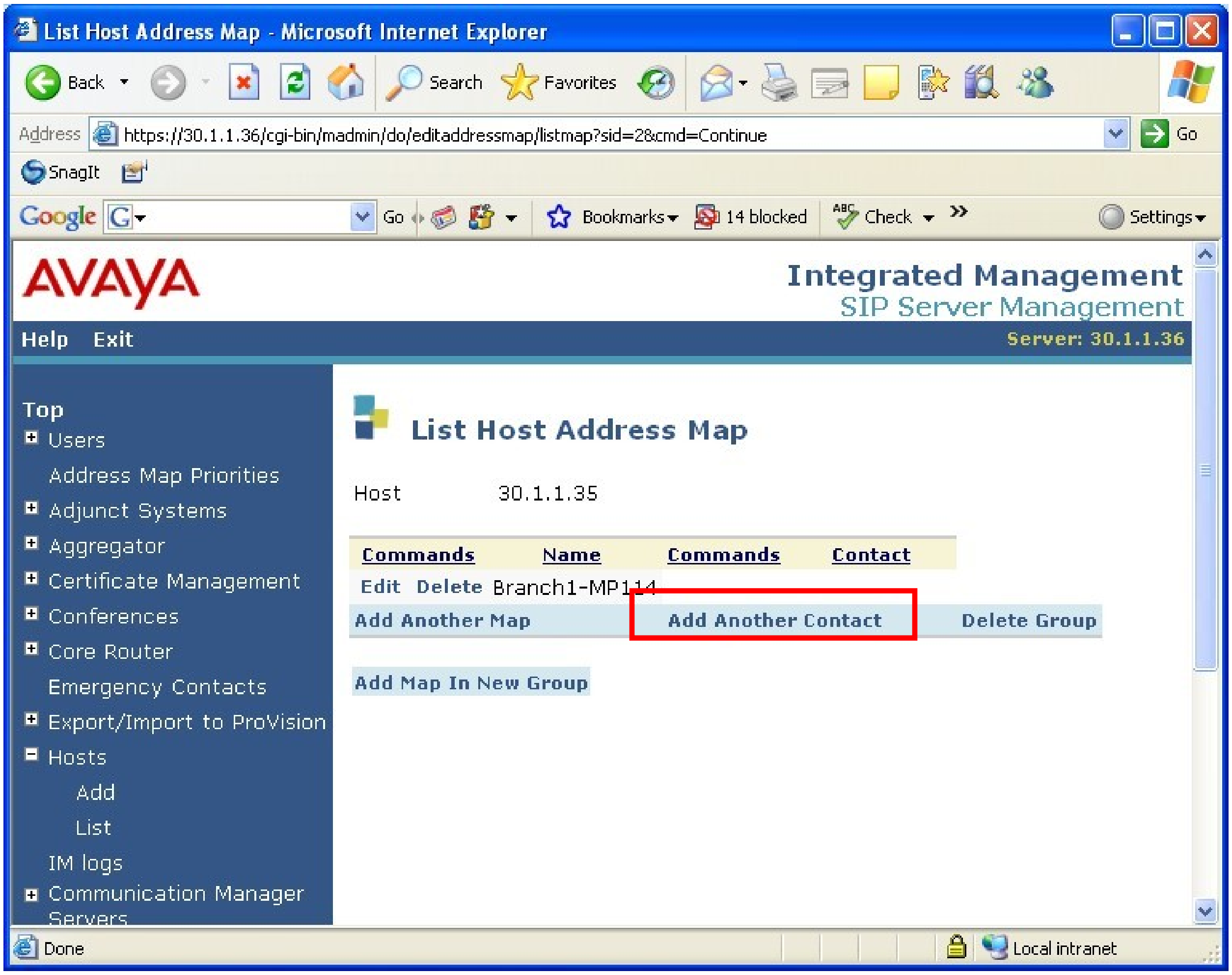
Select **Add Map In New Group**



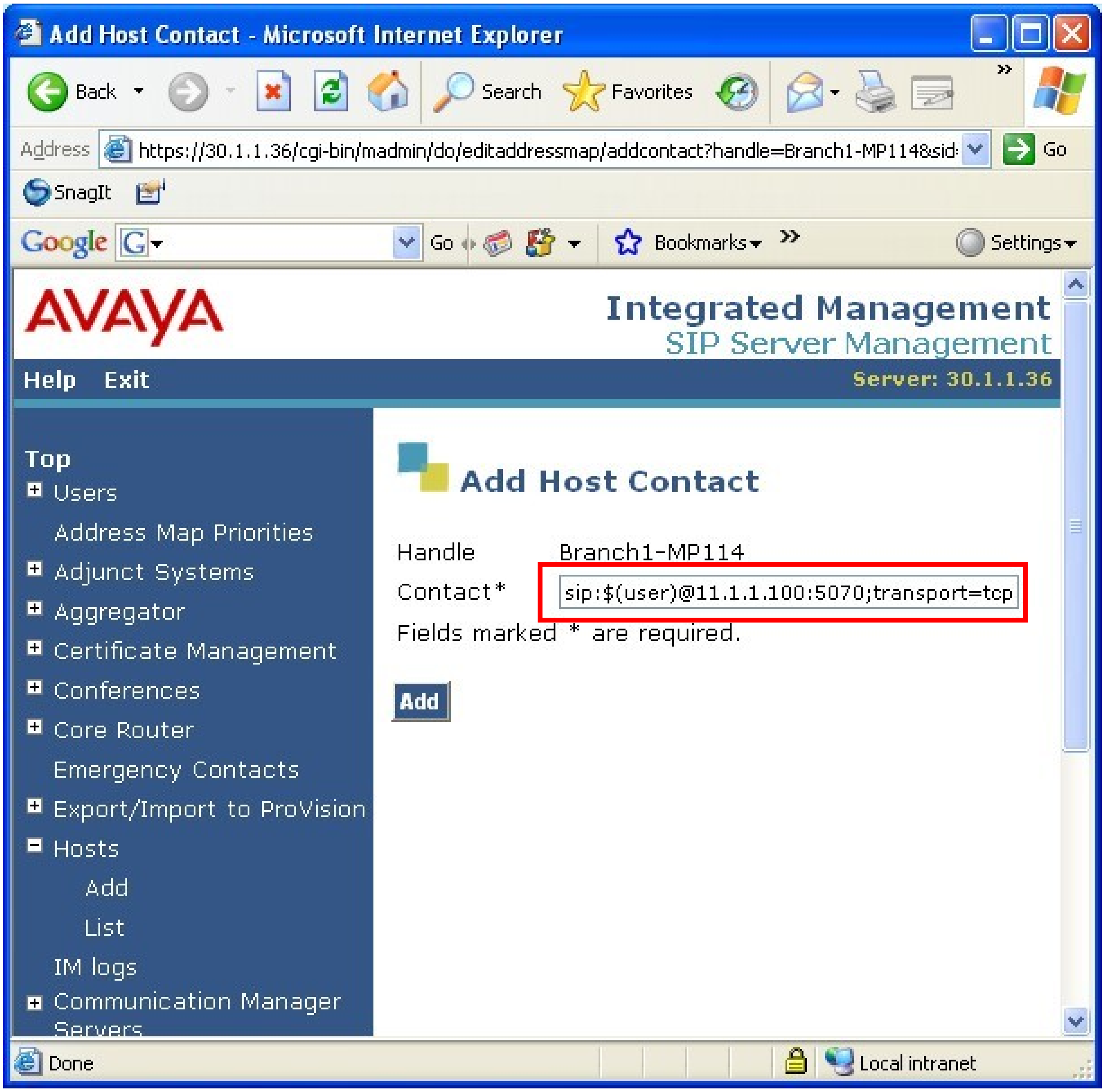
Enter a descriptive name and the pattern to match. The screen below shows the Host Address Map for Branch 1 matching on the 609 area code.



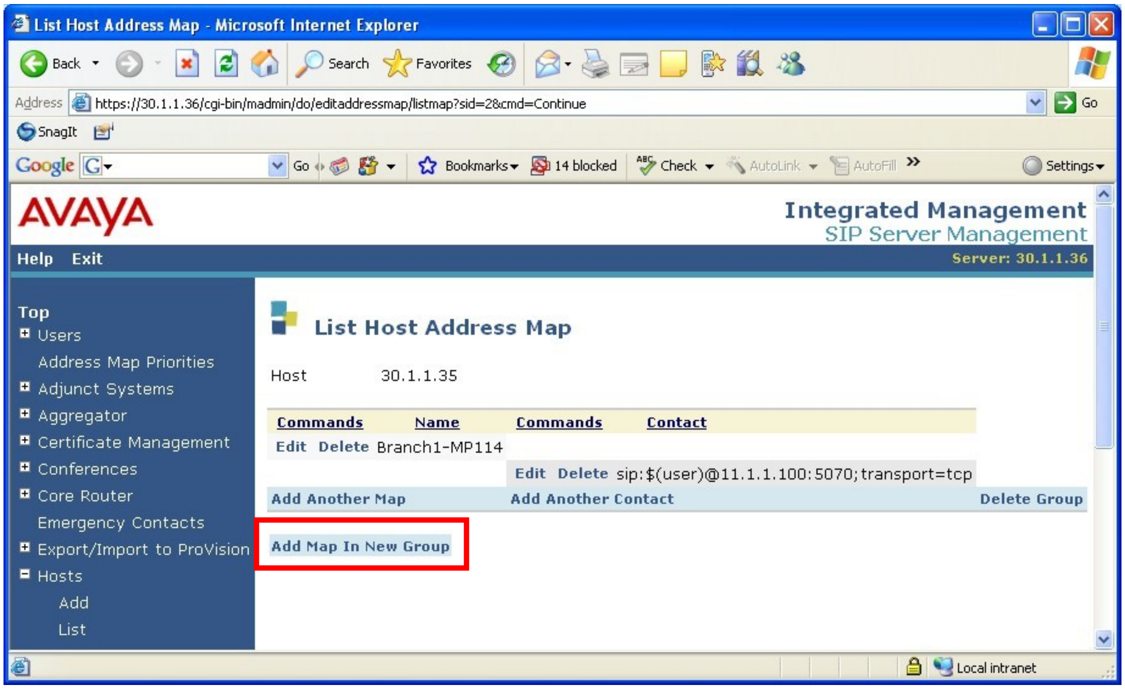
Select **Add Another Contact**.



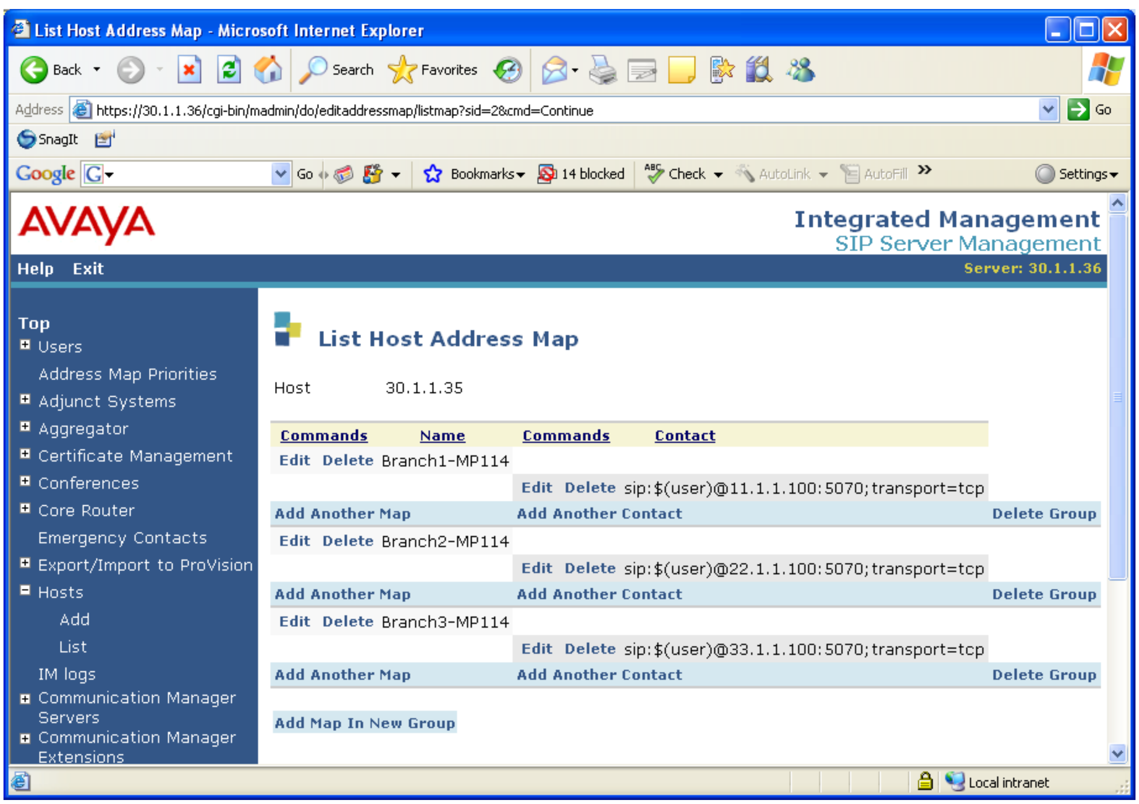
Following the format shown in the Contact field below, enter the AudioCodes MP-114 Gateway IP address, SIP listening port and transport method. The “sip:$(user)@” will be the same for all Host Address Map entries.



Select **Add Map In New Group** to create the Host Address Map for the next branch.



The following screen shows the Host Address Map entries for all three branches of the sample configuration.



5. Avaya Communication Manager

This section shows the necessary steps to configure Avaya Communication Manager to support the Avaya Communication Manager Survivable SIP Gateway Solution in a Distributed Trunking scenario. It is assumed that the basic configuration on Avaya Communication Manager and the required licensing has already been administered. See [3] for additional information. All commands discussed in this section are executed on Avaya Communication Manager using the System Access Terminal (SAT).

5.1. Locations

The locations of each branch as well as Headquarters must be defined within Avaya Communication Manager using the **change locations** command. The values used in the sample configuration are shown below. The location number, name and local area code (NPA) 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  16
                                   LOCATIONS

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

 Loc  Name           Timezone Rule  NPA  ARS  Atd Loc  Disp   Prefix   Proxy Sel
 No                   Offset             FAC  FAC Parm Parm              Rte Pat
 1:  HQ              + 00:00   0    201           1    1
 2:                      :
 3:                      :
 4:                      :
 5:                      :
 6:                      :
 7:                      :
 8:                      :
 9:                      :
10:                      :
11:  Branch 1        + 00:00   0    609           1    1
12:  Branch 2        + 00:00   0    732           1    1
13:  Branch 3        + 00:00   0    908           1    1
14:                      :
```

5.2. IP Codec Set

The voice codec to be used is defined in the IP Codec Set form. For the sample configuration, a single codec set is used with a single codec defined. The **change ip-codec-set** command is shown below to define Codec Set 1 where the G.711MU codec is entered.

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

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2:

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

On **Page 2 of 2,** set the "FAX" Mode to "t.38-standard". This is required for the T.38 fax interoperability testing where a group 3 (G3) or super group 3 (SG3) fax device is connected to the FXS port of AudioCodes MP-114/118 SIP Media Gateway at the branch.

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? y
              Maximum Call Rate for Direct-IP Multimedia:  5120:Kbits
     Maximum Call Rate for Priority Direct-IP Multimedia:  5120:Kbits

                    Mode                 Redundancy
    FAX             t.38-standard         0
    Modem           off                   0
    TDD/TTY         US                    3
    Clear-channel   n                     0
```

5.3. IP Network Region

IP Network Regions are defined for each branch location as well as the Headquarters location as defined in **Table 1** using the **change ip-network-region** command. The IP Network Regions are mapped to the Locations previously created. The values used in the sample configuration for Headquarters IP Network Region 1 are shown below. The Location, Name, Codec Set and Authoritative Domain field values shown are specific to the sample configuration. All remaining fields have been left at default values. The Authoritative Domain is the SIP domain name defined within the Avaya SES and used throughout the enterprise for SIP communications.

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: retail.com
    Name: Retail HQ
```

```
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? y
   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 values used in the sample configuration for Branch 2 IP Network Region 12 are shown below. The Location, Name, Codec Set and Authoritative Domain field values shown are specific to the sample configuration. All remaining fields have been left at default values. Follow the same steps to create the IP Network Regions for the remaining branch locations.

```
change ip-network-region 12                                    Page   1 of  19
                               IP NETWORK REGION
  Region: 12
Location: 12      Authoritative Domain: retail.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. For example, Codec Set 1 is specified for connections between network region 12 and network region 1.

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

                    Inter Network Region Connection Management

 src dst codec direct   WAN-BW-limits   Video      Intervening      Dyn
 rgn rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC IGAR AGL
 12  1    1     y    NoLimit                                             n   all
 12  2
```

5.4. IP Network Map

IP addresses are used to associate a device with a specific IP Network Region. The IP Network Region can be associated with a specific Location as previously described. The **change ip-network-map** command is used to perform the IP address to IP Network Region mapping. The IP Address Mapping used in the sample configuration is shown below based on the information from **Table 1**. In this case, the full subnet for each location is entered with the corresponding IP Network Region number.

```
change ip-network-map                                              Page   1 of  32
                            IP ADDRESS MAPPING
                                                                 Emergency
                                       Subnet                    Location
 From IP Address   (To IP Address   or Mask)  Region    VLAN   Extension
 30 .1  .1  .0        .   .   .        24      1         n
 11 .1  .1  .0        .   .   .        24      11        n
 22 .1  .1  .0        .   .   .        24      12        n
 33 .1  .1  .0        .   .   .        24      13        n
```

5.5. Node Names

Use the **change node-names ip** command to add host name and IP address entries to Avaya Communication Manager. Entries in the IP Node Names field simplify configuration on other forms, such as the Signaling Group form. The entries added to the IP Node Names form in the sample configuration are shown below.

```
change node-names ip                                               Page   1 of   1
                                      IP NODE NAMES
    Name                IP Address
C-LAN                 30.1.1.4
Ses-home1             30.1.1.35
```

5.6. SIP Trunks

Two SIP Trunks are used between Avaya SES and Avaya Communication Manager. One SIP Trunk, "SES to Avaya Communication Manager" is used for call signaling flowing from Avaya SES to Avaya Communication Manager. This is considered to be an incoming flow to Avaya Communication Manager requesting feature and routing treatment. The other SIP Trunk is used for outgoing calls flowing from Avaya Communication Manager to Avaya SES, after feature invocation, call routing and digit manipulations have been performed.

Note that a single trunk can be used and it is not required to use two trunks. However, the use of two trunks provides the added flexibility to change trunk parameters independently between incoming and outgoing. Tracing call legs within Avaya Communication Manager is also simplified.

5.6.1. SES to Avaya Communication Manager

Use the **add signaling-group** command to create a SIP signaling group. The values used in the sample configuration are highlighted on the Signaling Group form shown below. All remaining fields have been left at default values.

```
add signaling-group 7                                              Page   1 of   1
```

```
                              SIGNALING GROUP

 Group Number: 7              Group Type: sip
                        Transport Method: tls

         IP Video? n


   Near-end Node Name: C-LAN                  Far-end Node Name: ses-home1
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1
      Far-end Domain: retail.com

                                              Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
                                                         IP Audio Hairpinning? n
         Enable Layer 3 Test? n
 Session Establishment Timer(min): 3              Alternate Route Timer(sec): 6
```

Use the **add trunk-group** command to create a SIP trunk group. The values used in the sample configuration are highlighted on the Trunk Group form shown below. All remaining fields have been left at default values.

```
add trunk-group 7                                               Page   1 of  21
                               TRUNK GROUP

Group Number: 7                    Group Type: sip           CDR Reports: y
  Group Name: SES to ACM                  COR: 1        TN: 1        TAC: 107
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                                      Signaling Group: 7
                                                    Number of Members: 10
```

The Telephone Event Payload Type value must be entered on **Page 4** of the Trunk Group form. This value is used to identify DTMF transmissions using RFC 2833. This value must match the value used for the AudioCodes MP-114 parameter named **RFC 2833 Payload Type**. This parameter can be found on the AudioCodes MP-114 Configuration menu under **Protocol Configuration → Protocol Definition → DTMF & Dialing**. A value of 127 is used in the sample configuration.

```
add trunk-group 7                                              Page   4 of  21
                             PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? n
             Telephone Event Payload Type: 127
```

5.6.2. Avaya Communication Manager to SES

Use the **add signaling-group** command to create a SIP signaling group. The values used in the sample configuration are highlighted on the Signaling Group form shown below. All remaining fields have been left at default values.

```
add signaling-group 8                                           Page   1 of   1
                                SIGNALING GROUP

 Group Number: 8              Group Type: sip
                        Transport Method: tls

         IP Video? n

   Near-end Node Name: C-LAN                  Far-end Node Name: ses-home1
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1
       Far-end Domain: retail.com
                                              Bypass If IP Threshold Exceeded? n

        DTMF over IP: rtp-payload              Direct IP-IP Audio Connections? y
                                                         IP Audio Hairpinning? n
        Enable Layer 3 Test? n
 Session Establishment Timer(min): 3              Alternate Route Timer(sec): 6
```

Use the **add trunk-group** command to create a SIP trunk group. The values used in the sample configuration are highlighted on the Trunk Group form shown below. All remaining fields have been left at default values.

```
add trunk-group 8                                                 Page   1 of  21
                                TRUNK GROUP

Group Number: 8                  Group Type: sip          CDR Reports: y
  Group Name: ACM to SES                COR: 1        TN: 1        TAC: 108
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

                                                      Signaling Group: 8
                                                    Number of Members: 10
```

The Telephone Event Payload Type value must be entered on **Page 4** of the Trunk Group form. This value is used to identify DTMF transmissions using RFC 2833. This value must match the value used for the AudioCodes MP-114 parameter named **RFC 2833 Payload Type**. This parameter can be found on the AudioCodes MP-114 Configuration menu under **Protocol Configuration → Protocol Definition → DTMF & Dialing**. A value of 127 is used in the sample configuration.

```
add trunk-group 8                                               Page   4 of  21
                              PROTOCOL VARIATIONS

                        Mark Users as Phone? n
              Prepend '+' to Calling Number? n
        Send Transferring Party Information? n
                   Network Call Redirection? n
               Telephone Event Payload Type: 127
```

5.7. Automatic Route Selection (ARS)

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

5.7.1. 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:
```

5.7.2. ARS Location Specific Digit Analysis

The **change ars analysis location x y** 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 Avaya Communication Manager for source based routing. The location specific ARS entries for each Branch are shown below. Route Pattern 8 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:    1

          Dialed          Total      Route     Call   Node   ANI
          String         Min  Max   Pattern    Type   Num    Reqd
    1609                 11   11     8         natl          n
```

```
change ars analysis location 12 1732                          Page   1 of   2

                             ARS DIGIT ANALYSIS TABLE
                                   Location:  12          Percent Full:    1

          Dialed           Total     Route     Call   Node  ANI
          String         Min  Max  Pattern    Type   Num   Reqd
    1732                 11   11    8         natl         n
```

```
change ars analysis location 13 1908                          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
    1908                 11   11    8         natl         n
```

5.7.3. ARS Global 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 for the branch originating the call. The global ARS table using 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 1.

Route Pattern 1 is configured to use a Trunk Group with terminates on the Avaya G650 Media Gateway at the Headquarters location for PSTN terminations. The configuration of Route Pattern 1, the associated PSTN Trunk Group and the Avaya G650 Media Gateway are out of scope of these Application Notes and are therefore not included.

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

          Dialed           Total     Route     Call   Node  ANI
          String         Min  Max  Pattern    Type   Num   Reqd
    1                    11   11    1         fnpa         n
    2                    7    7     2         hnpa         n
    225                  7    7     8         hnpa         n
    3                    7    7     2         hnpa         n
    333                  7    7     333       natl         n
    4                    7    7     2         hnpa         n
    411                  3    3     deny      svcl         n
    5                    7    7     2         hnpa         n
    53011                5    5     1         natl         n
    53342                5    5     6         natl         n
```

5.8. Route Pattern

Use the **change route-pattern** command to modify a route pattern for calls destined for the AudioCodes MP-114 Gateways via the Avaya SES. The changes made to Route Pattern 8 in the sample configuration are highlighted below. Route Pattern 8 uses SIP Trunk Group 8 and strips the leading digit from the dialed number. In the case of the sample configuration, this leaves the area code as the lead digits sent to the Avaya SES. This is required to match the Avaya SES Host Address Maps for routing to the proper AudioCodes MP-114 as described in **Section 4.3.**

```
change route-pattern 8                                          Page   1 of   3

                    Pattern Number: 8   Pattern Name: To Branch MP114
                             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: 8    0                    1                                          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
```

5.9. Add Stations

A station must be created on Avaya Communication Manager for each SIP User account created in Avaya SES which includes a provisioned Communication Manager Extension. The extension assigned to the Avaya Communication Manager station must match the Communication Manager Extension assignment in Avaya SES.

Use the **add station** command to add a station to Avaya Communication Manager. The **add station** command for an Avaya 9630 SIP Phone located at Branch 2 using extension 222-1011 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 2221011                                          Page   1 of   6
                                     STATION

Extension: 222-1011                    Lock Messages? n               BCC: 0
     Type: 9600SIP                     Security Code:                  TN: 1
     Port: IP                        Coverage Path 1:                 COR: 1
     Name: Branch 2 - User 1         Coverage Path 2:                 COS: 1
                                     Hunt-to Station:
STATION OPTIONS
                                         Time of Day Lock Table:
              Loss Group: 19        Personalized Ringing Pattern: 1
                                               Message Lamp Ext: 222-1011
            Speakerphone: 2-way             Mute Button Enabled? y
        Display Language: english              Expansion Module: n
 Survivable GK Node Name:
          Survivable COR: internal            Media Complex Ext:
   Survivable Trunk Dest? y                        IP SoftPhone? n

                                                       IP Video? n
                                            Customizable Labels? Y
```

Use the **add off-pbx-telephone station-mapping** command to designate the station created above as a SIP station. The add off-pbx-telephone station-mapping command for extension 222-1011 is shown below. It is configured to use SIP Trunk Group 7 created in **Section 5.6.1.**

```
add off-pbx-telephone station-mapping                        Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                       Selection   Set
 222-1011          OPS           -     2221011            7           1
```

Repeat the addition of stations and off-pbx telephone station-mappings for each user account added to Avaya SES. The following list command output summarizes the configuration relevant to the sample configuration. Each Avaya SIP Telephone at the branch (e.g., 222-1010 through 222-1012), each analog device connected to an FXS port on the AudioCodes gateway (e.g., 222-1020 and 222-1021), and the station corresponding to the "gateway user" (e.g., 222-0000) can be observed. The corresponding registration of these users to the Avaya SES is shown in **Section 8.3**.

```
list off-pbx-telephone station-mapping                                Page   1

                    STATION TO OFF-PBX TELEPHONE MAPPING

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

222-0000       OPS        2220000          1  /   7       both         all
222-1010       OPS        2221010          1  /   7       both         all
222-1011       OPS        2221011          1  /   7       both         all
222-1012       OPS        2221012          1  /   7       both         all
222-1020       OPS        2221020          1  /   7       both         all
222-1021       OPS        2221021          1  /   7       both         all
```

6. AudioCodes MP-114

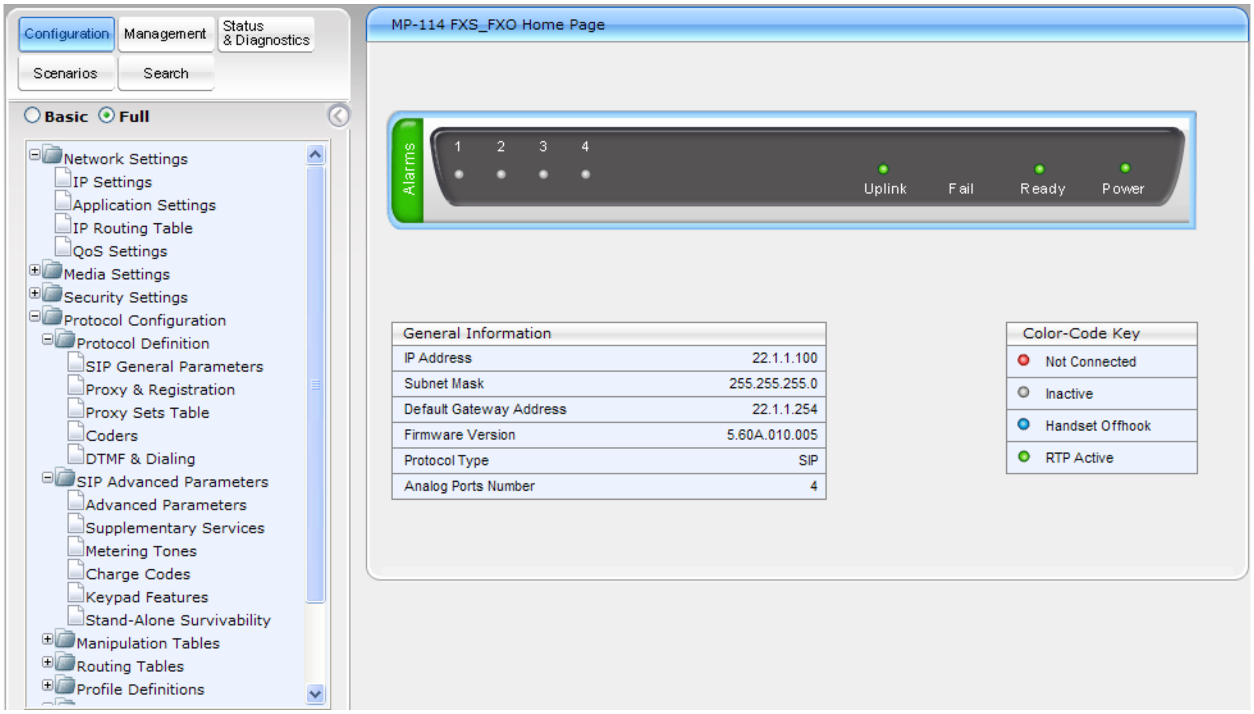
This section shows the necessary steps to configure the AudioCodes MP-114 Gateway to support the Avaya Communication Manager Survivable SIP Gateway Solution in a Distributed Trunking scenario. It is assumed that the basic configuration of the AudioCodes MP-114 has already been administered. See [5] and [6] for additional information.

All parameters of the AudioCodes MP-114's used in the sample configuration were set to factory default values prior to configuration. This icon  on the AudioCodes MP-114 configuration screens indicates the corresponding parameter value has been changed. All parameters with this icon shown in the following screens are relevant to the Avaya Communication 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.

6.1. MP-114 Access

From a web browser, enter the AudioCodes MP-114 IP address in the URL. A pop-up window similar to the one shown below will appear. Enter the appropriate User Name and Password.

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-Series SIP Telephone 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 3 was idle. The FXO line on port 4 was not configured).



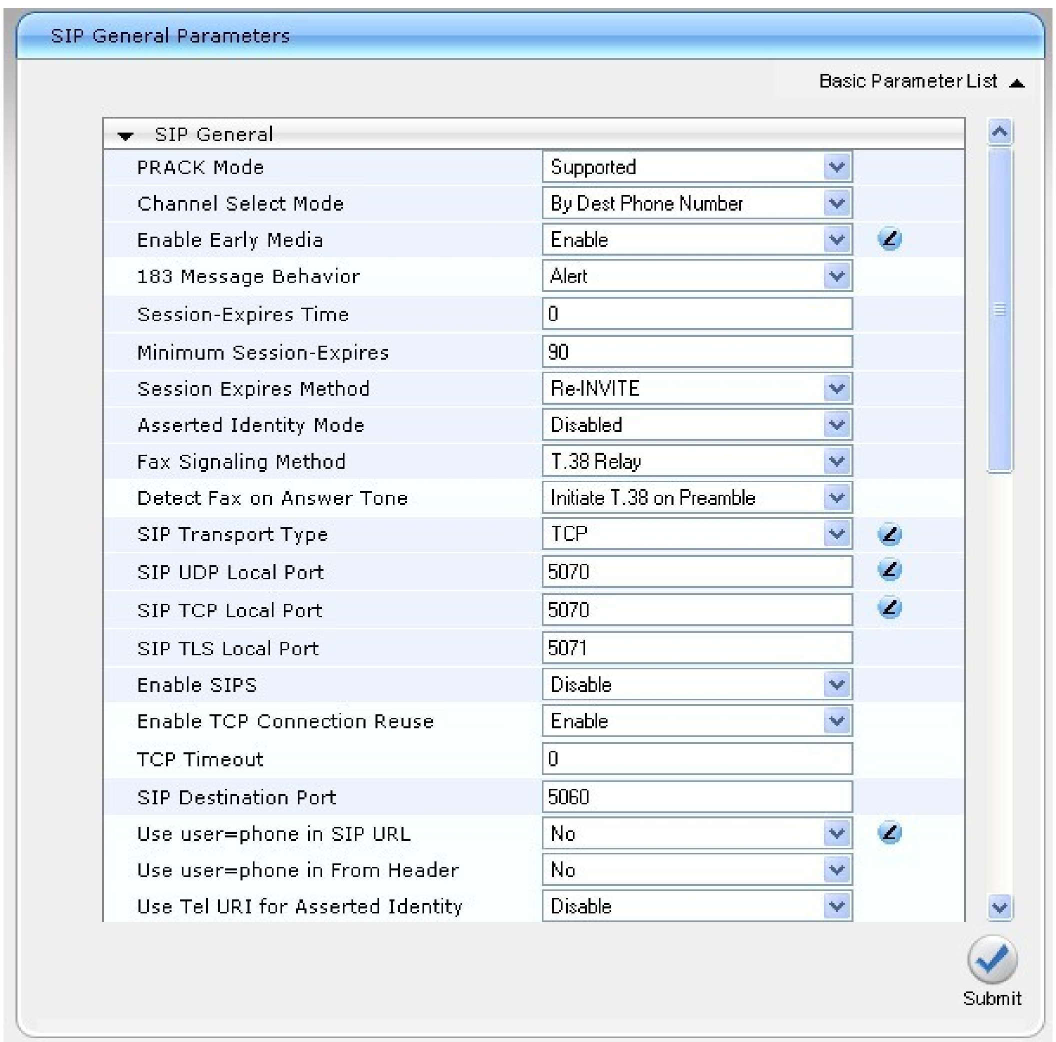
6.2. SIP General Parameters

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.

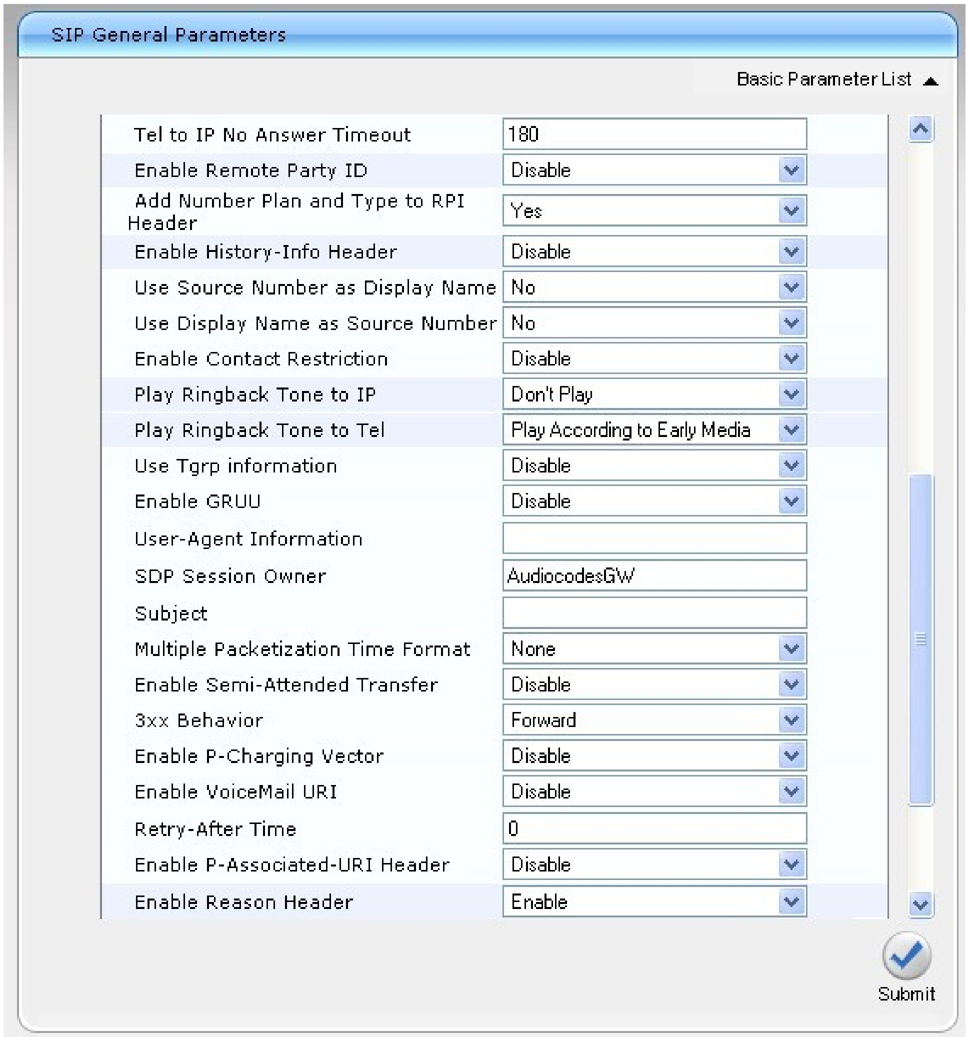
These key parameter values on this screen instruct the AudioCodes MP-114, when functioning as a media gateway, to use TCP as the transport and listen on port 5070 for SIP messages.

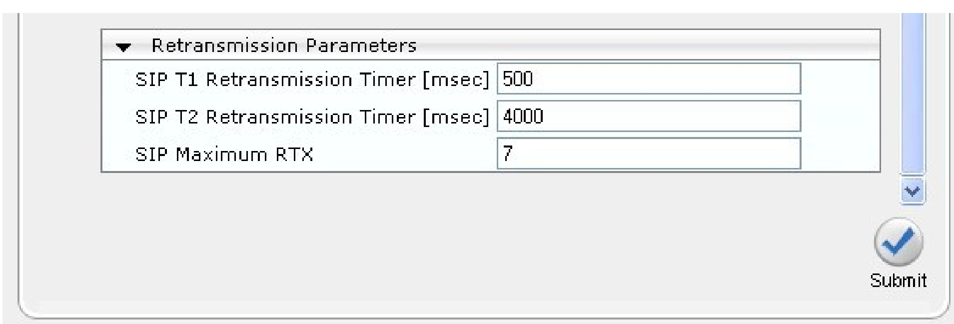
The parameter "Fax Signaling Method" should have the value "T.38 Relay" for the T.38 fax interoperability operations between the fax device connected to the FXS port of AudioCodes MP-114 and another fax device connected to either the Avaya Communication Manager port network and Avaya Media Gateway of the enterprise network or the PSTN line. AudioCodes MP-114 supports the T.38 fax relay over the IP using the SIP Re-INVITE message to negotiate the T.38 capabilities.

Once the "T.38 Relay" is set, the default values of parameters at another page "Fax/Modem/CID Settings" should be used for the T.38 Fax over IP. No additional configurations are needed.



The remaining fields of the SIP General Parameters screens maintain the default values. The continuation of the screens with default values are shown below as a reference.

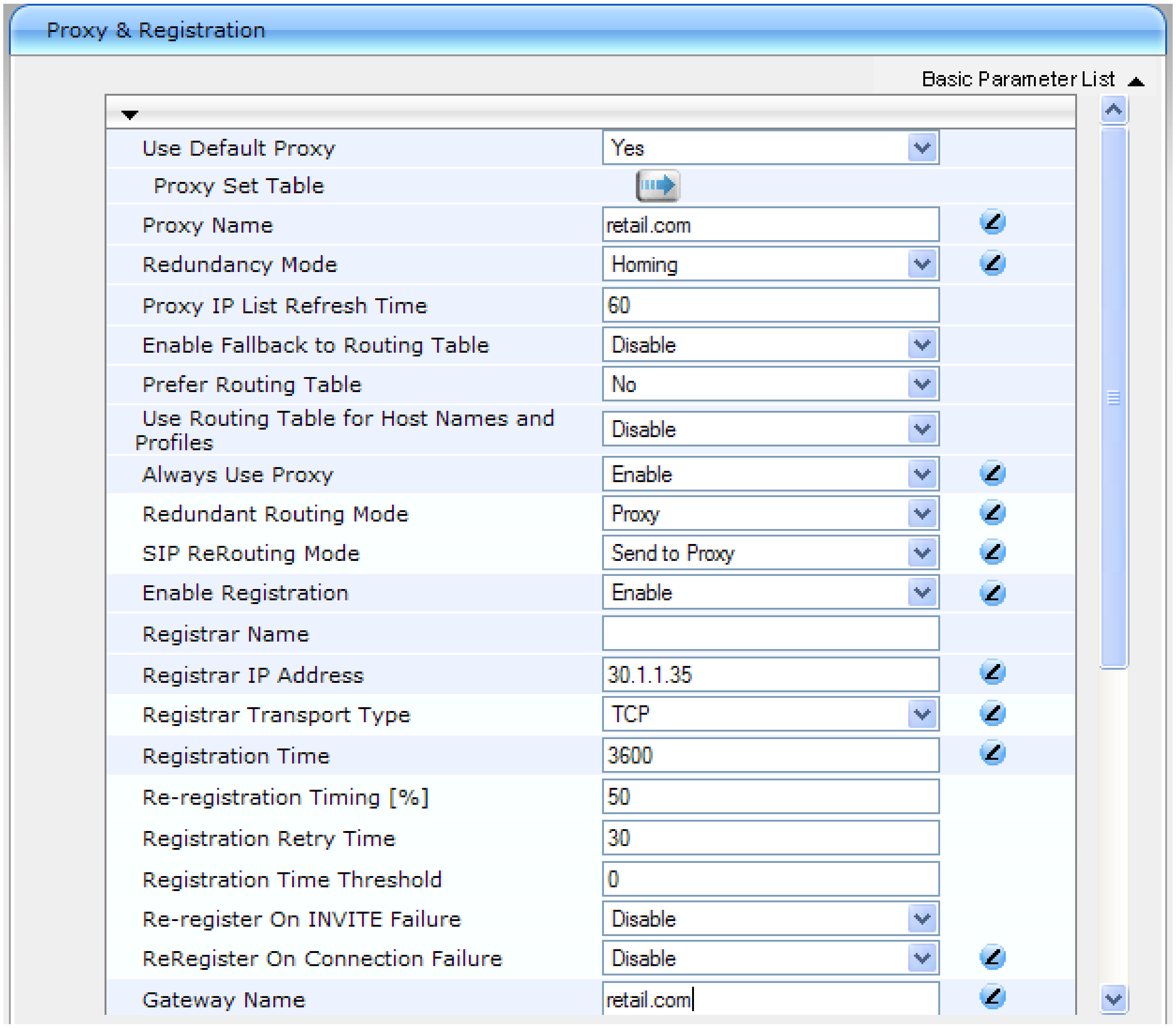




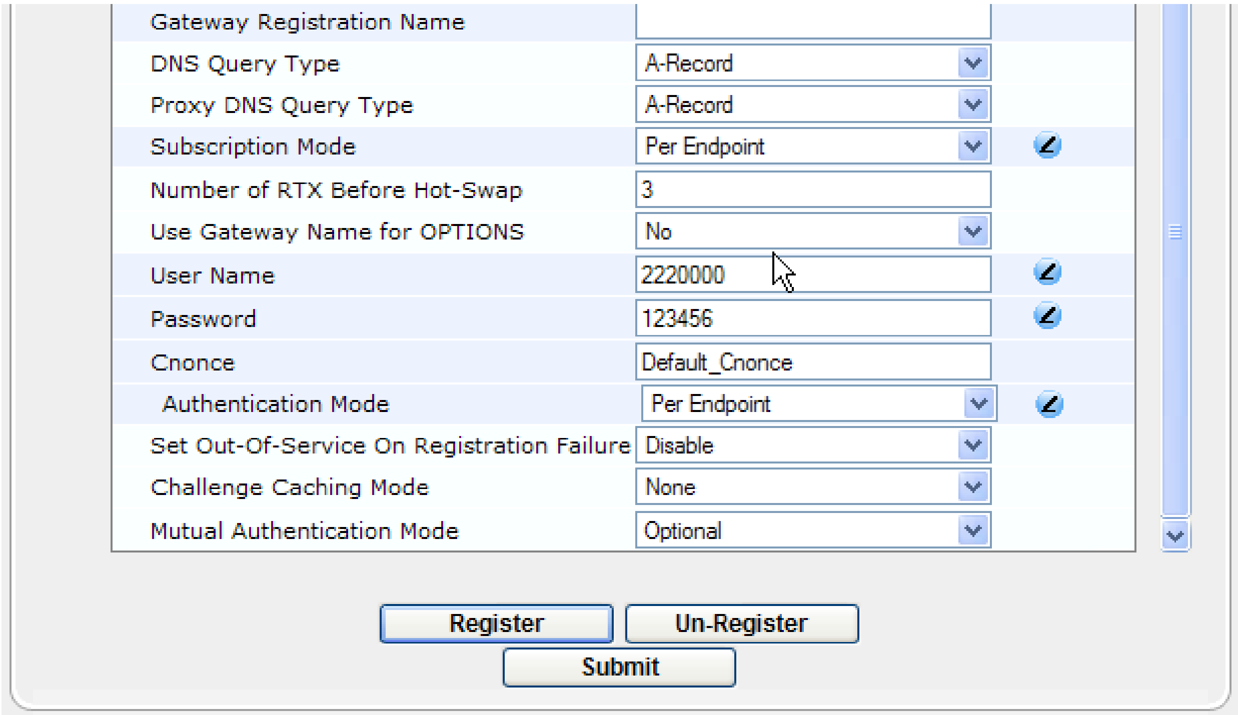
6.3. Proxy & Registration

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

The value of "retail.com" used throughout this form is the SIP Domain name used in the sample configuration and matches the SIP Domain name configured on Avaya SES and Avaya Communication Manager. These parameter values instruct the AudioCodes MP-114 to use the Avaya SES as a SIP Proxy server and a SIP Registrar. The AudioCodes MP-114 should register each FXS station with Avaya SES using TCP transport, refreshing every 3600 seconds.



The “User Name” and “Password” parameters must match the AudioCodes MP-114 user account created on the Avaya SES in **Section 4.2.2**.



6.3.1. Changes Required on Proxy & Registration with AudioCodes Version 5.6

Starting with AudioCodes version 5.6, the SAS (stand-alone survivability) application has been enhanced at AudioCodes MP-114 and the following fields should be set to:

Name	Value
Redundancy Mode	**Homing**
Redundant Routing Mode	**Proxy**

When the AudioCodes MP-114 is in survivability mode of the SAS application, the AudioCodes MP-114 Media Gateway serves as the SIP Proxy as defined in the “Proxy Sets Table”.

The Avaya SES defined as the “home” in the “Proxy Sets Table” takes over the SIP Proxy role once the SAS application of AudioCodes MP-114 detects the restore of the home proxy (Avaya SES Server) and changes back to the normal mode.

The “Proxy Sets Table” referred to by the SAS application should have the Avaya SES Server as its 1st SIP Proxy (home) followed by the AudioCodes MP-114 itself.

6.4. Proxy Sets Table

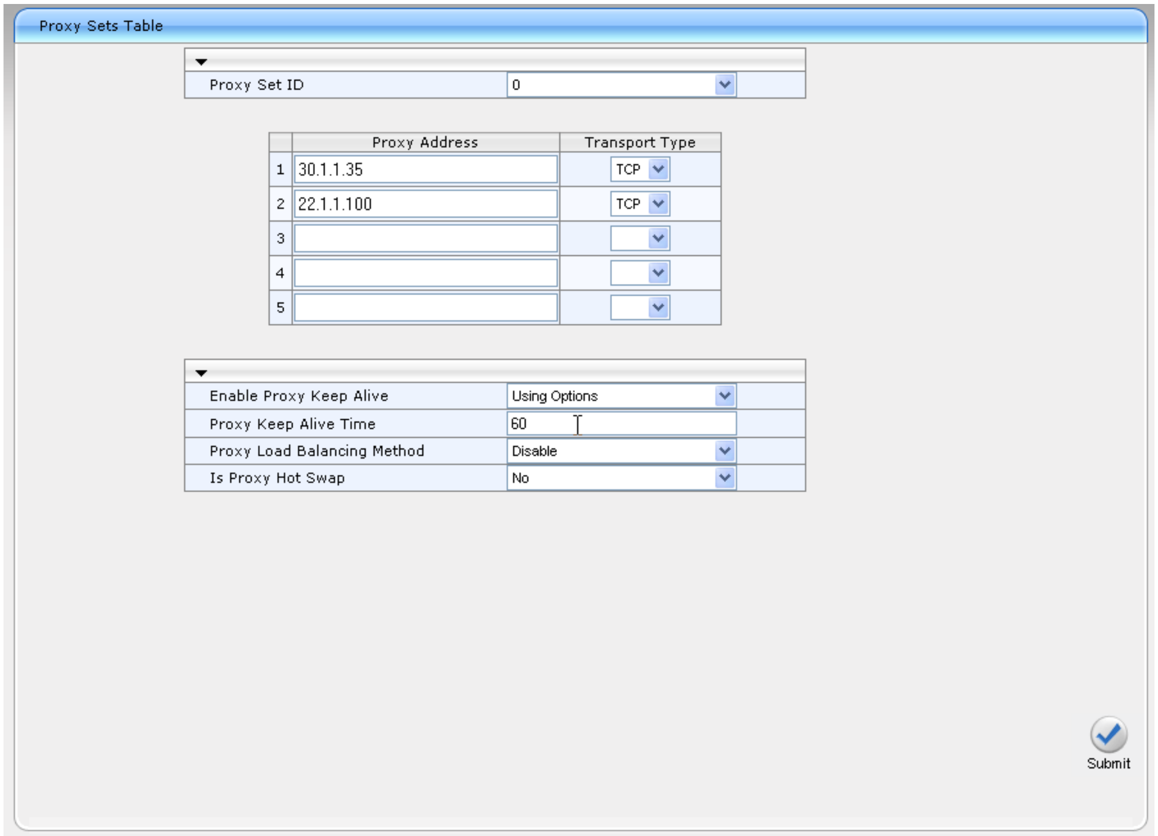
From the left navigation panel, navigate to the Proxy Sets Table screen by selecting **Protocol Configuration → Protocol Definition → Proxy Sets Table**. The values of the fields with an adjacent icon have changed from the default.

The Proxy Sets Table with “Proxy Set ID” set to “0” specifies a list of SIP Proxy servers the AudioCodes MP-114 is going to monitor for connectivity health to determine when to become active as a Normal Server or a Survivability Server. In this case, both Avaya SES and AudioCodes MP-114 should be administered and the Avaya SES is positioned as the first Proxy (Home, the Normal Server).

The mechanism used to monitor the Avaya SES is also specified. SIP Options is used in the sample configuration with the AudioCodes MP-114 default Proxy Keep Alive Time of 60 seconds. This results in the AudioCodes MP-114 sending SIP Options messages to the Avaya SES and using the response as an acknowledgement that the Avaya SES is accessible from the branch location. If a response to a SIP Options message is not received, the AudioCodes MP-114 will continue to attempt to contact the Avaya SES for 60 seconds, the Proxy Keep Alive Time value, and then activate its SAS survivable SIP server feature.

Enter the IP Address of the Avaya SES in the Proxy address table and select TCP from the Transport Type drop-down list. Following the Avaya SES, enter the IP address of AudioCodes MP-114 and its supported Transport Type.

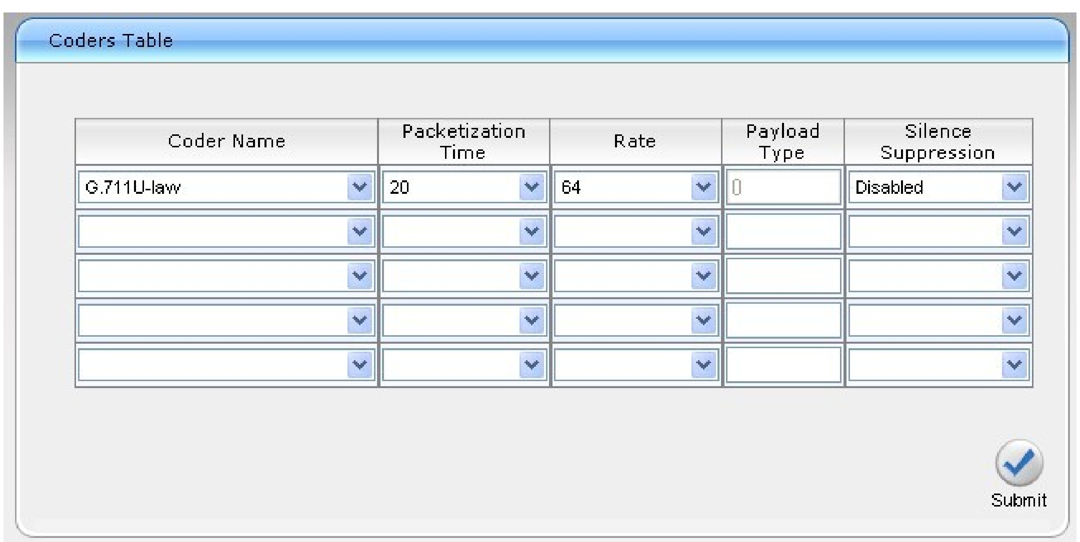
For Enable Proxy Keep Alive, select “Using Options” from the drop-down list.



6.5. Coders Table

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

Select the codec from the drop own list that matches the codec configured in Avaya Communication Manager.



6.6. 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 on the Avaya Communication Manager SIP Trunks in **Section 5.6.1** and **Section 5.6.2**. Specifically, the "Telephone Event Payload Type" on **Page 4** of the SIP Trunk Group form.

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

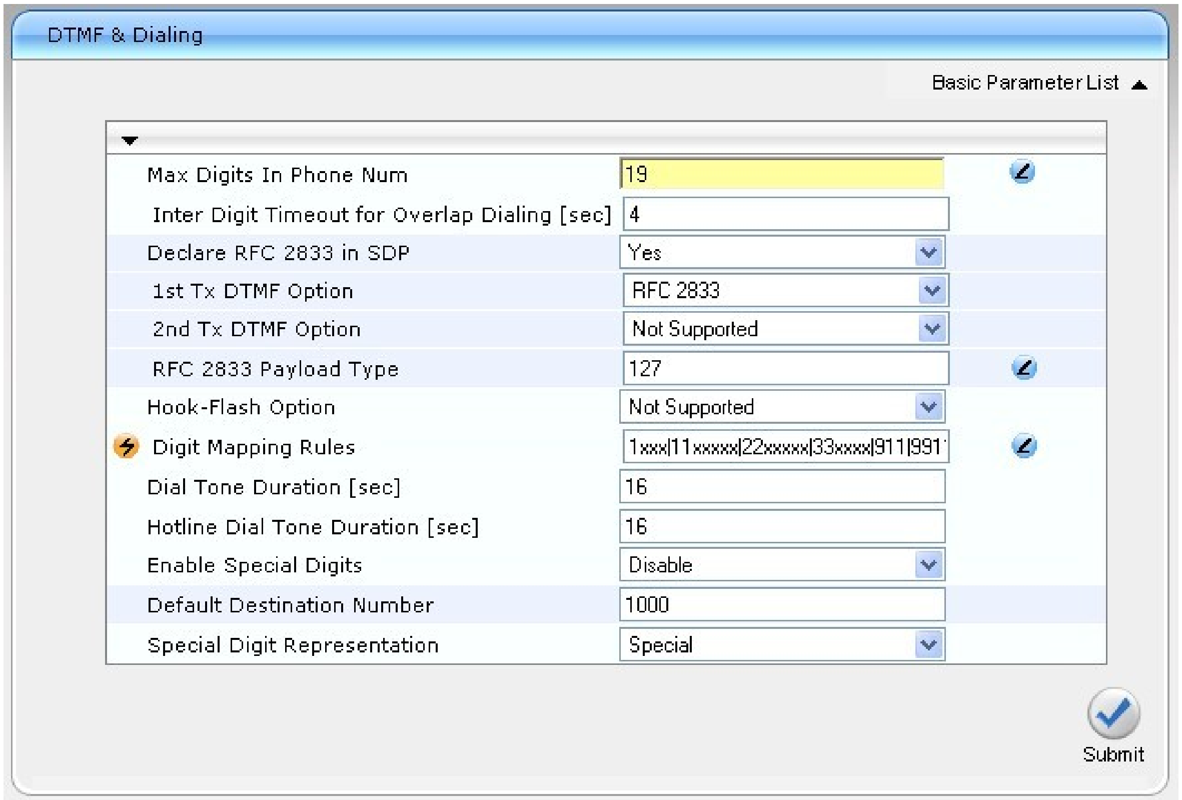
1xxx|11xxxxx|22xxxxx|33xxxxx|911|9911|91xxxxxxxxxx|9011x.T|732xxxxxxx

The details of the Digit Mapping Rule are captured in **Table 3**. Refer to [6] for additional information on digit mapping rules.

Digit String To Match	Sample Configuration Use
1xxx	Headquarters extensions
11xxxxx|22xxxxx|33xxxxx	Branch extensions
911|9911	Emergency dialing
91xxxxxxxxxx	North American Numbering Plan
9011x.T	International dialing
732xxxxxxx	Local Area Code for Branch 2

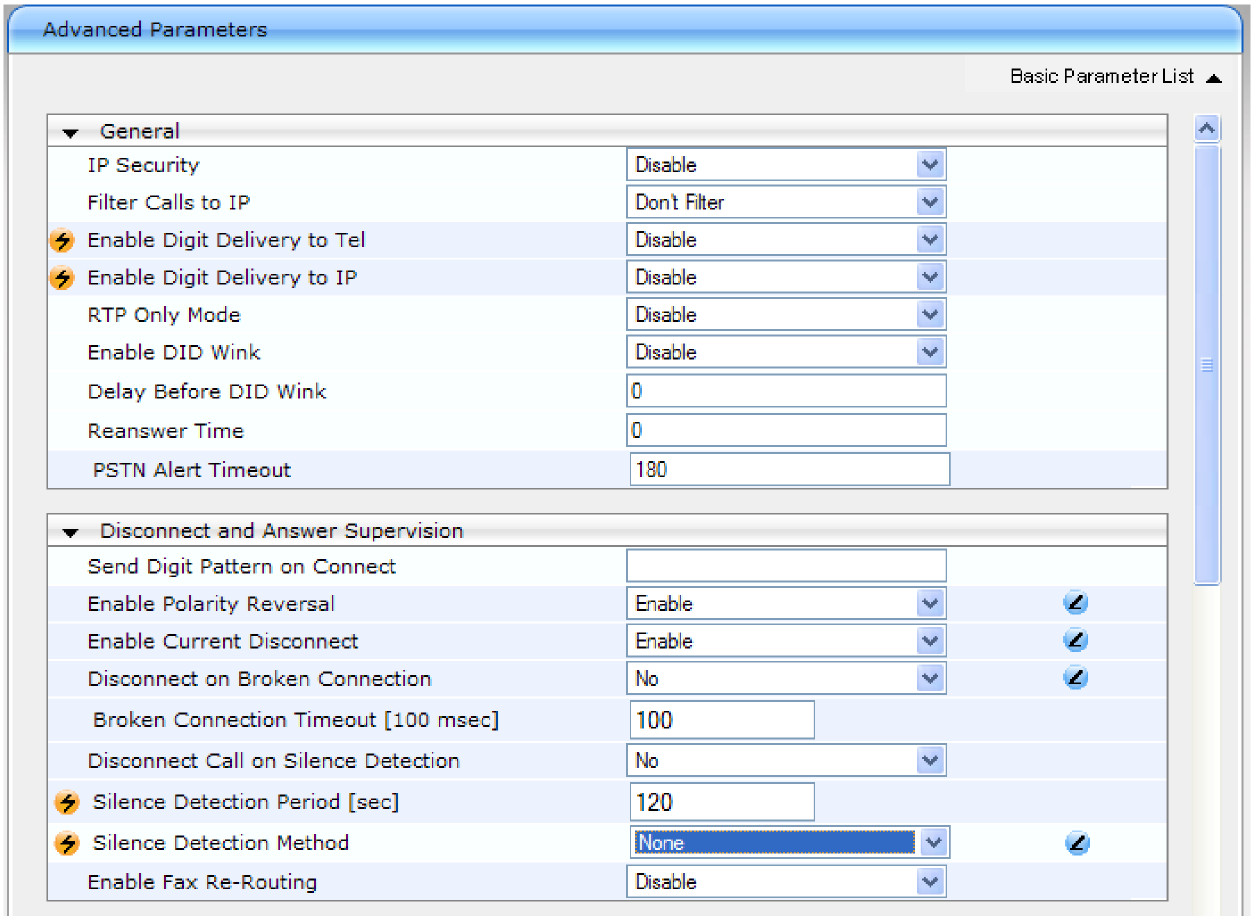
Table 3 – Digit Mapping Rule used in Sample Configuration

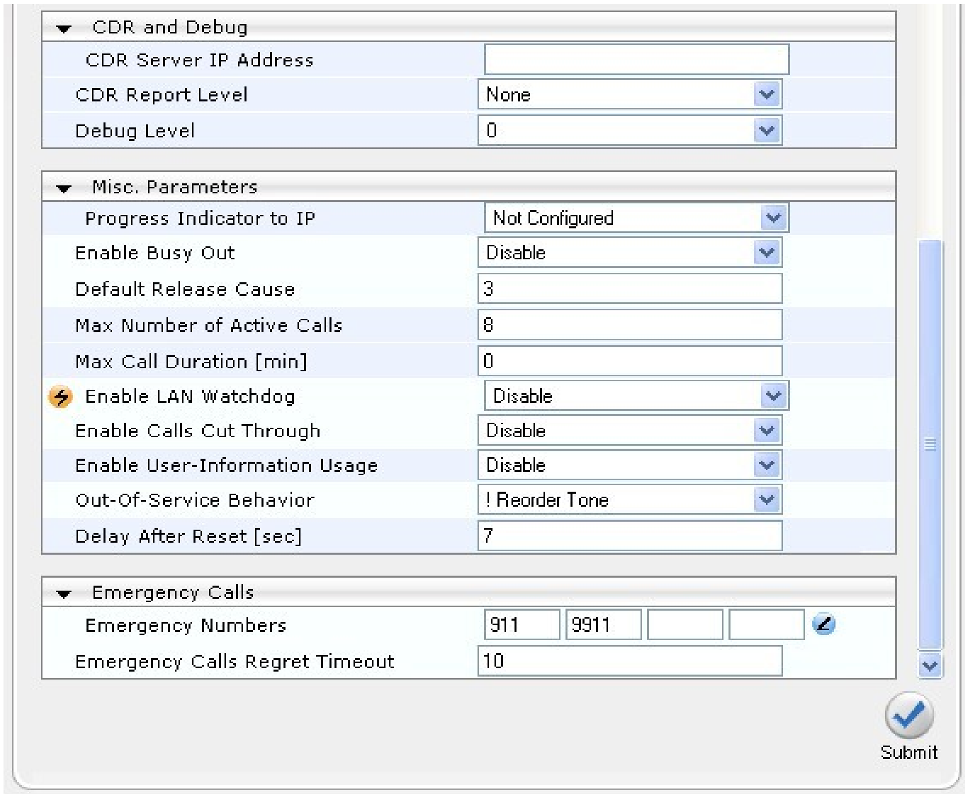
Note the Local Area Code entry used for Branch 1 is 609xxxxxxx and for Branch 3 is 908xxxxxxx.


DTMF & Dialing
Basic Parameter List
Max Digits In Phone Num
19
Inter Digit Timeout for Overlap Dialing [sec]
4
Declare RFC 2833 in SDP
Yes
1st Tx DTMF Option
RFC 2833
2nd Tx DTMF Option
Not Supported
RFC 2833 Payload Type
127
Hook-Flash Option
Not Supported
Digit Mapping Rules
1xxx|11xxxxx|22xxxxx|33xxxx|911|991
Dial Tone Duration [sec]
16
Hotline Dial Tone Duration [sec]
16
Enable Special Digits
Disable
Default Destination Number
1000
Special Digit Representation
Special
Submit

6.7. 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.





6.7.1. "Disconnect on Broken Connection" Parameter

The AudioCodes gateway provides a parameter called "Disconnect on Broken Connection". This parameter controls whether the AudioCodes gateway should release the active call if RTP packets are not received within a user-defined timeoute period. This timeout period is controlled by the value in the "Broken Connection Timeout [100 msec]" field. The "Disconnect on Broken Connection" parameter should be set to "No". The active call should not be released if RTP packets are not received within the timeout interval.

Avaya 96xx series SIP phones do not send any RTP packets when active calls are on hold. This hold state is initiated by pressing the [Hold]/[Transfer]/[Conference] softkey buttons on 96xx phones. If this parameter is not set to "No", the active call is dropped after being on hold for 10 seconds (default). This scenario happens when the AudioCodes gateway is in SAS mode.

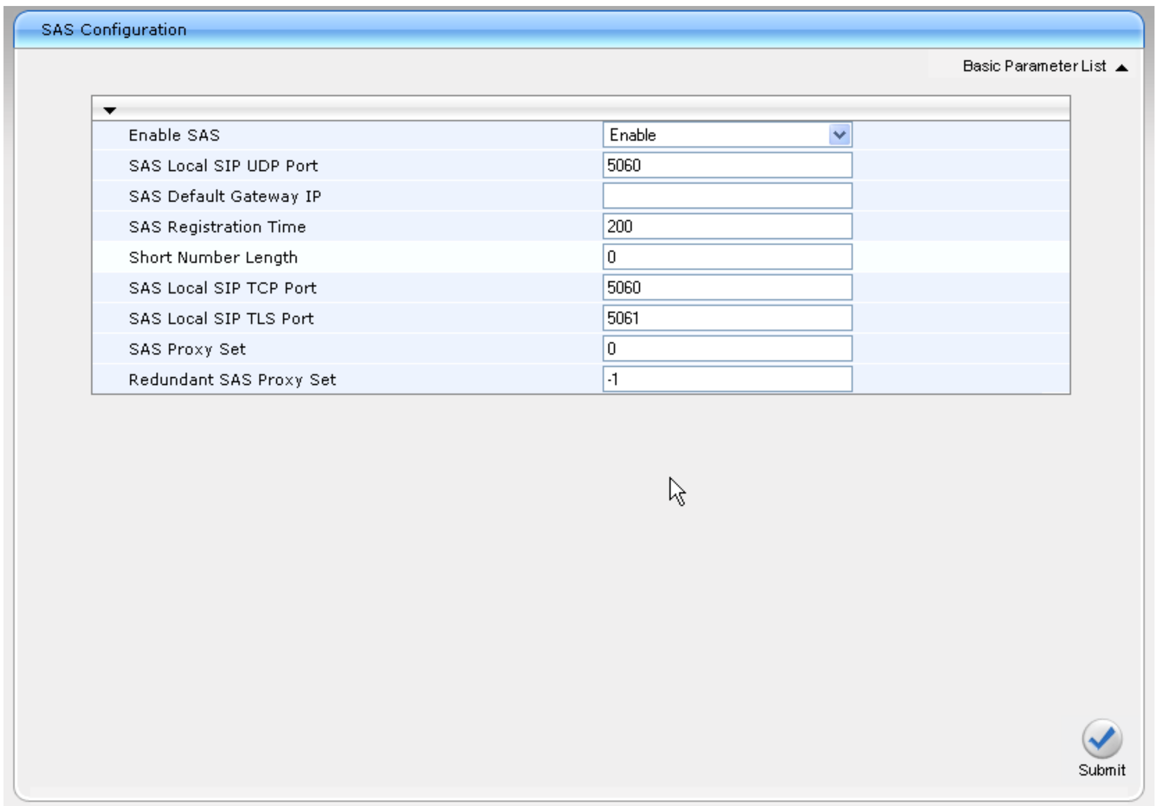
6.8. Stand-Alone Survivability

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

These key parameter values on this screen enable the AudioCodes MP-114 survivability feature, SAS. The SAS SIP Proxy and SIP Registrar will listen on TCP port 5060 for SIP messages. This must match the Avaya SES Survivable Call Processors configuration in **Section 4.1**.

AudioCodes software version 5.6 introduces two new fields in addition to the fields presented at its previous software version 5.4. The default values of new fields should be used. The “Proxy Sets Table” (see section 6.4) with Proxy Set ID of 0 is used as the SAS Proxy Set. Since the Redundant SAS Proxy Set is not used, set the value to “-1”.

Name	Value
SAS Proxy Set	**0**
Redundant SAS Proxy Set	**-1**



6.9. 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 string (for numbers matching the **Destination Prefix**) for IP to PSTN calls while in Survivability Mode. In Normal Mode, this is done by Avaya Communication Manager.

As an example, the dialed number 9 1-732-555-1111 would strip the 9 leaving 1-732-555-1111 presented to the PSTN via the AudioCodes MP-114 FXO interface.

Destination Phone Number Manipulation Table for IP -> Tel Calls

Table Index: 1-10

	Destination Prefix	Source Prefix	Source IP	Stripped Digits Number	Prefix (Suffix) to Add	Number of Digits to Leave
1	917	*	*	1		
2						
3						
4						
5						
6						
7						
8						
9						
10						

6.10. IP to Hunt Group Routing

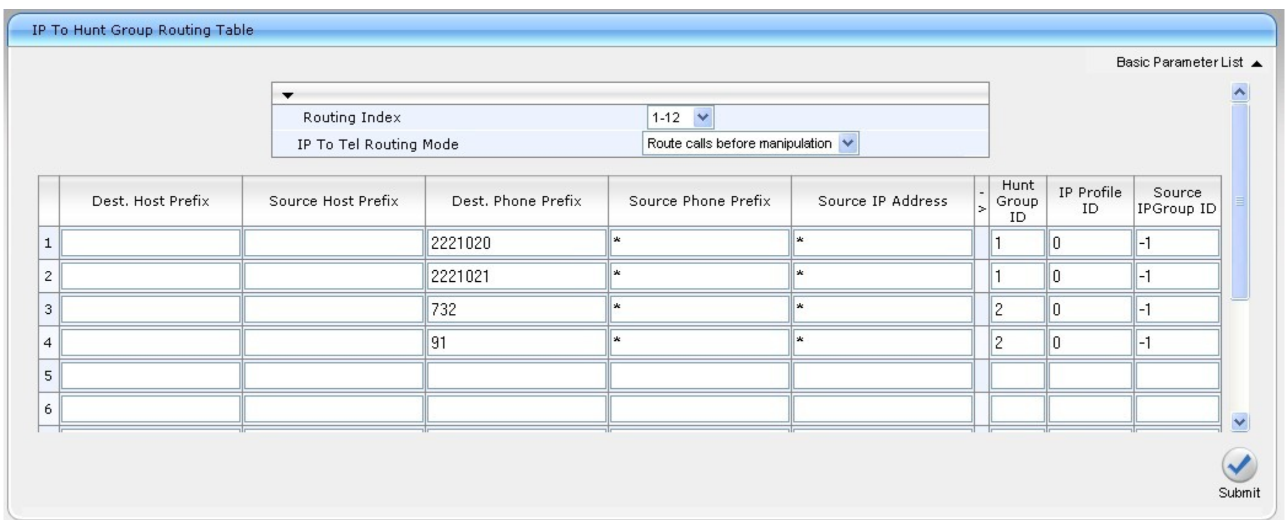
From the left navigation panel, navigate to the IP to Hunt Group Routing Table screen by selecting **Protocol Configuration → Routing Tables → IP to Trunk Group Routing**.

The entries in this table are used by the AudioCodes MP-114 Gateway 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 Avaya Communication Manager. The prefix of the called number is used to determine the AudioCodes MP-114 Hunt Group to route the call to. In the sample configuration, the FXS analog phone numbers are entered explicitly and route to Hunt Group ID 1. The 732 and 91 prefixes route 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 6.14**. **Table 4** shows a summary of the Hunt Group assignments.

Channel	Hunt Group ID
FXS 1	1
FXS 2	1
FXO 3	2
FXO 4	Un-assigned

Table 4 – Hunt Group Assignments


IP To Hunt Group Routing Table

Basic Parameter List

Routing Index: 1-12

IP To Tel Routing Mode: Route calls before manipulation

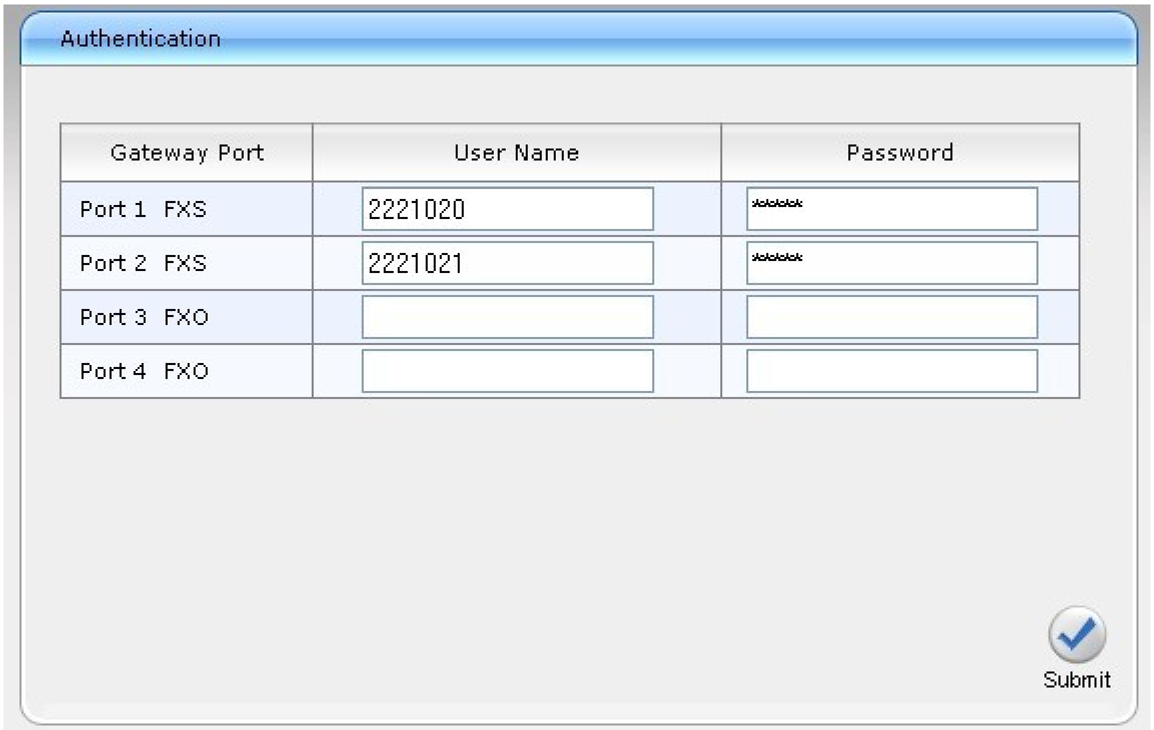
	Dest. Host Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Hunt Group ID	IP Profile ID	Source IPGroup ID
1			2221020	*	*		1	0	-1
2			2221021	*	*		1	0	-1
3			732	*	*		2	0	-1
4			91	*	*		2	0	-1
5									
6									

Submit

6.11. Authentication

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

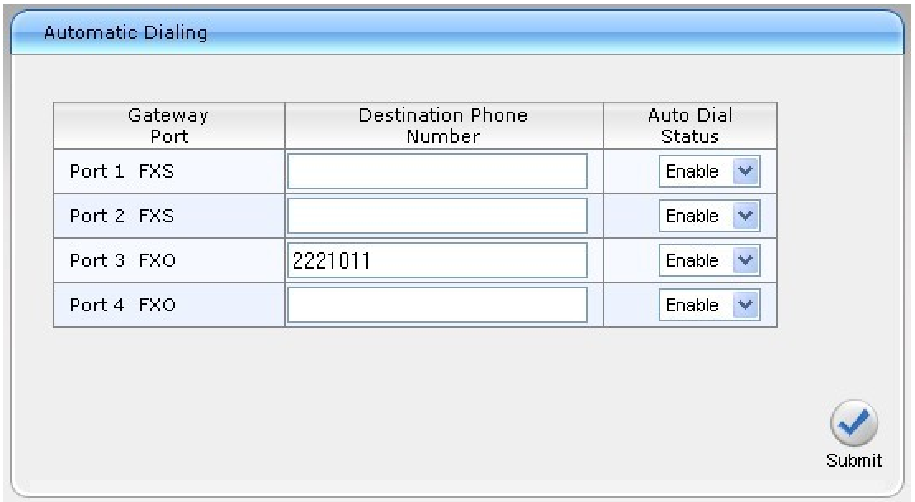
Enter the SIP user name and password that match the AudioCodes MP-114 FXS Analog Phone User Account created on Avaya SES in **Section 4.2.3**.



6.12. Automatic Dialing

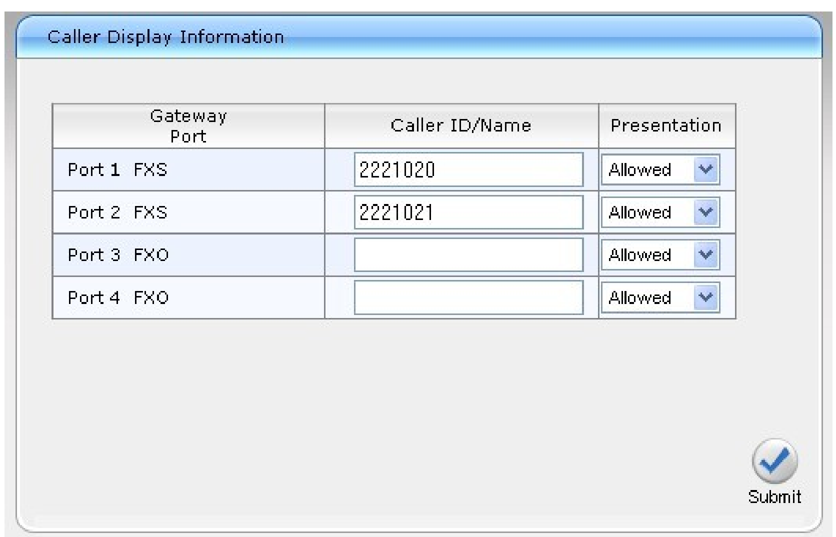
From the left navigation panel, navigate to the Automatic Dialing screen by selecting **Protocol Configuration → Endpoint Settings → Automatic Dialing**.

Enter the Branch extension to be automatically dialed on incoming calls from the PSTN to the Branch on the AudioCodes MP-114 FXO interface. The extension of an Avaya 9600 SIP Phone located at Branch 2 is used in the sample configuration.



6.13. 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 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 Avaya Communication Manager.



6.14. 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-114 as well as the associated Hunt Group ID. Channels 1 and 2 are the FXS interfaces. Channels 3 and 4 are the FXO interfaces.

	Channel(s)	Phone Number	Hunt Group ID	Tel Profile ID
1	1	2221020	1	1
2	2	2221021	1	1
3	3	2220000	2	1
4				

6.15. Hunt Group Settings

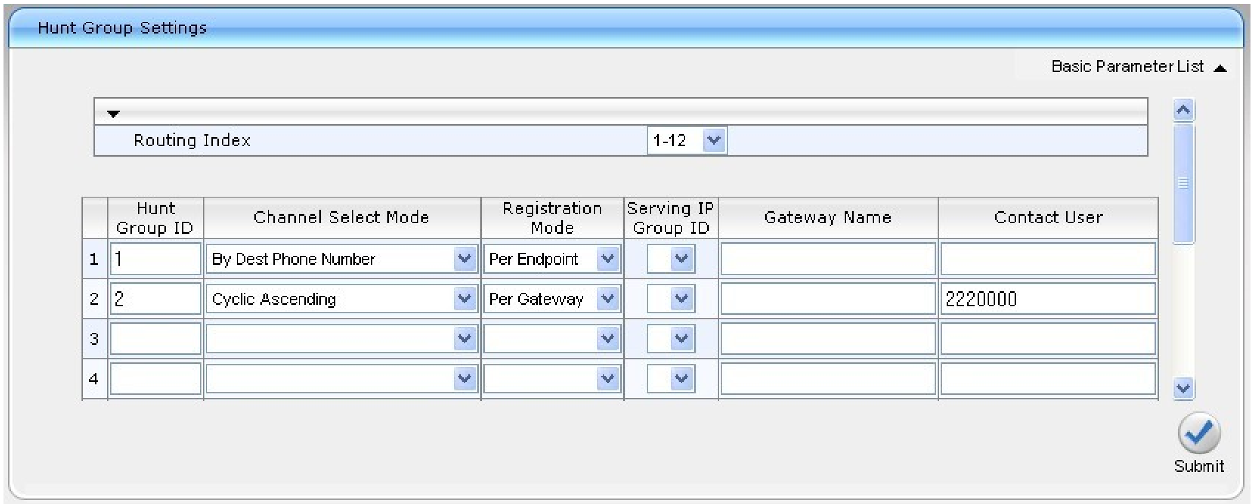
From the left navigation panel, navigate to the Hunt Group Settings screen by selecting **Protocol Configuration → Hunt/IP Group → Hunt Group Settings**.

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.

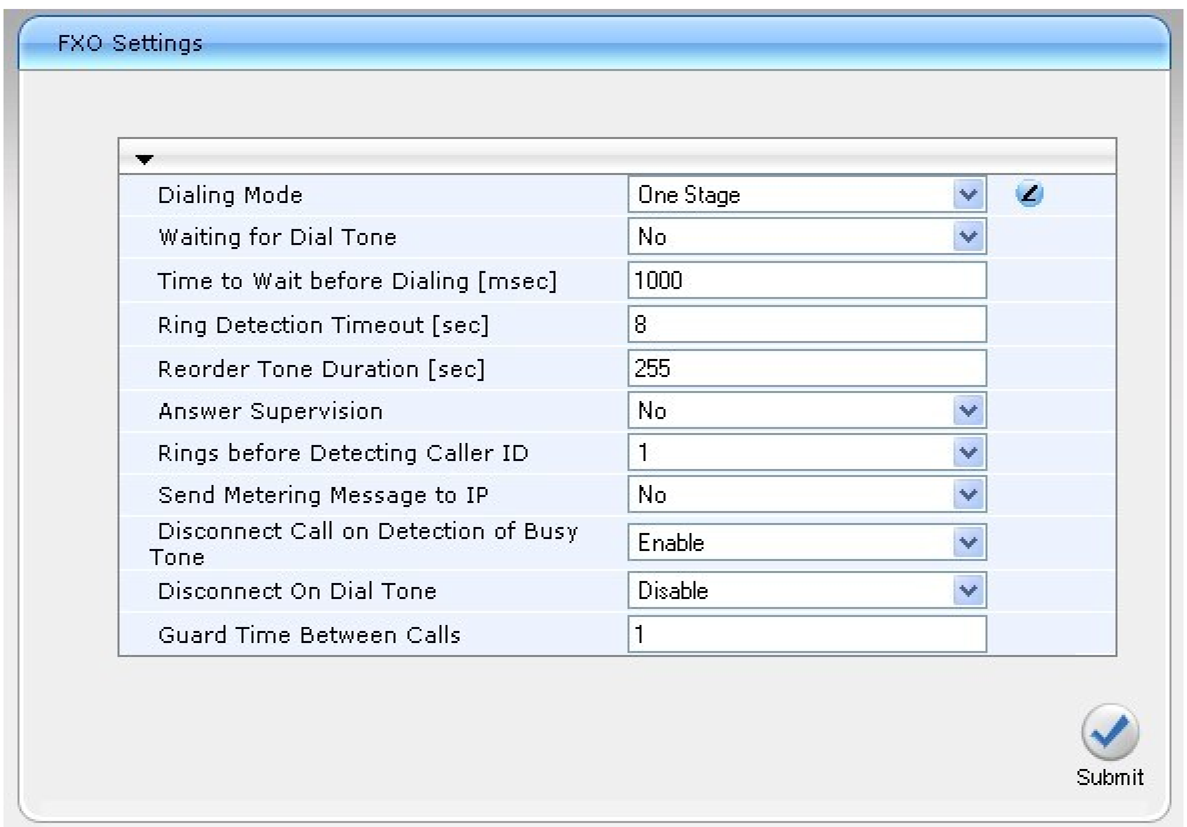
Hunt Group 2, containing 1 FXO interface to the PSTN, is configured to select any interfaces in this Hunt Group in a Cyclic Ascending order. Cyclic Ascending is the default. Since only one FXO interface is configured for Hunt Group 2 in the sample configuration, no channel cycling is occurring.

The Contact User field for the Hunt Group 2 entry contains the SIP extension of the MP-114. This value is used in the Contact field of SIP INVITE and Registration messages from the MP-114 Gateway.



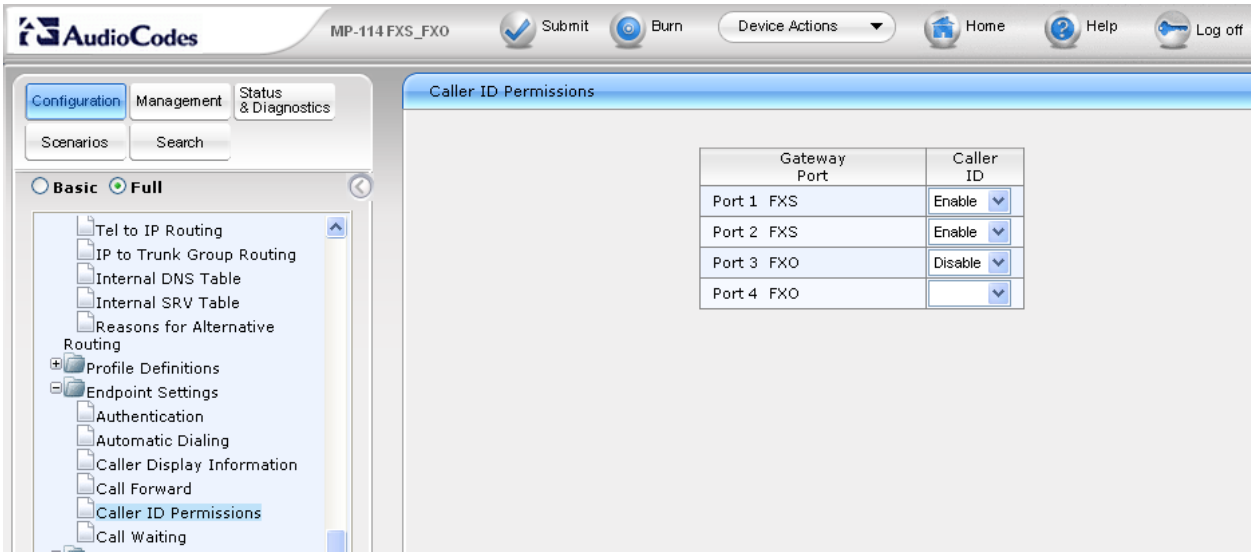
6.16. Advanced Applications → FXO Settings

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



6.17. Disable Caller ID on Analog FXO

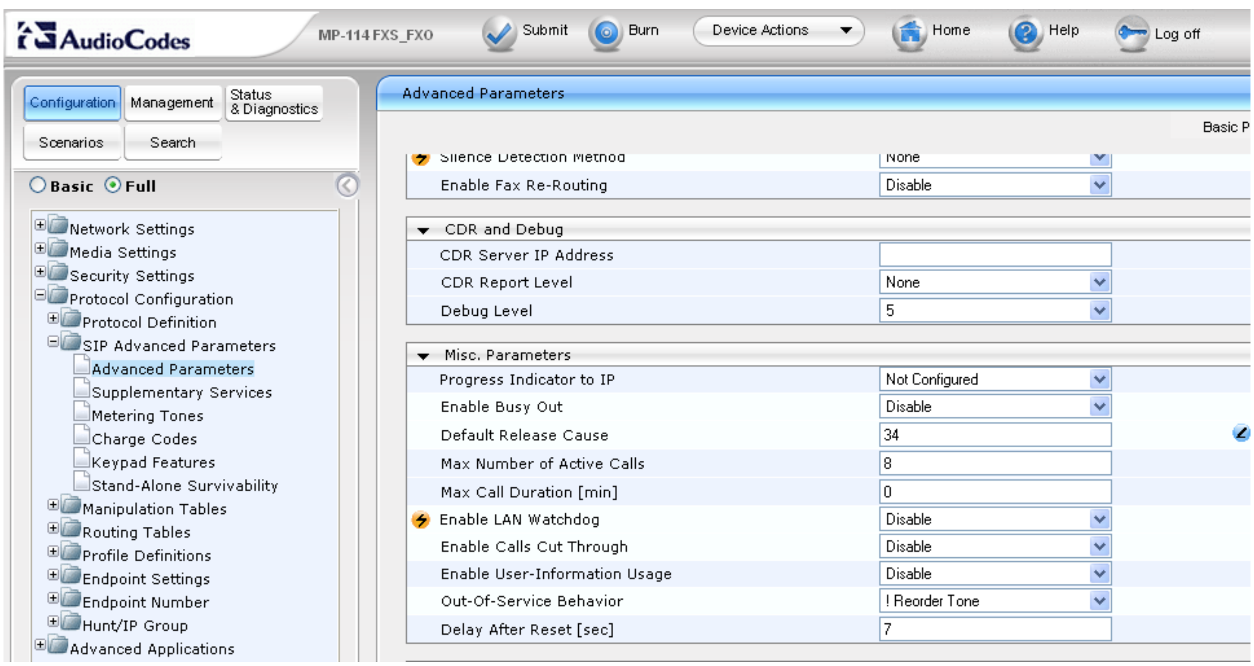
In the sample configuration, incoming caller ID arriving from the PSTN via the AudioCodes FXO port is not supported in Normal Mode. It is recommended that caller id be explicitly disabled for the FXO ports as follows. Navigate to **Endpoint Settings → Caller ID Permissions**. For each FXO port, select Disable from the drop-down menu, and click **Submit** in the lower right corner of the screen (not shown in the abbreviated screen below).



6.18. Change SIP Response for All FXO Busy Condition (Optional)

Avaya Communication Manager supports a feature called Look-Ahead Routing (LAR) that enables an outbound trunk call to complete automatically using an alternate trunk if errors are encountered after initial trunk selection occurs. In the distributed trunking configuration, local PSTN calls from branch users are routed by Avaya Communication Manager through Avaya SES to the AudioCodes Gateway, so that an FXO port on the gateway can be used to complete the local call. If all FXO ports on the gateway are already in use, by default, the AudioCodes Gateway will respond with a SIP "404" response, which is not among the SIP responses that would trigger Avaya Communication Manager LAR. If Avaya Communication Manager LAR is desired to allow the call to complete automatically using an alternate trunk in the route pattern (e.g., a trunk at headquarters), the AudioCodes Gateway response to "all FXO busy" conditions can be changed using the procedure in this section. For more information on Avaya Communication Manager Look-Ahead Routing as it applies to SIP Trunk Groups, please see reference [5] in **Section 10**.

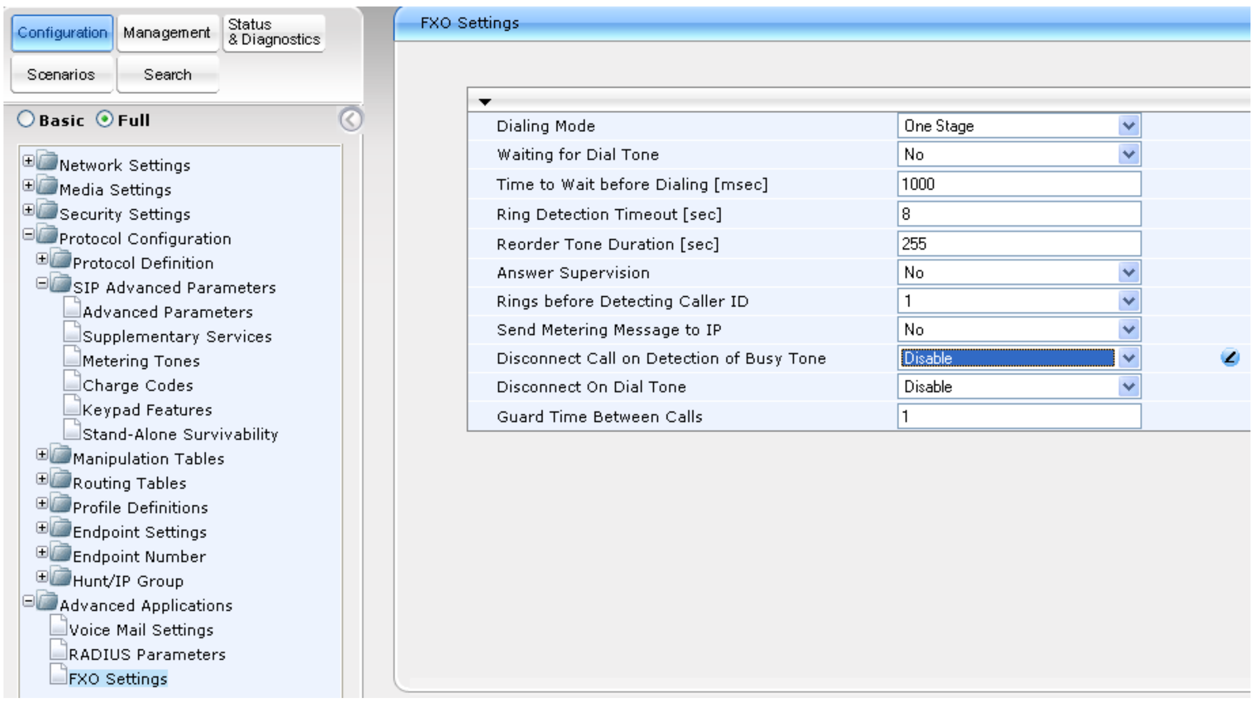
Navigate to **Protocol Configuration → SIP Advanced Parameters → Advanced Parameters**. Under the heading **Misc. Parameters**, change the **Default Release Cause** to the value **34**. With this change, the AudioCodes Gateway will return a SIP "503" response when all FXO trunks are busy. The SIP "503" response is among the list of SIP responses that can trigger Avaya Communication Manager LAR, if LAR is configured. Click **Submit** in the lower right corner of the screen (not shown in the abbreviated screen below).



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

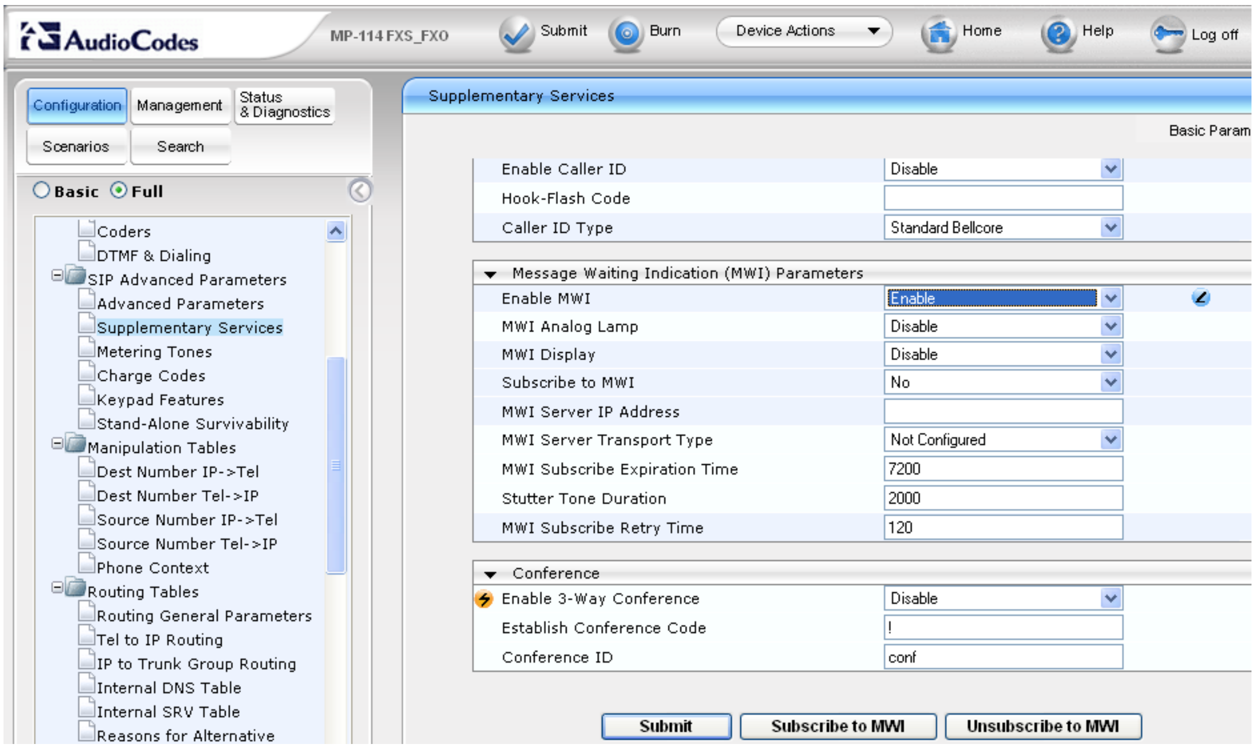
The AudioCodes Gateway can automatically detect when a call is connected to busy tone from the PSTN on an FXO line, and disconnect the call if desired. For the sample configuration, it is recommended that this feature be disabled. If the feature remains enabled, and an Avaya SIP Telephone in the branch makes a call to a local PSTN number 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**. Using the drop-down menu, change the **Disconnect Call on Detection of Busy Tone** parameter to the value **Disable**. Click **Submit** in the lower right corner of the screen (not shown in the abbreviated screen below).



6.20. 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**. Select "Enable" from the **Enable MWI** drop-down, as shown in the following screen. Press the **Submit** button. When a SIP user registers, or the message waiting status of a registered user changes, the Avaya SES 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 an Avaya Modular Messaging mailbox). It is not necessary that the AudioCodes Gateway subscribe to MWI, but this option is also available. Observe that the **Stutter Tone Duration** can also be configured.



6.21. .ini File

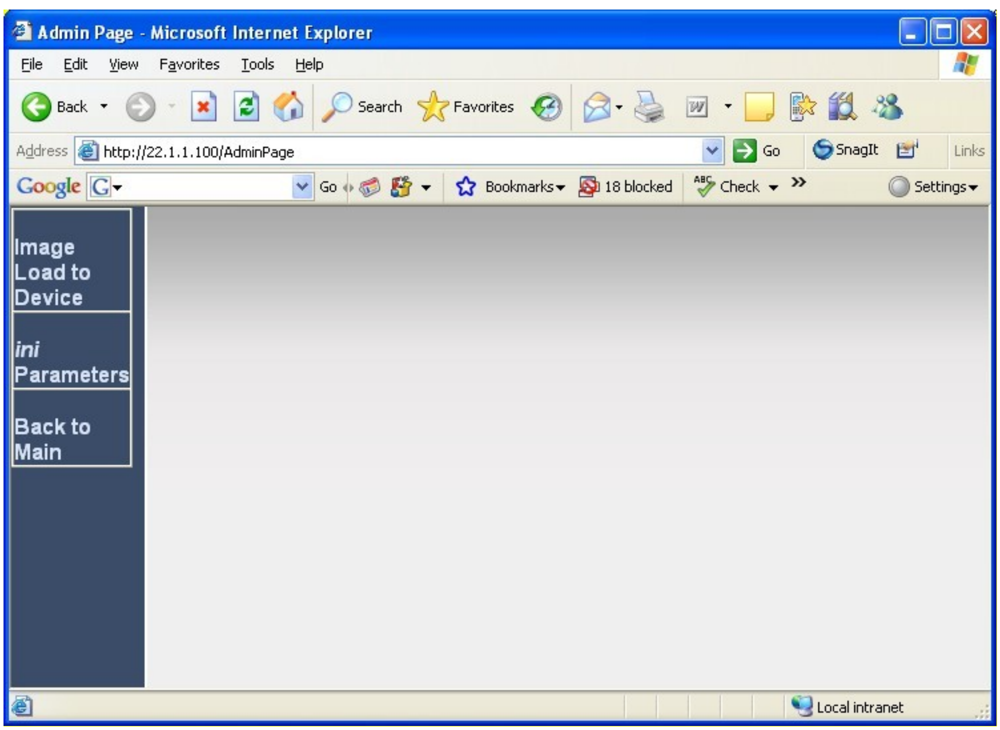
The AudioCodes MP-114 utilizes an initialization text file with a .ini extension. The .ini file contains MP-114 parameters that have been set by the WebUI, such as the parameters described in the previous sections. See [6] for additional information on the ini configuration file.

As of the AudioCodes MP-114 firmware version listed in **Table 2**, the following parameters are not configurable from the WebUI and must be modified directly in the .ini file.

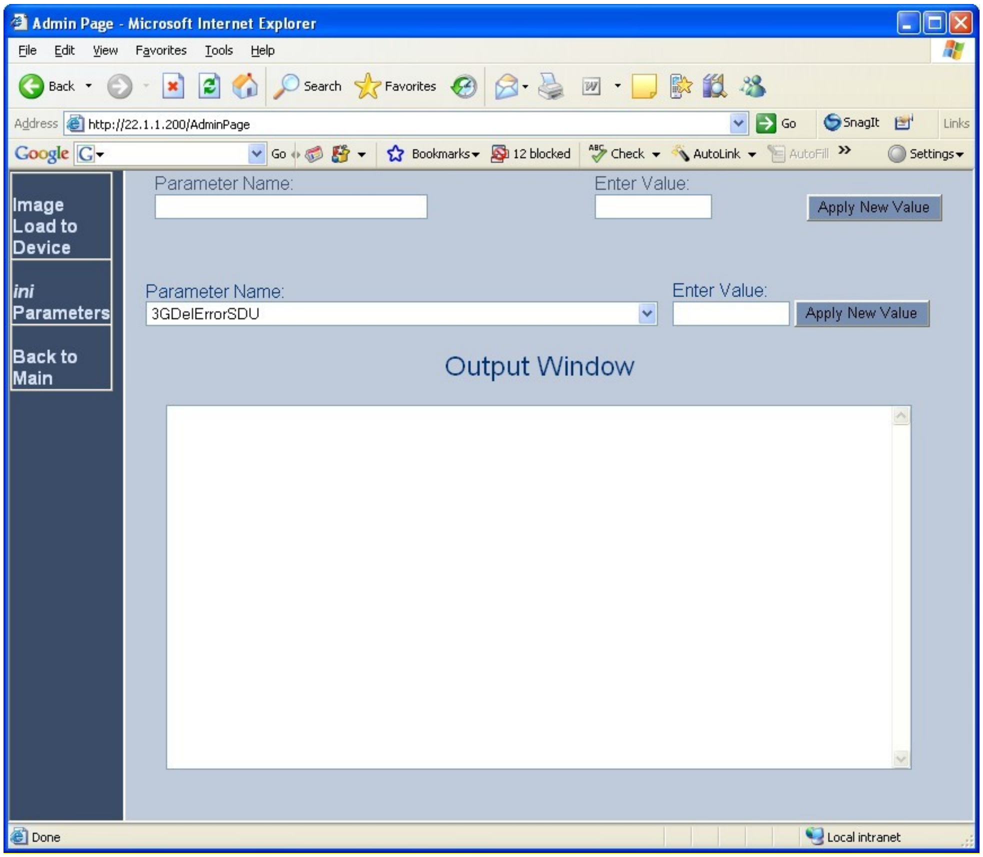
- SASSurvivabilityMode
- ReliableConnectionPersistentMode
- CurrentDisconnectDuration

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-114 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.

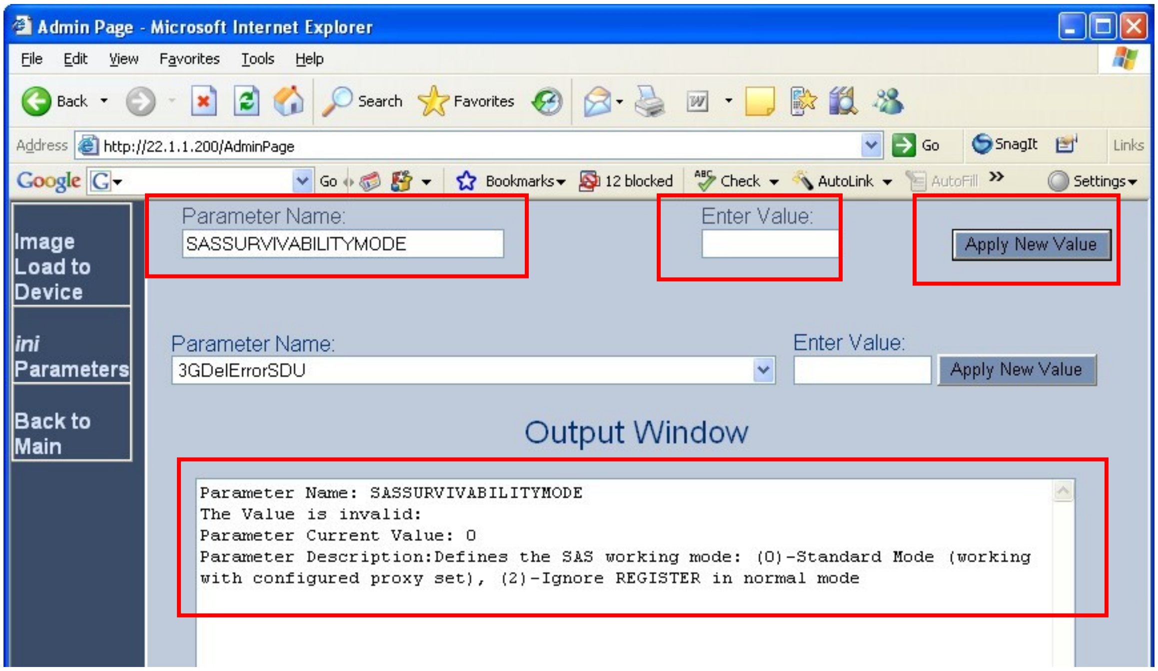


6.21.1. SASSurvivabilityMode

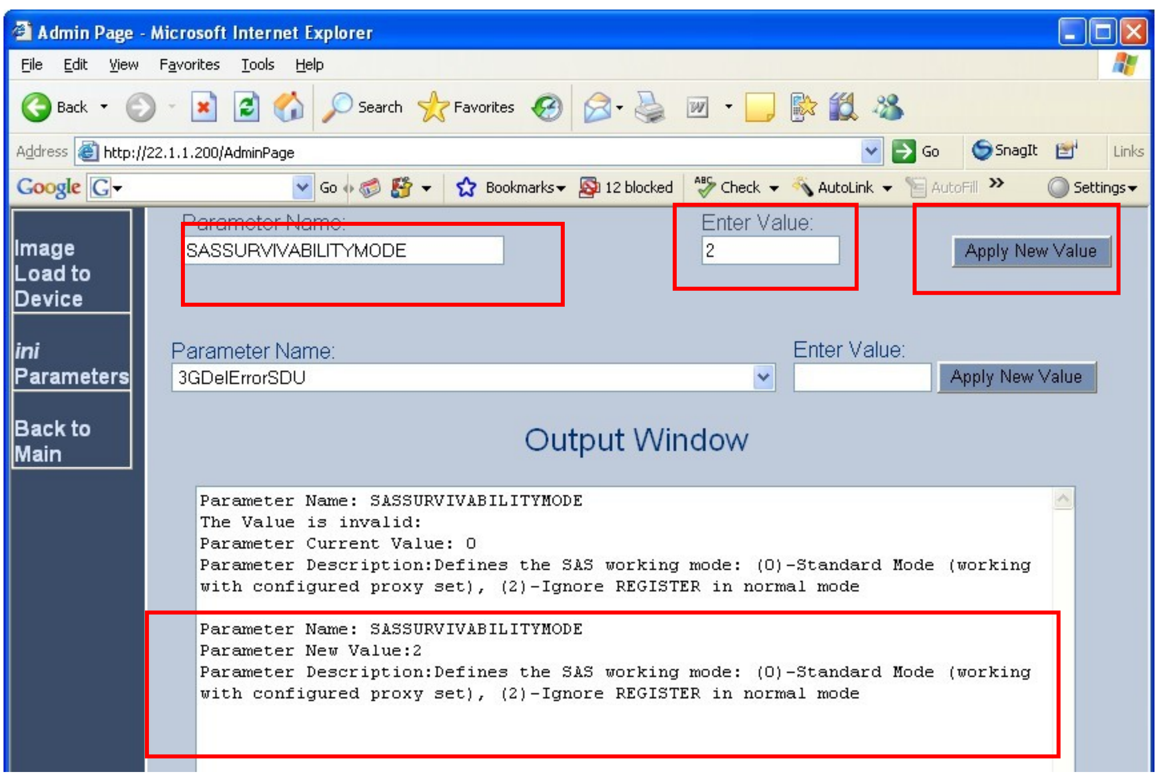
The **SASSurvivabilityMode** parameter determines how the SAS feature of the AudioCodes MP-114 will operate. By default, **SASSurvivabilityMode** is set to a value of 0 which enables SAS to be able to accept SIP Registrations while the AudioCodes MP-114 can simultaneously communicate with the Avaya SES.

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 Avaya SES.

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" of the following screen shows the **SASSurvivabilityMode** parameter is currently set to the default value of 0.



To change the value of a parameter, enter the new parameter value in the "Enter Value" field. The following screen shows the **SASSurvivabilityMode** parameter being set to 2. The text appended to the "Output Window" shows the **SASSurvivabilityMode** parameter was successfully set to a value of 2.

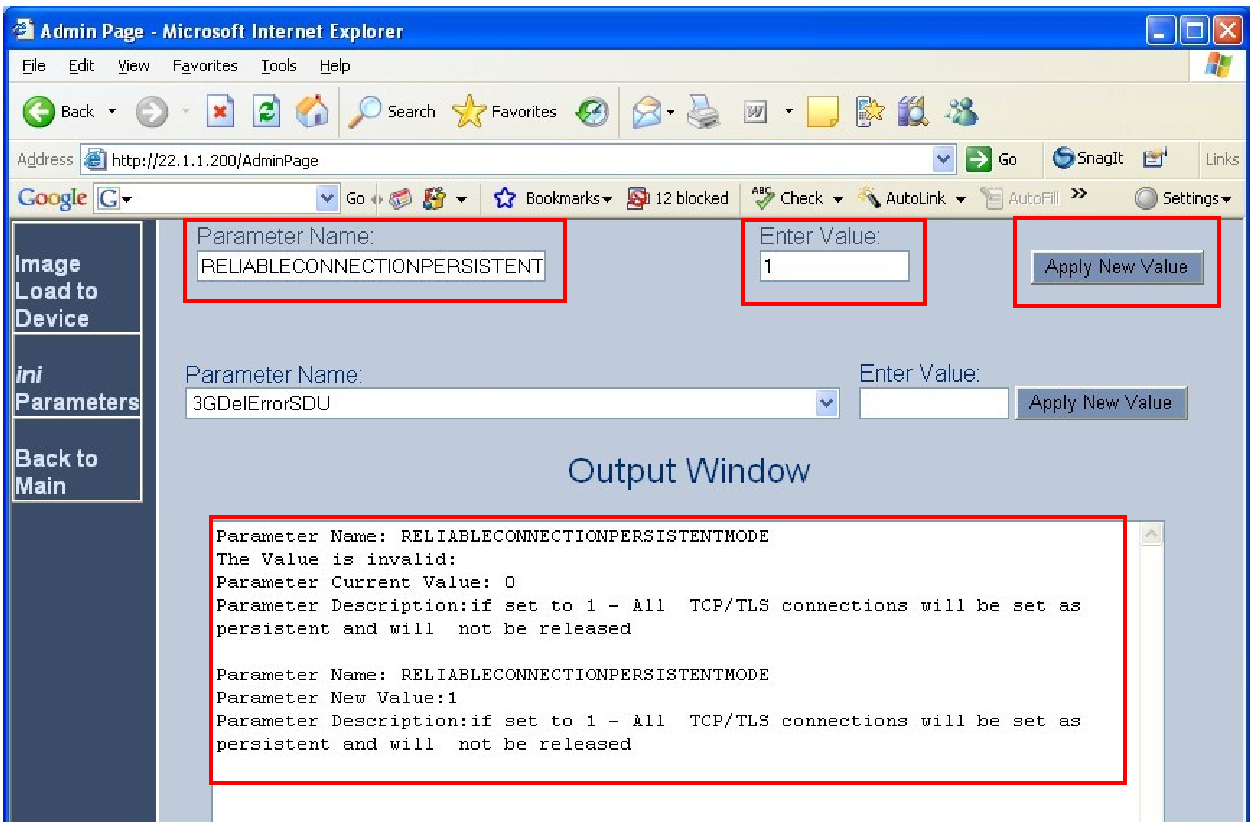


6.21.2. ReliableConnectionPersistentMode

The **ReliableConnectionPersistentMode** parameter determines how the AudioCodes MP-114 establishes TCP connections. When **ReliableConnectionPersistentMode** is set to the default value of 0, all TCP/TLS connections established by the AudioCodes MP-114 are non-persistent connections.

ReliableConnectionPersistentMode must be changed from the default value of 0 to a value of 1. This configures the AudioCodes MP-114 to establish all TCP connections as persistent connections ensuring that they will not be prematurely released.

The following screen shows the value of the **ReliableConnectionPersistentMode** parameter was successfully set to a value of 1.



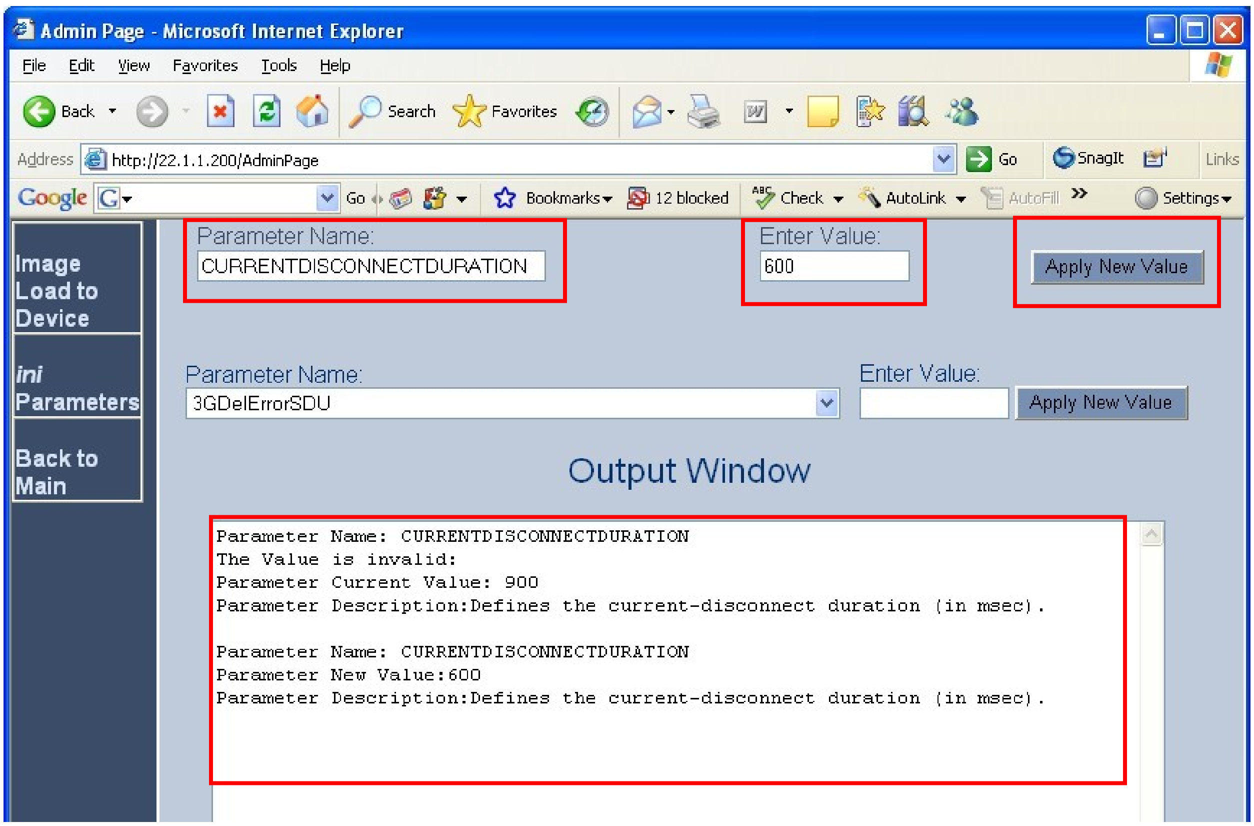
6.21.3. CurrentDisconnectDuration

The **CurrentDisconnectDuration** parameter determines the duration of time in milliseconds the analog line current is dropped indicating a disconnect pulse to the AudioCodes MP-114 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-114 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 6.7**.

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



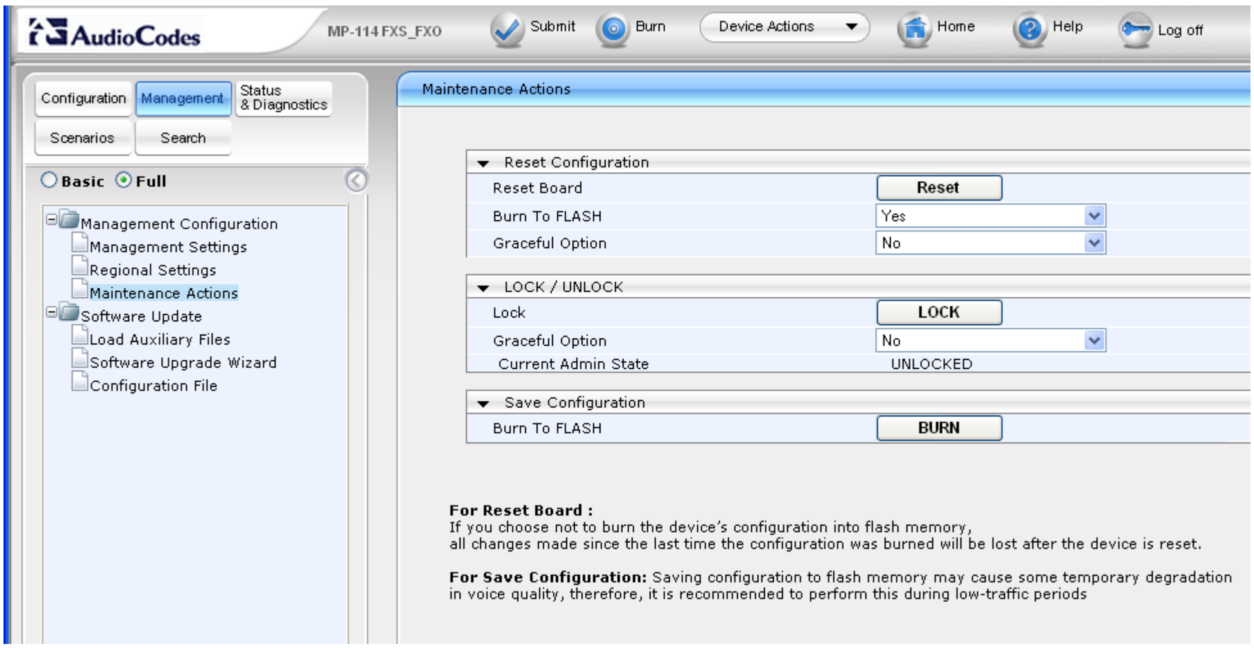
6.22. Saving Changes to the AudioCodes Gateway

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.



7. Avaya 9600 SIP Phone

The configuration parameters of the Avaya 9600 SIP Phone specific to SIP Survivability and the sample configuration are described in this section. See reference [1] before setting or changing any of the parameters.

46xxsettings.txt Parameter Name	Value Used in Sample Configuration	Description
SIP_CONTROLLER_LIST	30.1.1.35	A priority list of SIP Servers for the phone to use for SIP services. The sample configuration uses the Avaya SES Survivable Call Processor feature to specify the details of the survivable server. As a result, only the value of the Avaya SES IP Address is specified. The port and transport use the default values of 5061 and TLS when not specified. This parameter is provided as an alternative method to the Avaya SES Survivable Call Processor for setting each phones SIP Server list. The example below shows the values used for this parameter for a phone in Branch 2 if the Avaya SES Survivable Call Processor method were not used. The Avaya SES is the first priority SIP Server listed using the default port and transport of 5061 and TLS. Separated by a comma, the Branch 2 AudioCodes MP-114 is the next priority SIP Server using port 5070 and TCP transport. 30.1.1.35,22.1.1.100:5070;transport=tcp

		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 Avaya SES and the Branch 1 AudioCodes MP-114 while the list for Branch 2 phones will include the Avaya SES and the Branch 2 AudioCodes MP-114. To accomplish this, the GROUP system value mechanism can be implemented as described in [1].
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	1010	The number dialed when the Message button is pressed and the phone is in Normal Mode.
PSTN_VM_NUM	917325551010	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	1xxx|11xxxxx|22xxxxx|33xxx|911|9911|91xxxxxxxxx|9011x+|732xxxxxxx	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 learns the dial plan from SES and does not require these settings to expedite dialing. The dialplan values used in the phone match the values used by the AudioCodes MP-114 in **Section 6.6**. See [1] 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		The enterprise SIP domain. Must be the same for all SIP controllers in the configuration. SIPDOMAIN is set to "retail.com" in the sample configuration.

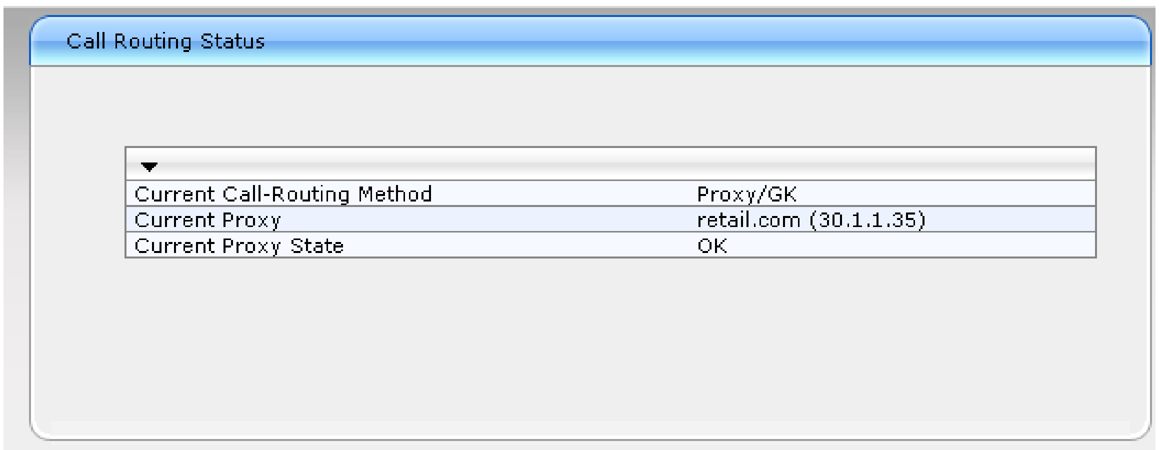
8. Verification and Troubleshooting

8.1. AudioCodes MP-114 Call Routing Status

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

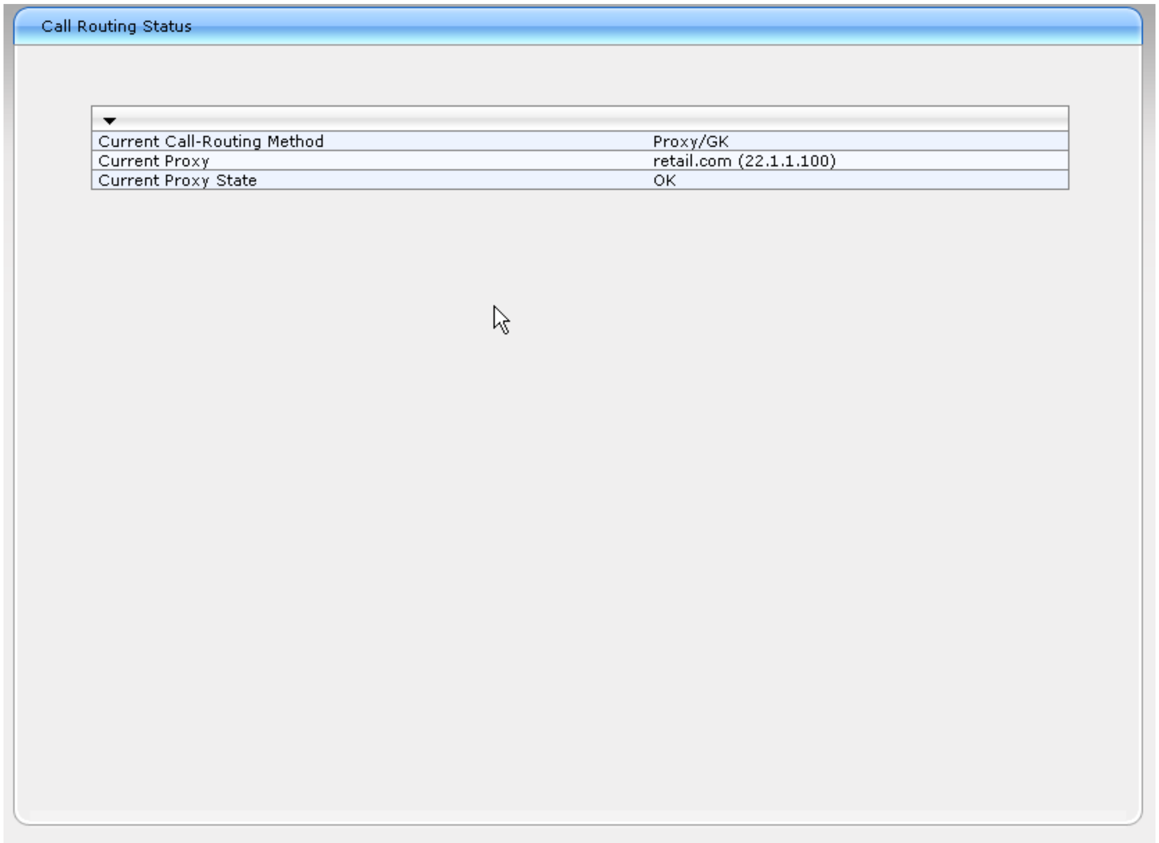
The Call Routing Status screens from the Branch 2 AudioCodes MP-114 while in Normal Mode and Survivable Mode are shown below.

Normal Mode:
The status shows all call routing is using the centralized Avaya SES IP address named retail.com which is in a OK state.



Survivable Mode:

The status shows all call routing is using the internal AudioCodes SAS Proxy named retail.com and the Current Proxy State is in the "OK" state as presented by AudioCodes Software Version 5.6.

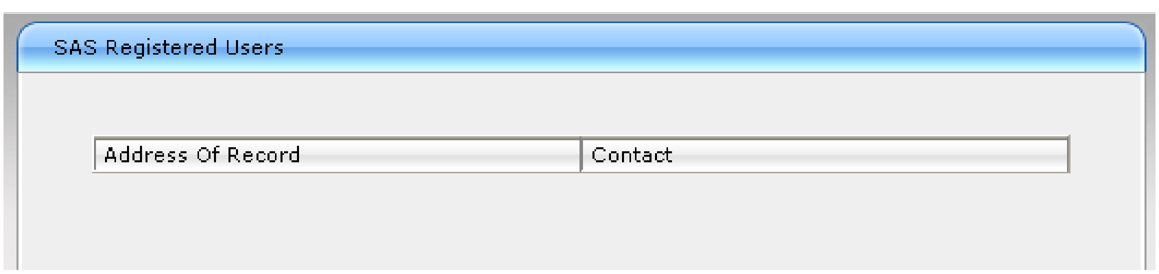


8.2. SAS/SBC Registered Users

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-114 while in Normal Mode and Survivable Mode are shown below.

Normal Mode:
The screen shows no active SAS users.



Survivable Mode:
The screen shows three Branch 2 Avaya 9600 SIP Phones actively registered to the AudioCodes MP-114 SAS.

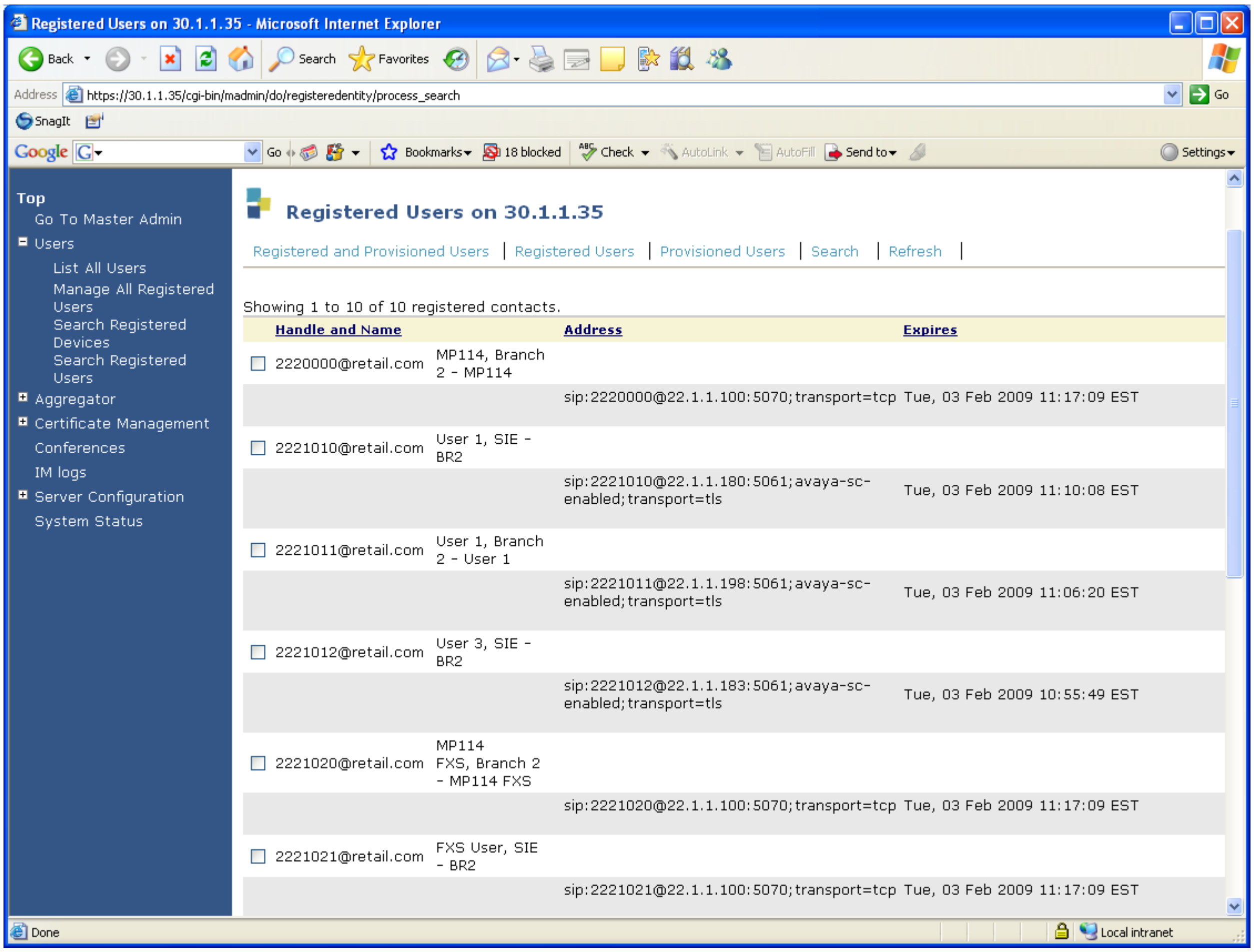
SAS Registered Users

Address Of Record	Contact
2221010@RETAIL.COM	<sip:2221010@22.1.1.180;avaya-sc-enabled;transport=tcp>;q=1;expires=7200;reg-id=2;+sip.instance="<urn:uuid:00000000-0000-1000-8000-00073bd274d0>"
2221011@RETAIL.COM	<sip:2221011@22.1.1.198;avaya-sc-enabled;transport=tcp>;q=1;expires=7200;reg-id=2;+sip.instance="<urn:uuid:00000000-0000-1000-8000-00073bb5aad2>"
2221012@RETAIL.COM	<sip:2221012@22.1.1.183;avaya-sc-enabled;transport=tcp>;q=1;expires=7200;reg-id=2;+sip.instance="<urn:uuid:00000000-0000-1000-8000-00040dec5487>"

8.3. SES Registered Users

The following screen shows Avaya SES registered users from Branch 2 in normal mode, when the WAN is up. This screen can be accessed from the Avaya SES Home server by clicking on **Users → Search Registered Users**.

Note the user registration for the "gateway user" (first record), the three Avaya SIP phones (second, third, and fourth records), and the two FXS stations connected to the AudioCodes gateway at the branch (last two records).



8.4. 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 Avaya SES. 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.

8.5. 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 Avaya SES for normal service, once the branch communications with the central Avaya SES 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 Avaya SES, idle Avaya SIP Telephones in the branch will typically be registered with the Avaya SES 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 SES before others. For example, some may register within 30 seconds, others within 45 seconds, with others registering in approximately one minute.

9. 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 Avaya Communication Manager Survivable SIP Gateway Solution to minimize service disruptions to these remote branch SIP endpoints.

10. References

Avaya Application Notes and additional resources can be found at the following web address http://www.avaya.com/gcm/master-usa/en-us/resource/. Avaya Product Support web site can be found at the following web address http://support.avaya.com/.

The following Avaya references are relevant to these Application Notes:

[1] ***Avaya one-X Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide***
Doc ID: 16-601944, Issue 4, December 2008

[2] ***Administering SIP Enablement Services on the Avaya S8300 Server,***
Doc ID: 03-602508, Issue 1, January 2008

[3] ***Administrator Guide for Avaya Communication Manager,***
Doc ID: 03-300509, Issue 4, January 2008

[4] ***Avaya Communication Manager Survivable SIP Gateway Solution using the AudioCodes MP-114 in a Centralized Trunking Configuration – Issue 1.1***

[5] ***Sample Configuration for SIP Private Networking and SIP Look-Ahead Routing using Avaya Communication Manager, Issue 1.0***

http://www.avaya.com/master-usa/en-us/resource/assets/applicationnotes/sip-pvt-lar.pdf

The following AudioCodes references are relevant to these Application Notes:

[6] ***AudioCodes SIP MP-124 & MP-11x Release Notes Version 5.6,***
Version 5.6

http://www.audiocodes.com/filehandler.ashx?fileid=42853

[7] ***AudioCodes SIP MP-124 & MP-11x Users Manual Version 5.6***
http://www.audiocodes.com/filehandler.ashx?fileid=36362

11. Change History

Issue	Date	Reason
1.1	04/27/2009	Revised version with additional configuration changes on AudioCodes MP-114 which has been upgraded to software version 5.6. Without these changes, the FXS stations of MP-114 fail to make outgoing calls when the MP-114 is in Survivability Mode. Support of T.38 Fax Relay mode at AudioCodes MP-114.
1.0	03/14/2009	First Published version. The AudioCodes MP-114 runs software version 5.4 in the sample configuration.

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, Avaya Communication Manager uses ARS location-based routing to route the call back to the Avaya SES. The Avaya SES is configured with maps that match on the leading digits of the PSTN number (e.g.., an area code), and direct the call to the proper AudioCodes Gateway. The 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, Avaya Communication Manager can insert a branch prefix code so that the maps configured on the Avaya SES can distinguish the proper AudioCodes Gateway based on the branch prefix. This approach uses the Avaya 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 the Avaya SES. This Appendix shows the additions to the configuration to enable this approach to 911.

In Avaya 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 12.

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

          Dialed           Total      Route     Call   Node   ANI
          String          Min  Max   Pattern    Type   Num    Reqd
    1732                  11   11     8         natl          n
    911                    3    3     12        emer          n
```

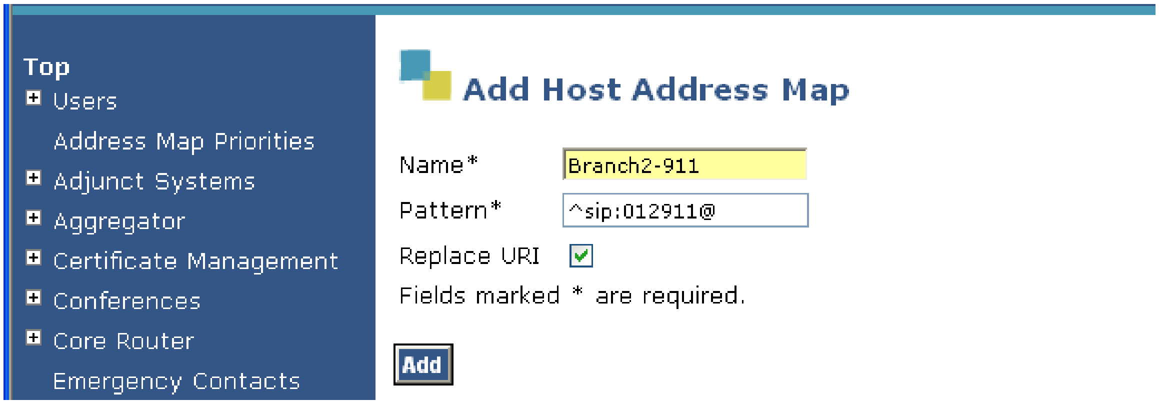
In route pattern 12, insert a prefix to uniquely identify the branch. In the sample below, the number "012" is chosen to match the location number used for ARS location-based routing. It is not necessary to match the location number. Trunk group 8 is a SIP trunk to the Avaya SES.

```
change route-pattern 12                                         Page   1 of   3

                    Pattern Number: 12   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: 8    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
```

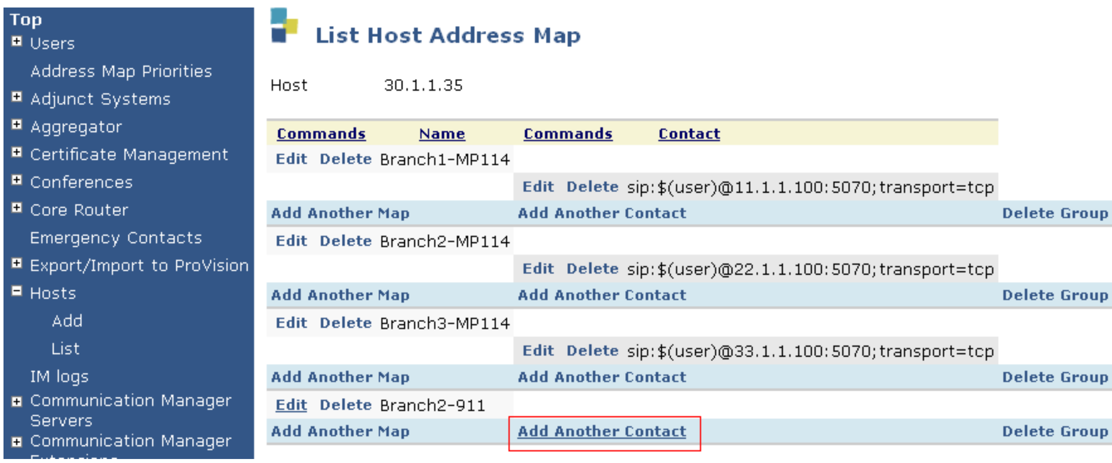
From the Avaya SES, configure a host map matching the number "012911". The contact for the map will be the AudioCodes Gateway for that location.

As in **Section 4.3,** from the Avaya SES, select **Hosts → List → Map**. Then select "Add Map in New Group" as shown below.

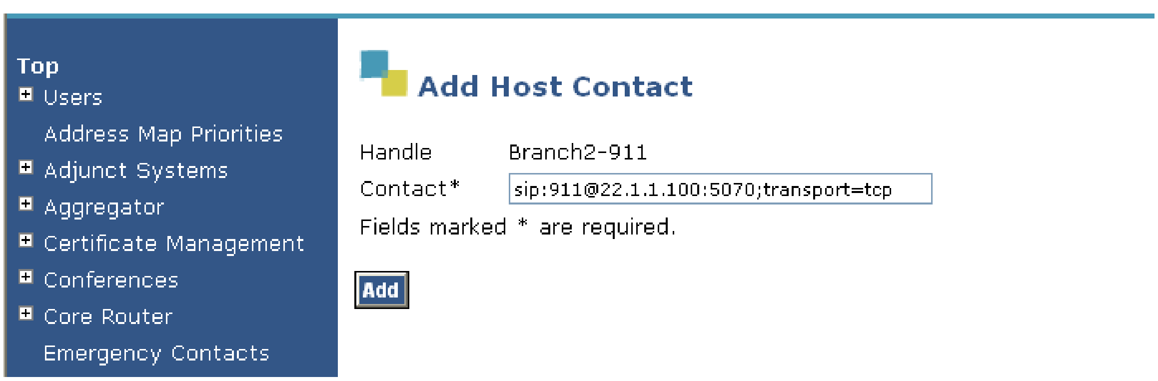
In the **Pattern** field, enter a pattern that will match the branch prefix inserted on the route pattern, followed by 911. In this case, "^sip:012911@" is entered, as shown below. Enter a descriptive **Name**. Click **Add**.



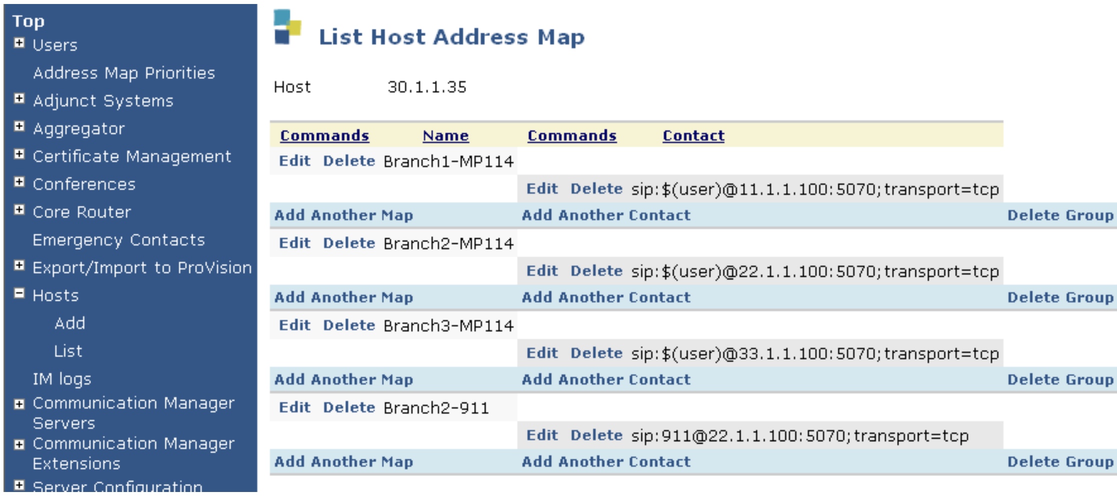
Click **Continue** at the confirmation screen (not shown). For the Map just created, click **Add Another Contact** as shown below.



In the **Add Host Contact** screen, enter the **Contact** such that the call will route directly to 911 at the proper AudioCodes Gateway IP Address, port, and transport. The TCP protocol and port number match the configuration, similar to the map with name “Branch2-MP114” used for calls to area code 732.. Click **Add**. Then click **Continue** at the confirmation screen (not shown).



The following screen shows the list of host address maps, after the new “Branch2-911” map has been added.



The sample configuration of the AudioCodes Gateway in these Application Notes does not require any changes to allow the AudioCodes Gateway to route the 911 call out an FXO port. The 911 call will be directed to hunt group 2, and FXO port 3.

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 3 of the branch 2 MP-114 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. Avaya SES maps 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.