Avaya Solution & Interoperability Test Lab

Avaya Communication Manager Survivable SIP Gateway Solution using the Cisco 2821 Integrated Services Router with Survivable Remote Site Telephony enabled, in a Distributed Trunking Configuration – Issue 1.0

Abstract

These Application Notes describe the configuration of the Avaya Communication Manager Survivable SIP Gateway Solution using the Cisco 2821 Integrated Services Router (SRST enabled) 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 centralised 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 central 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 centralised Avaya SIP call control platform. When connectivity loss is detected, Avaya one-X Deskphone™ SIP 9600 Series IP Telephones along with the Cisco Integrated Service Router dynamically switch to surivability mode, restoring telephony services at the branch for intra-branch and Public Switched Telephone Network calling.

Testing was conducted via the Interoperability Program at the Avaya Solution and Interoperability Test Lab at the request of the Solutions and Marketing Team.

1. Introduction

These Application Notes describe the configuration of the Avaya Communication Manager Survivable SIP Gateway Solution using the Cisco Integrated Service Router (ISR) in a distributed trunking scenario.

SIP endpoints deployed at remote branch locations risk a loss of service if a break in connectivity to the centralised SIP call control platform occurs. Connectivity loss can be caused by WAN access problems being experienced at the branch or network problems at the Headquarters (HQ) site blocking access to the Avaya SIP call control platform. The branch Avaya one-X Deskphone™ SIP 9600 Series IP Telephones (2.4 firmware release), monitor connectivity to HQ. 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 remote branch to HQ Avaya SIP Enablement Services is restored, SIP components can dynamically switch back to normal operation.

2. Overview

This section describes the major components, test environment and test method covered in these Application Notes.

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

The Avaya one-X Deskphone SIP 9600 Series IP Telephone Release 2.4, 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 firmware release 2.4 of the Avaya 9600 SIP Phone includes new feature capabilities specific to SIP survivability enabling the phone to monitor connectivity to the Avaya SIP Enablement Services and dynamically failover to the local Cisco Integrated Services Router (ISR) as an alternate or survivable SIP server. See reference [1] for additional information on the Avaya 9600 SIP Phone.

2.2. Public Switched Telephone Network Trunking Configurations

The Avaya Communication Manager Survivable SIP Gateway Solution can interface with the Public Switched Telephone Network (PSTN) in either a Centralised 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 Centralised Trunking and Distributed Trunking as related to the Avaya Communication Manager Survivable SIP Gateway Solution:

- **Centralised 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 E1 interface (in this sample configuration) of the local branch Cisco ISR. This has the

potential benefits of saving bandwidth on the branch access network, off loading the WAN and centralised media gateway resources, avoiding Toll Charges, and reducing latency.

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 Centralised Trunking configuration, see the Application Notes titled “Avaya Communication Manager Survivable SIP Gateway Solution using the Cisco ISR in a Centralised Trunking Configuration”.

2.3. Cisco 2821 Integrated Service Router

The Cisco 2821 Integrated Services Router, referred to as Cisco ISR throughout the remainder of this document, takes on various roles based on call flows and network conditions. The Cisco ISR includes the Survivable Remote Site Telephony or SRST feature enabled. The following roles are supported by the ISR:

- SIP PSTN Media Gateway
  (NM-DH with VWIC-2MFT-E1-DI interfaces to PSTN)
- SIP Analog Terminal Adapter
  (Vic2-4FXS interfaces to analog endpoints)
- SIP Registrar and Proxy
  (Configured as service applications, used during loss of connectivity between Branch1 and HQ Avaya SIP Enablement Services)

2.4. Network Modes

PSTN call routing is further determined within each of the trunking configurations based on the network status of each branch. These are as follows:

- **Normal Mode:** Branch has WAN connectivity to the main Headquarters/Datacenter location and the HQ Avaya SIP call control platform is being used for all branch calls.
- **Survivable Mode:** Branch has lost WAN connectivity to the Headquarters/Datacenter location and the local branch Cisco ISR assumes SIP call control for all calls at that branch.
  Note: If the Avaya SIP Enablement Services loses connectivity to the WAN, all branches will go into survivable mode simultaneously.

2.5. Call Flows

This 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. Call Control Description for Distributed Trunking in Normal Mode

The following list details the types of call control existing in the sample configuration for normal mode operation.

- **SIP Call Control**: All SIP call control and call routing is provided by the central Avaya SIP Enablement Services 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 Cisco ISR E1 interface.
- **Branch PSTN Outbound non-Local**: All PSTN outbound calls from the branch are routed to a centralised Avaya G650 Media Gateway.
- **Branch PSTN Inbound**: Calls from the PSTN to a branch enter the network at the local branch Cisco ISR E1 interface then route to the Avaya SIP Enablement Services /Avaya Communication Manager for call treatment.
- **Headquarters PSTN Inbound**: Calls to Headquarters endpoints enter the network at the Headquarters Avaya G650 Media Gateway.

2.5.2. Call Flow Examples for Distributed Trunking in Normal Mode

The following are examples of call flow scenarios for distributed trunking in normal mode setup.

- **Avaya 9600 SIP Phone at branch to H.323 IP phone at Headquarters.**
  Avaya 9600 SIP → SIP Enablement Services → Avaya Communication Manager → H.323 IP phone
- **Avaya 9600 SIP Phone at branch to Digital/Analog phone at Headquarters.**
  Avaya 9600 SIP → SIP Enablement Services → Avaya Communication Manager → Avaya Media Gateway → Digital/Analog phone
- **Avaya 9600 SIP Phone at branch to PSTN endpoint – Local Number**
  Avaya 9600 SIP→ SIP Enablement Services → Avaya Communication Manager → SIP Enablement Services → Branch Cisco ISR → PSTN phone
- **Avaya 9600 SIP Phone at branch to PSTN endpoint – non-Local Number**
  Avaya 9600 SIP→ SIP Enablement Services → Avaya Communication Manager→ PSTN phone
- **Avaya 9600 SIP Phone at branch to Avaya 9600 SIP phone at same branch.**
  Avaya 9600 SIP → SIP Enablement Services → Avaya Communication Manager → SIP Enablement Services → Avaya 9600 SIP
- **PSTN phone to Branch Avaya 9600 SIP phone.**
  PSTN phone → Cisco ISR E1 → SIP Enablement Services→ Avaya Communication Manager → SIP Enablement Services → Branch Avaya 9600 SIP

Figure 1 presents a high level network view of the test sample in a Distributed Trunking Normal Mode scenario.

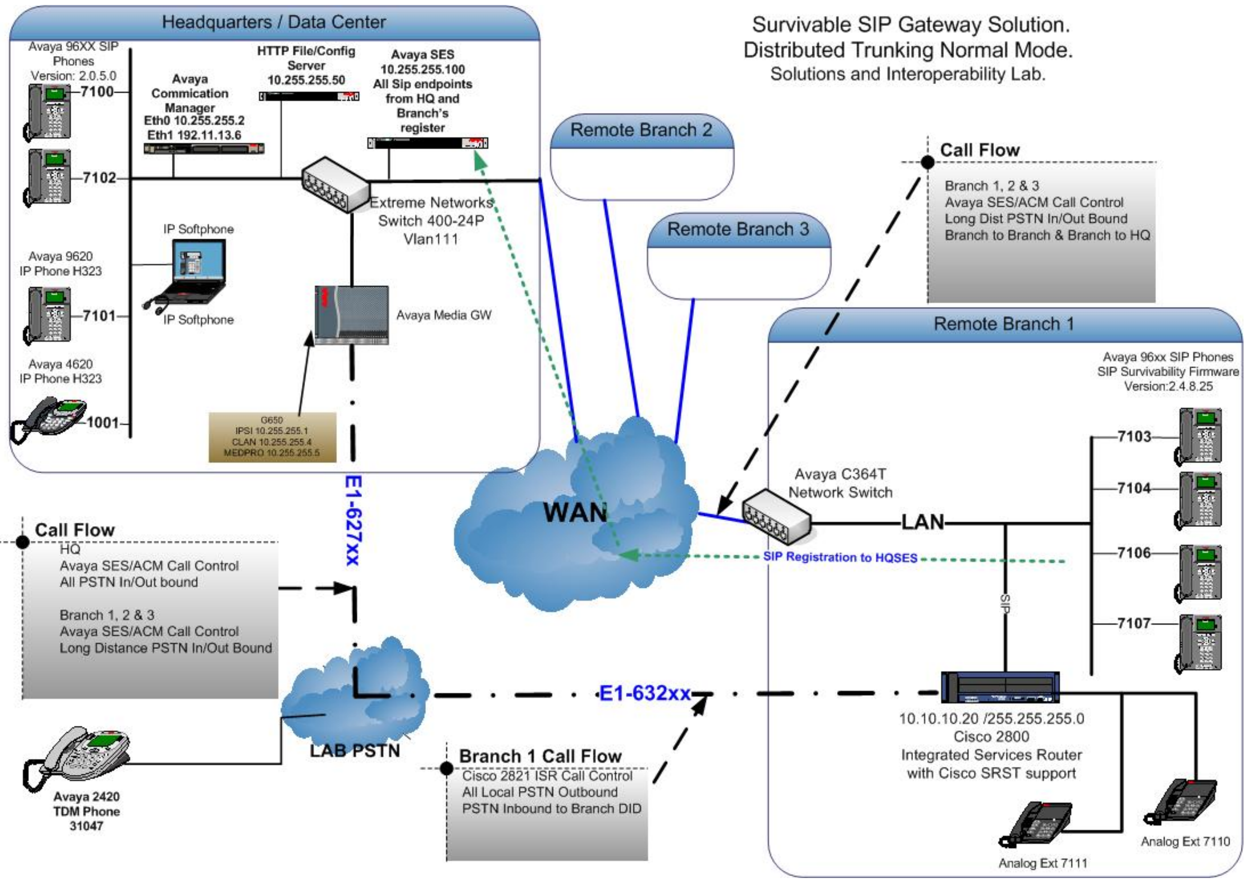


Figure 1: Network Diagram Distributed Trunking Normal Mode

2.5.3. Call Control Description for Distributed Trunking in Survivability Mode

The following list details the types of call control existing in the sample configuration for survivable mode operation.

- **SIP Call Control**: All SIP call control and call routing is provided by the local branch Cisco ISR.
- **SIP Registration**: All branch Avaya 9600 SIP Phones are transitioned and registered to the branch Cisco ISR.
- **Branch PSTN Outbound**: All PSTN Outbound calls are routed to the Branch Cisco ISR E1 interface.
- **Branch PSTN Inbound**: Calls from the PSTN to the branch enter the network at the local branch Cisco ISR E1 interface. The Cisco ISR routes the calls accordingly.

2.5.4. Call Flows for Distributed Trunking in Survivable Mode

The following are examples of call flow scenarios for distributed trunking in survivable mode setup.

- **Avaya 9600 SIP Phone at branch to PSTN endpoint**
  Avaya 9600 SIP → Branch Cisco ISR E1 → PSTN phone
- **PSTN phone to Branch Avaya 9600 SIP phone.**
  PSTN phone → Branch Cisco ISR E1 → Avaya 9600 SIP
- **Avaya 9600 SIP Phone at branch to Avaya 9600 SIP phone at same branch.**
  Avaya 9600 SIP → Branch Cisco ISR E1 → Avaya 9600 SIP

Figure 2 presents a high level view of the test sample in a Distributed Trunking Survivable Mode scenario.

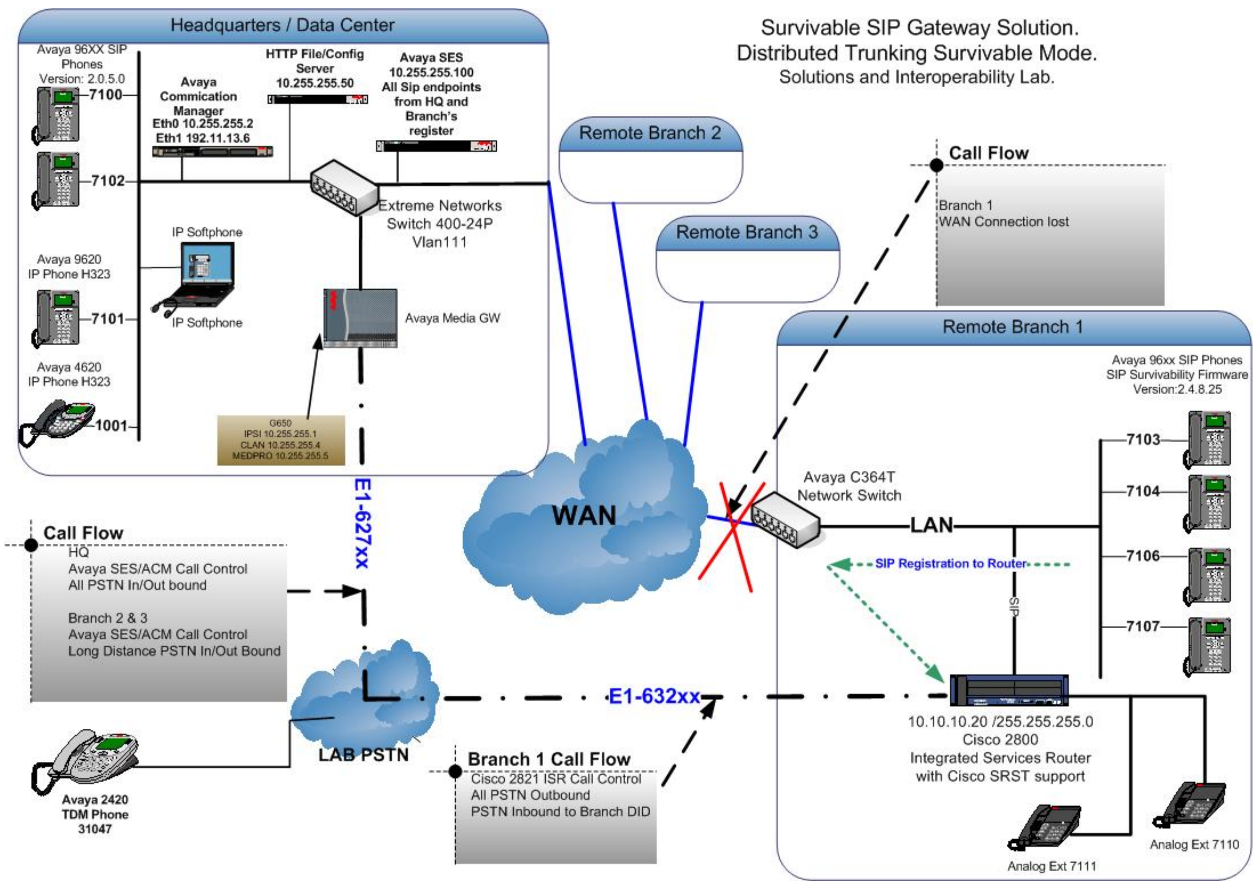


Figure 2: Network Diagram Distributed Trunking Survival Mode

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 for Branch1, Branch2 and Branch3 are set-up in the same way as is instructed for Branch1

IP Network	IP Network Region	Branch Name	Location	Area Code	Branch Cisco ISR IP Address
10.255.255.0/24	1		1	201	NA
10.10.10..0/24	2	Branch1	2	609	10.10.10.20
11.1.1.0/24	3	Branch2	3	709	11.1.1.20
12.1.1.0/24	4	Branch3	4	809	12.1.1.20

Table 1: Details of IP Network Regions

2.5.5. Call Flow Example for Location Based Call Routing in Normal Mode

Many of the Avaya SES and Avaya Communication Manager configuration steps presented in **Section 5** and **Section 6** 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

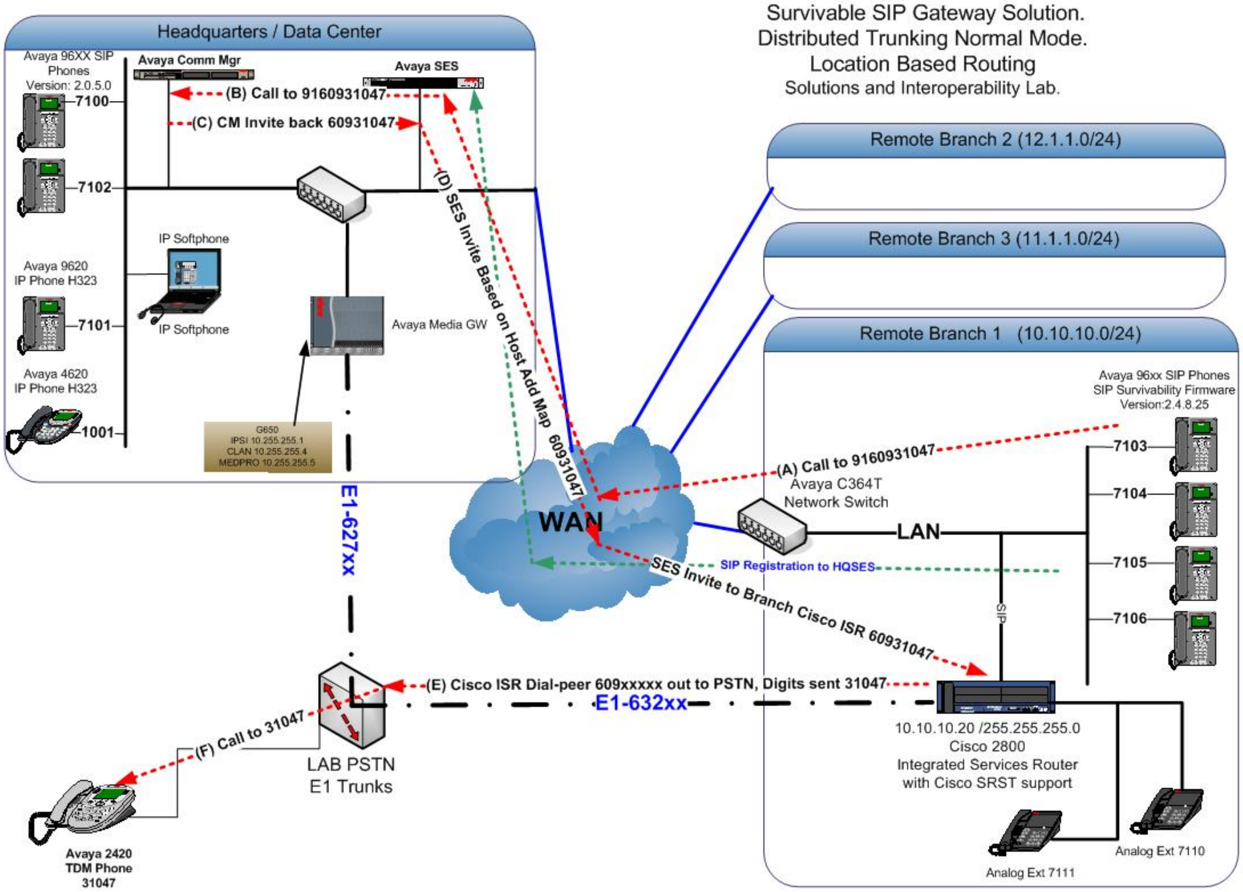


Figure 3: Network Diagram Illustration of Location Based Call Routing

An example of the Location Based Call Routing is illustrated in **Figure 3** above. The progress of this call example is as follows:

- **(A)** Call from Branch1 registered endpoint e.g. **7103** to local **PSTN 9160931047**
- **(B)** The call is routed to the HQ Avaya Communications Manager from HQ Avaya SES. The Invite origin IP address is detected and is matched to a Network region and Location, in this case **Location 2**. The leading digit **9** is identified through the Feature Access Code (FAC) as a call for Auto Route Selection (ARS).
- **(C)** Based on the invites origin IP Address, the ARS Digit Analysis Table **Location 2** is queried with the remaining leading digits **1609**, a match is made to **Route Pattern 11**. An Invite is then sent back to HQ Avaya SES, with the digits **60931047**.
- **(D)** The Host Address Mapping on the HQSES matches the digits and an Invite is sent to Branch1 Cisco ISR
- **(E)** The SIP invite is matched to a dial-peer on the Branch1 Cisco ISR, routing the call to the PSTN interface, forwarding the digits **31047**.
- **(F)** Finally the Call is established to extension **31047**.

3. Network Topology

The network implemented for the sample configuration shown in **Figure 1** & **Figure 2** are 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 diagrams, the Branch 1 configuration is expanded.

The Headquarters location hosts an Avaya SIP Enablement Services and Avaya Communication Manager providing enterprise-wide SIP call control and advanced feature capabilities. Avaya Communication Manager is running on an Avaya S8500 server. A flat network of 10.255.255.0/24 is implemented at Headquarters. The Headquarters location also hosts the following components: an Avaya G650 Media Gateway with PSTN trunks and 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.

4. Equipment and Software Versions

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

Equipment	Software
Avaya S8500 Server	Avaya Communication Manager 5.1 (R015x.01.2.416.4)
Avaya G650 Media Gateway IPSI TN2312BP CLAN TN799DP MEDPRO TN2302AP DS1 TN2464CP	 HW28 FW044 HW16 FW031 HW32 FW118 HW13 FW022
Avaya SIP Enabled Services (SES) Server	Release 5.1.1 (5.1.1.415.1)
Cisco ISR 2821 Dual 10/100/1000 Fast Ethernet Ports NM-DH with VWIC-2MFT-E1-DI Vic2-4FXS PVDM2-16	IOS 12.4(20)T C2800NM-IPVOICEK9-MSRST Enabled
Avaya one-X Deskphone	SIP G.A R2.4.1
Avaya one-X Deskphone	H323 2.9
Avaya 4600 Series IP Telephones	H.323 2.9
Avaya 2420-Digital Handset	NA

Table 2: Hardware and Software Version Information

5. Avaya SIP Enablement Services

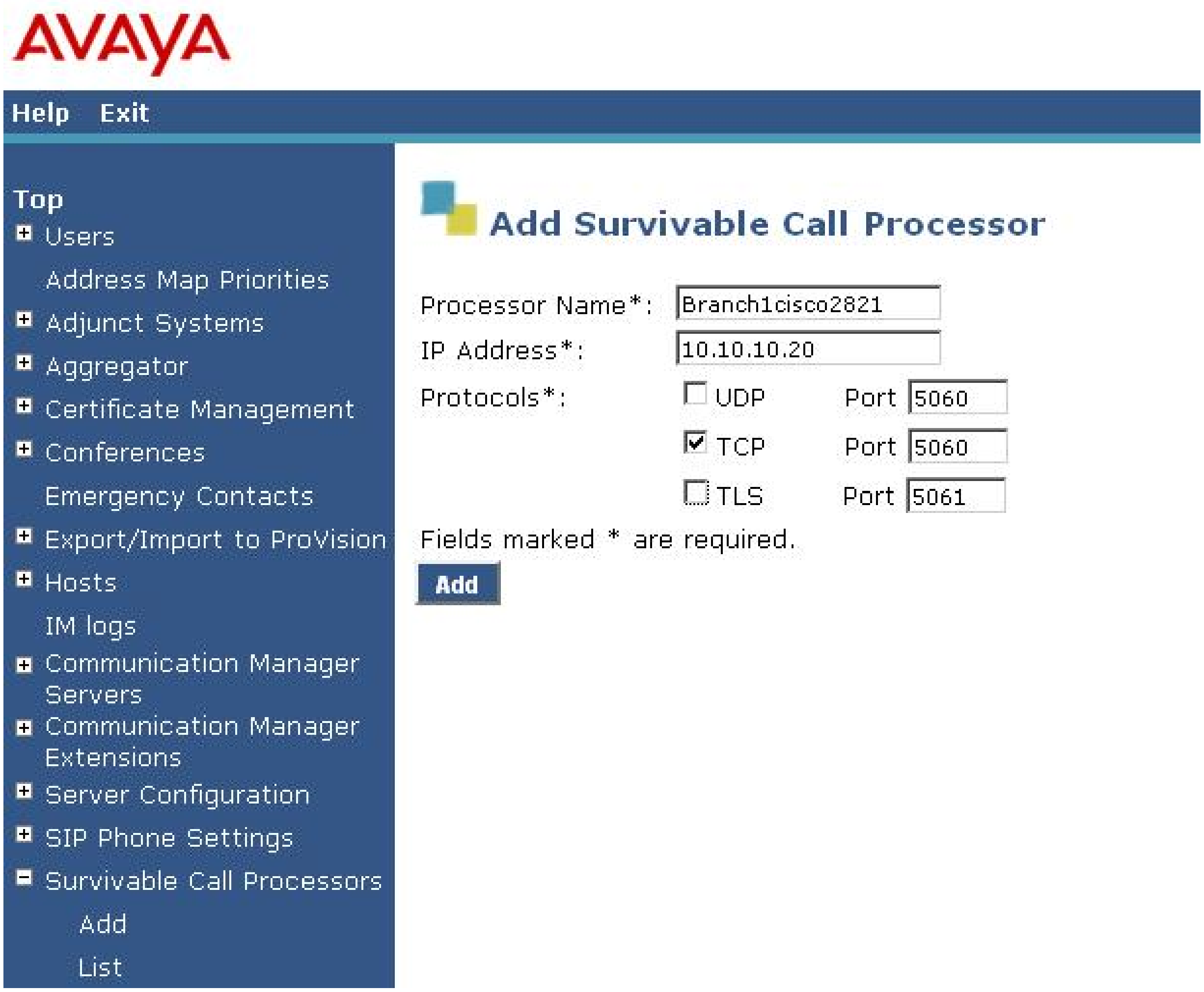
This section describes the configuration steps for the Avaya SIP Enablement Services.

5.1. Survivable Call Processors

The Survivable Call Processors feature of the Avaya SIP Enablement Services allows survivable SIP servers to be defined and then assigned to individual SIP Enablement Services user accounts. The Cisco ISR at each branch is configured as a Survivable Call Processor within Avaya SIP Enablement Services. The following screen illustrates the Cisco ISR for Branch 1 being added as a Survivable Call Processor.

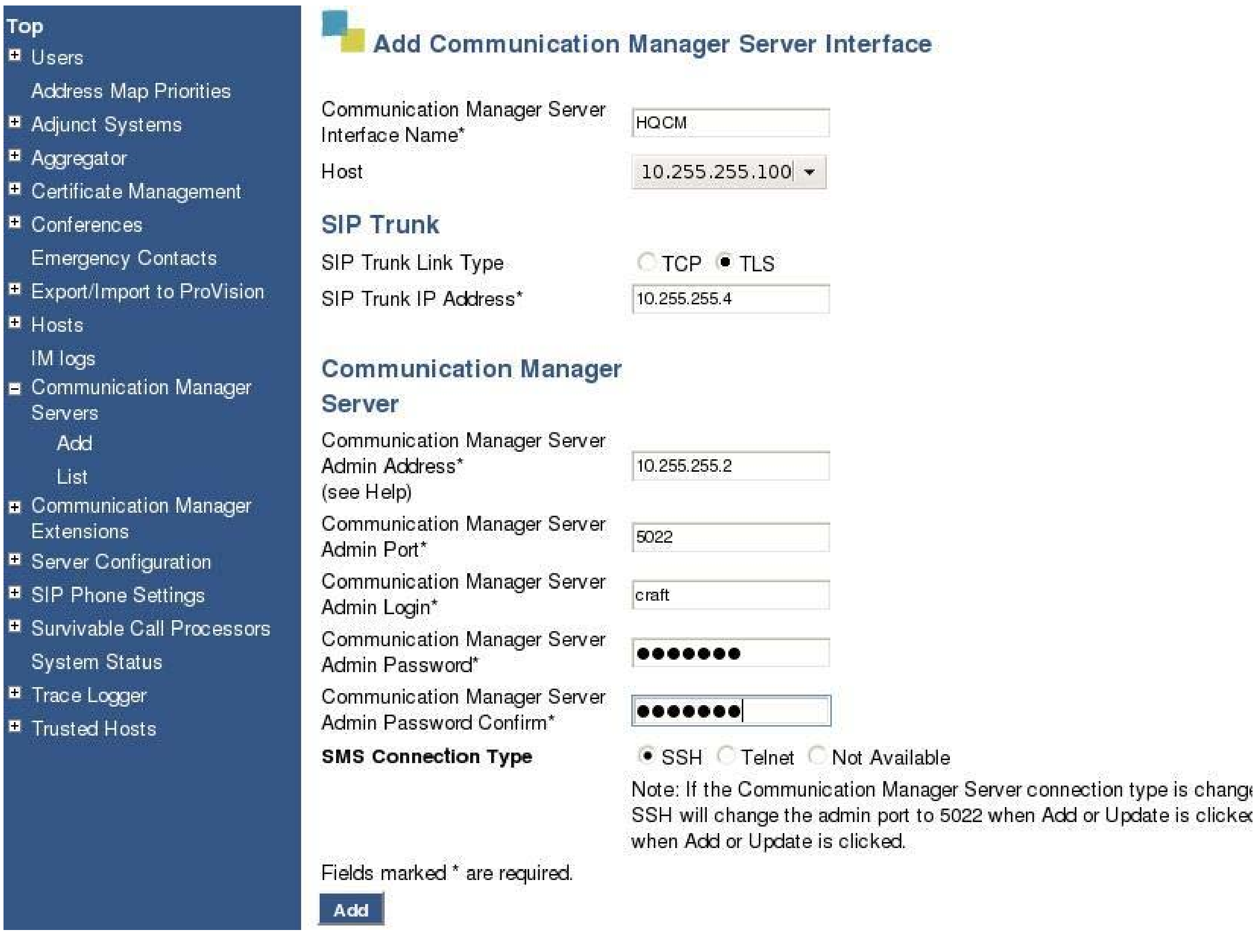
Note. The Survivable Call Processor configuration for the Cisco ISR is set to use the TCP transport protocol on port 5060. The Avaya 9600 SIP Phone will use these same parameters when registering with the Cisco ISR in survivable mode.

Log into the Avaya SIP Enablement Services web interface using appropriate user credentials. On the Avaya SIP Enablement Service web page interface, choose **Launch SES Administration Interface**. Choose **Survivable Call Processors → Add** to display the **Add Survivable Call Processor** screen. Enter the **Processor Name** (choose an appropriate name for the branch ISR) and **IP Address** (IP address of the branch ISR). Set **Protocols** to **TCP**. Press **Add** to continue.



5.2. Adding Avaya Communication Manager

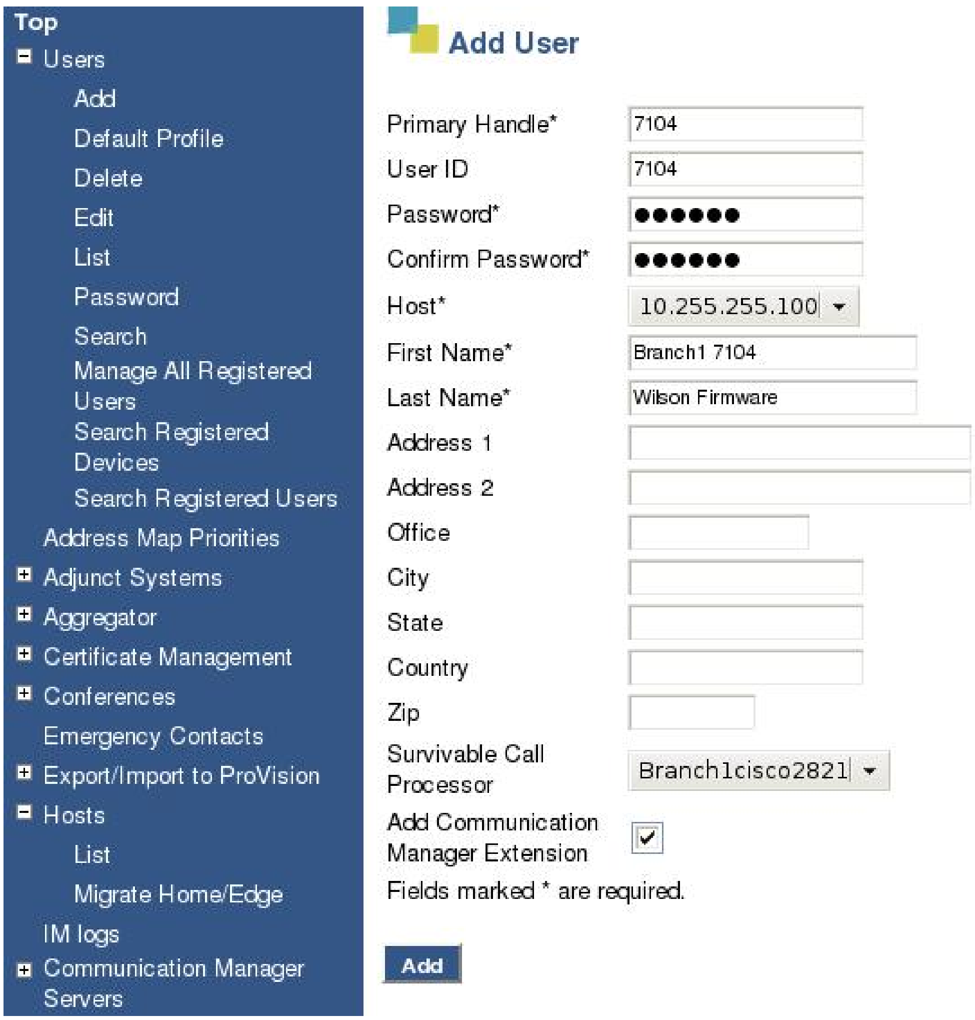
The following steps were used to add Avaya Communication Manager to the Avaya SIP Enablement Services configuration. On the Avaya SIP Enablement Service web page interface, choose **Launch SES Administration Interface**. Choose **Communications Manager Servers → Add**. Enter details for **Communications Manager Server Interface Name**, the **SIP Trunk Line Type** is set to **TLS**. For **SIP Trunk IP Address** enter the IP of the CLAN card. Enter the IP address of the **Communication Manager Server Admin Address**, **Communication Manager Server Admin Port**, and the login details. Leave the **SMS Connection Type** at default value SSH. Click **Add** to apply changes.



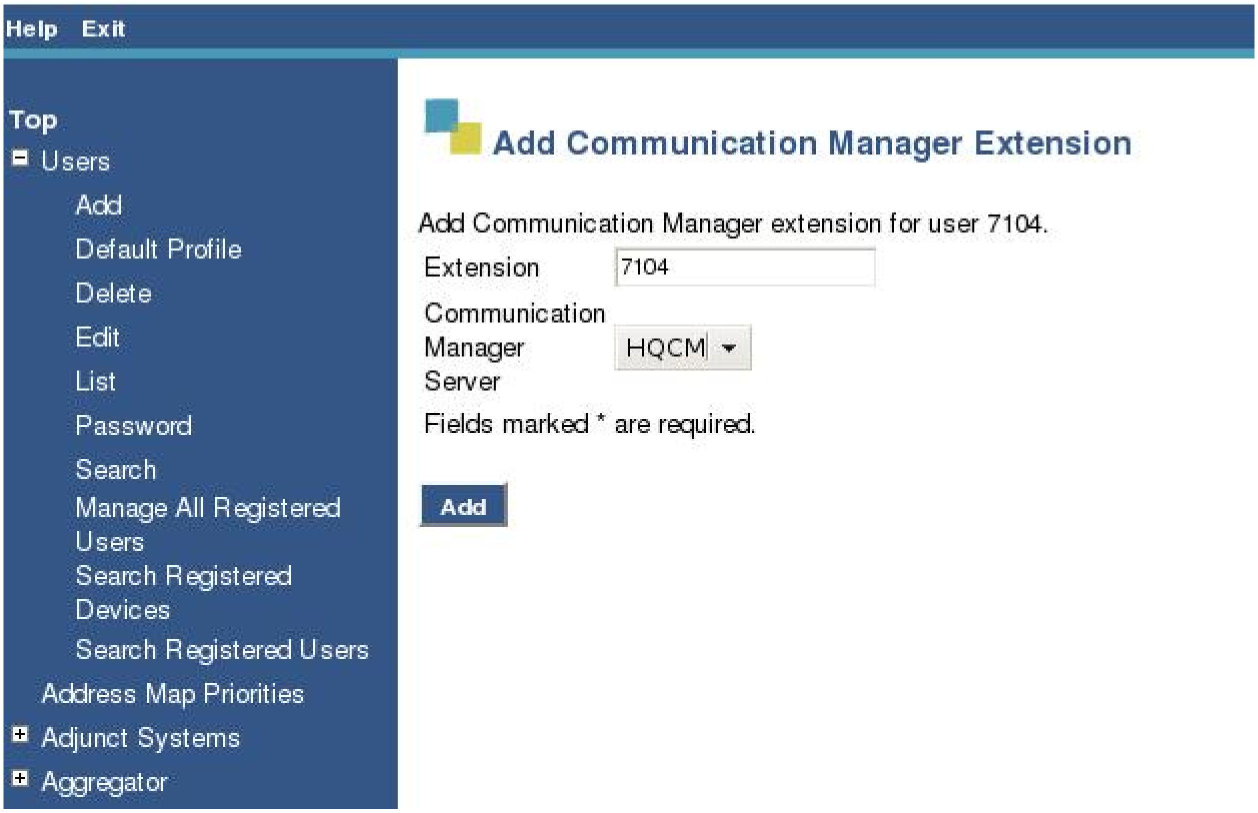
5.3. SIP User Accounts

5.3.1. Avaya 9600 SIP Phone Accounts

An account should be created for each Avaya 9600 SIP Phone user by selecting **Users → Add** from the Avaya SIP Enablement Services left navigation panel. The account should be configured with the Survivable Call Processor for the branch location that the user is located. Each user account should also be configured with a Communication Manager Extension. The screen below illustrates the creation of user account **7104** for **Branch1** for this sample configuration. Note that the Cisco ISR Survivable Call Processor **Branch1cisco2821** was selected for all **Branch1** 9600 SIP IP Deskphones. Also verify **Add Communication Manager Extension** is ticked. Click **Add** to apply changes.



The next screen displayed is the **Add Communication Manager Extension** screen, similar to the one shown below. Enter the appropriate extension, typically the same **Extension** as the Primary Handle of the user account. Click **Add** to apply changes. This Communication Manager Extension should also be created on Avaya Communication Manager as described in **Section 5.2.**



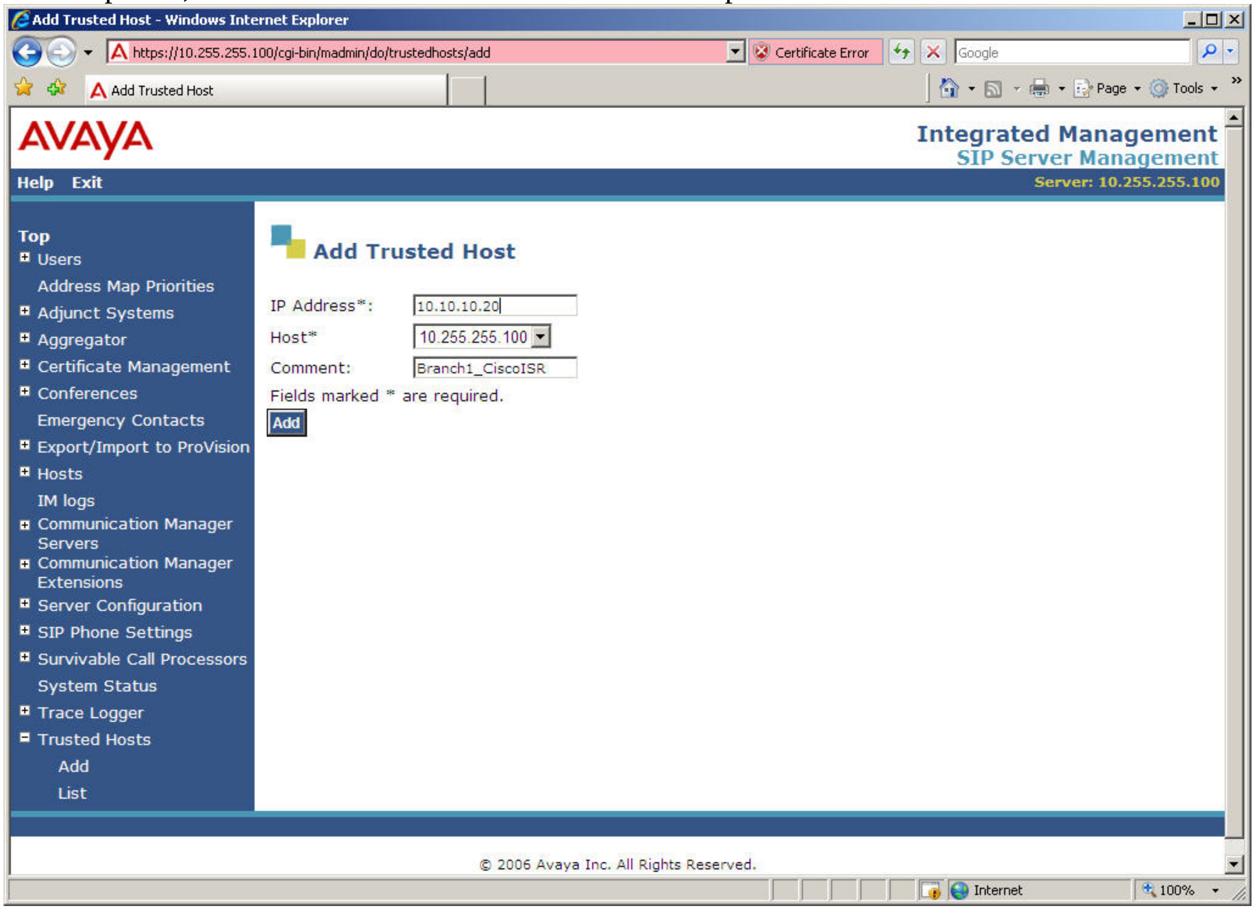
5.3.2. Cisco ISR FXS Analog Phone SIP User Account

Each Branch Cisco ISR FXS Analog Phone should be configured with a SIP user account on Avaya SIP Enablement Services and a corresponding extension on Avaya Communication Manager. The user is created as described in **Section 5.3.1.**

Note: When creating SIP user accounts for the Branch FXS ports, the Survivable Call Processor field is left blank. The Branch Cisco ISR provides survivable service for the FXS ports.

5.3.3. Cisco ISR FXS Analog Phone SIP User Account

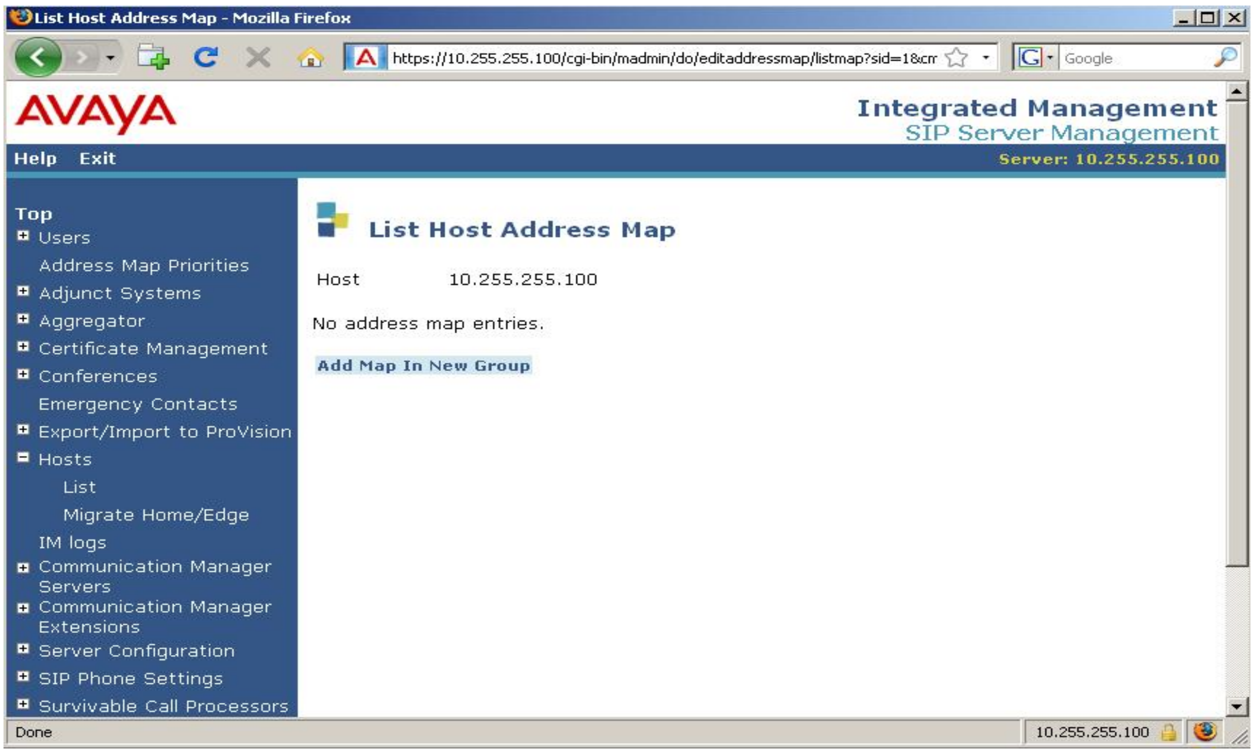
In order to allow the Branch Cisco ISR router to register its local FXS ports with the HQSES as SIP endpoints, each of the branch's will have to be set-up as Trusted Hosts.



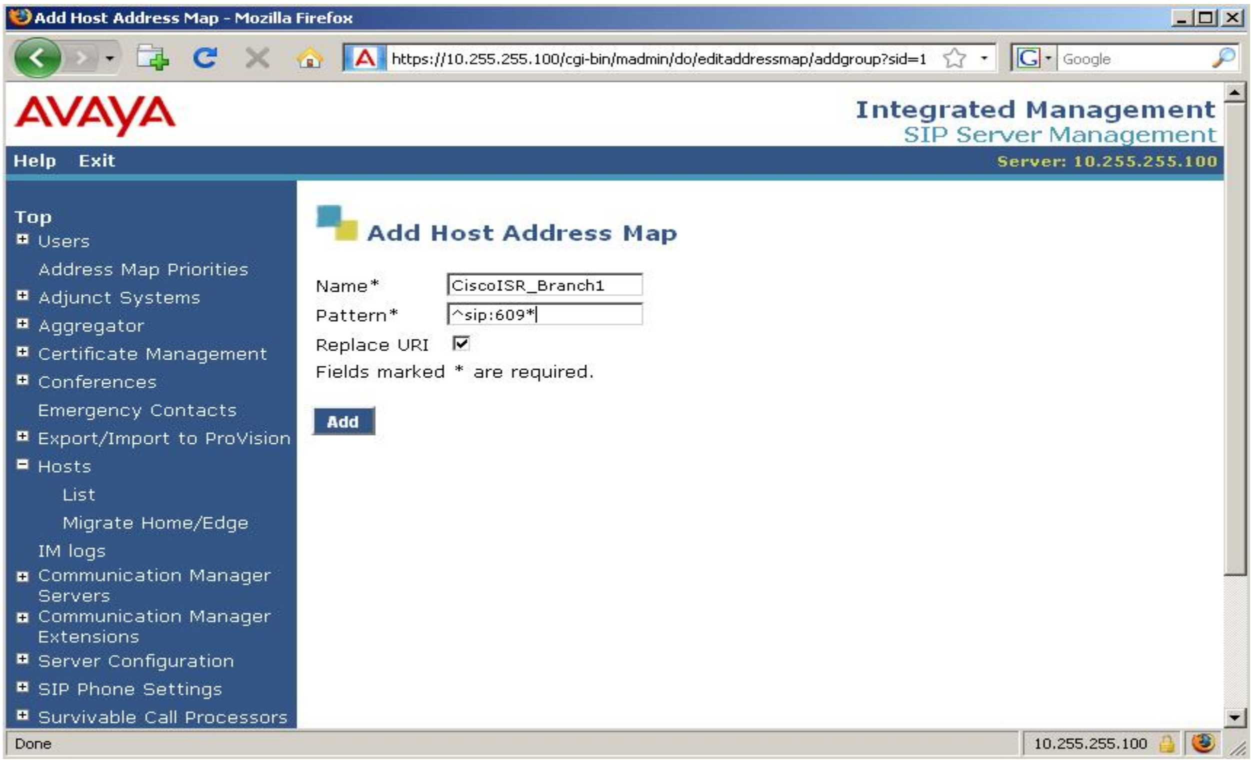
5.3.4. Host Address Map

For Branch endpoints, calls to Local PSTN will route to HQ Avaya Communications Manager and HQ Avaya SIP Enablement Services, the location based routing set-up at HQ will route the call back to the Branch Cisco ISR PSTN interface. In a typical enterprise scenario, there will be multiple remote branches, the HQ Avaya SES should route the call to the proper branch Cisco ISR.
To accomplish this, Host Address Maps are created by selecting **Hosts → List → Add Map In New Group** from the Avaya SES left navigation panel. The Map should be added to the Avaya SES server to which the branch Cisco ISR is registered.

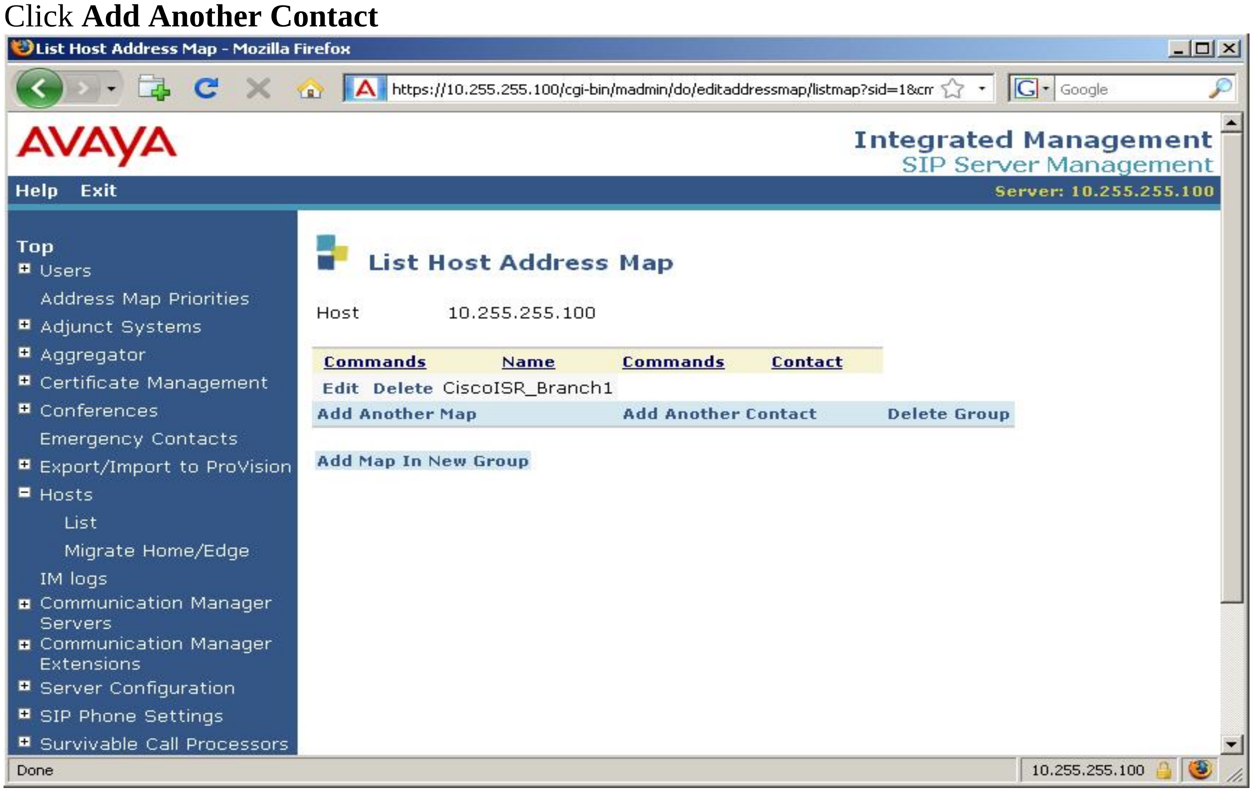
Click **Add Map In New Group**



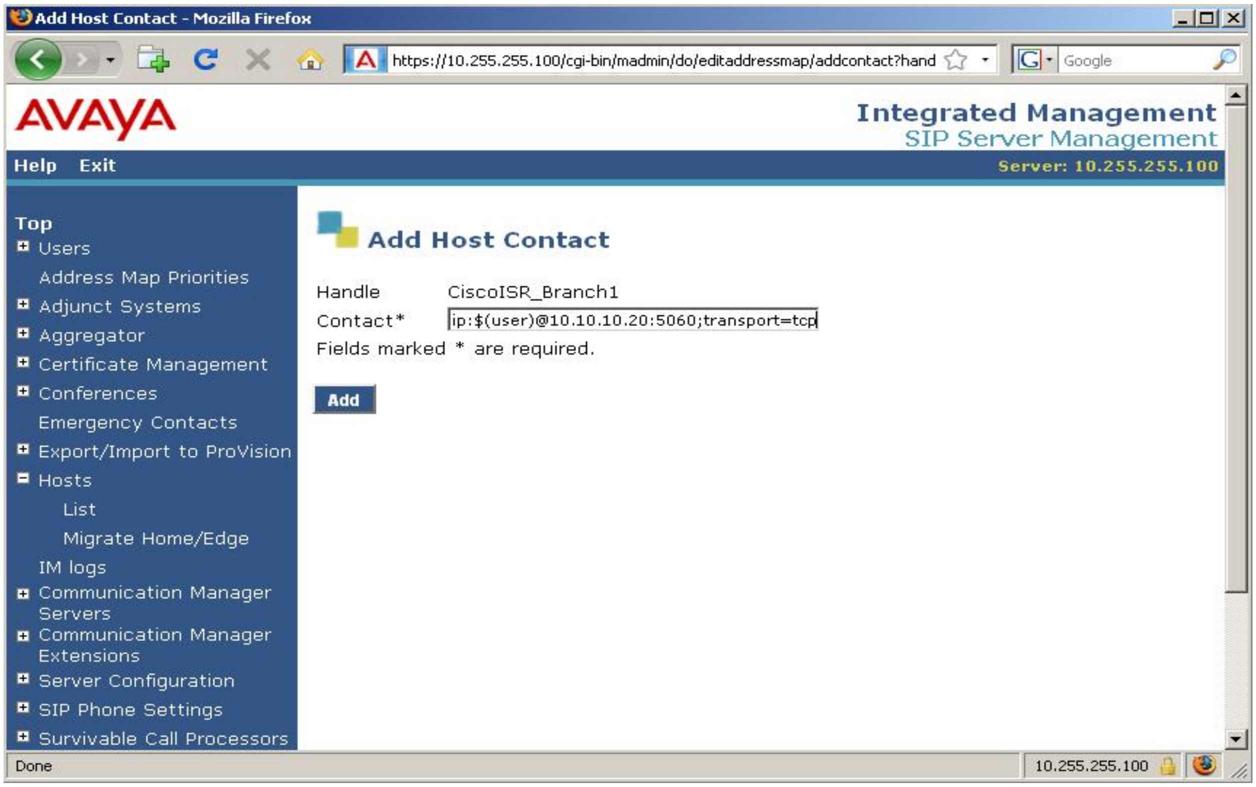
Enter the appropriate details for the **Name** e.g., **CiscoISR_Branch1** and **Pattern ^sip:609*.** The **Pattern** should match the configuration in the HQ Avaya Communications Manager for each remote branch. Click **Add** and **Continue** to return to the main Hosts config page**.**



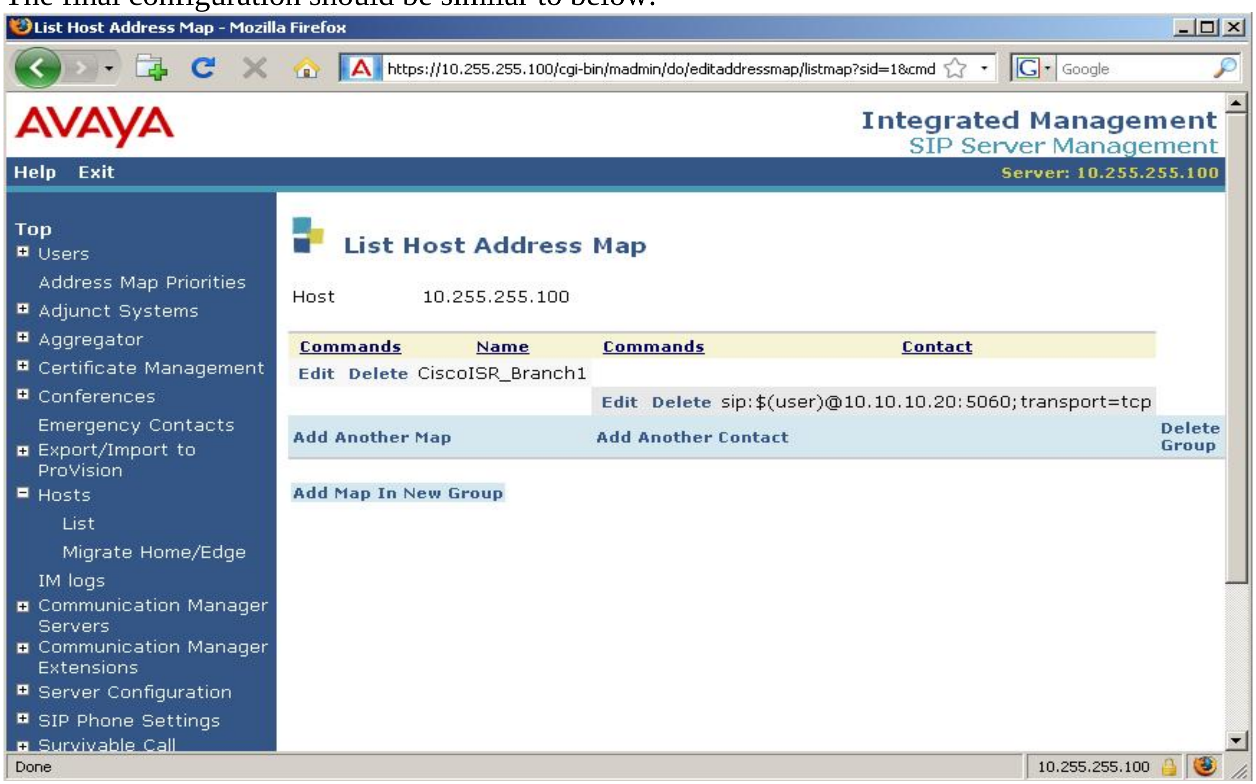
Click **Add Another Contact**



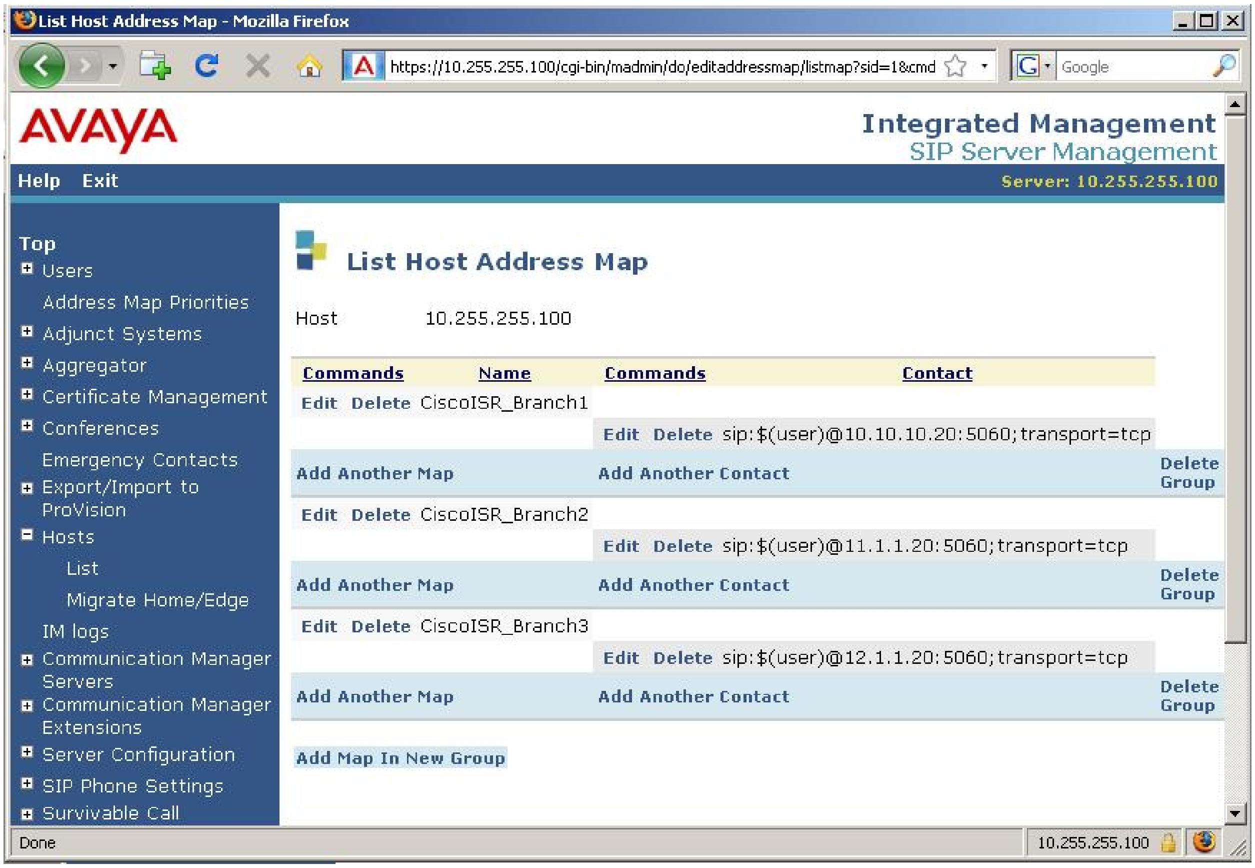
Enter the details for the **Contact** field, the format is **sip:$(user)@BranchCiscoISR_IP:5060;transport=tcp** where BranchCiscoISR_IP is the IP address of the Branch Cisco ISR. Click **Add** and **Continue** to apply changes.



The final configuration should be similar to below.



Repeat this for all remote branches.



6. Avaya Communication Manager

This section describes 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 is setup and the required licensing has already been administered.

6.1. Administer IP node-names

Use the command **change node-name ip** to enter the appropriate configuration menu. Add the **Name** and **IP Address** for the **clan** and **medpro** cards, and also for the HQ SIP Enablement Services, **HQSES**.

```
change node-names ip                                        Page   1 of   2
                                  IP NODE NAMES
    Name                IP Address
HQSES               10.255.255.100
clan                10.255.255.4
default             0.0.0.0
medpro              10.255.255.5
procr               10.255.255.2




( 5  of 5    administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

6.2. Administer IP Interfaces for CLAN and Medpro cards.

List the configuration details using the command **list configuration all**, take note of the **board number** for the **CONTROL-LAN** and **IP MEDIA PROCESSOR.**

```
list configuration all

                              SYSTEM CONFIGURATION

Board                                                         Assigned Ports
Number   Board Type              Code     Vintage    u=unassigned t=tti p=psa

01A00    POWER SUPPLY              655A
01A01    IP SERVER INTFC         TN2312BP HW28 FW044 01 02 03 04 05 06 07 08
01A02    CONTROL-LAN              TN799DP HW16 FW024 u  u  u  u  u  u  u  u
                                                     u  u  u  u  u  u  u  u
                                                     17
01A03    IP MEDIA PROCESSOR      TN2302AP HW32 FW117 01    03    05    07
01A04    DS1 INTERFACE           TN2464CP HW13 FW022 01 02 03 04 05 06 07 08
                                                     09 10 11 12 13 14 15 16
                                                     17 18 19 20 21 22 23 24
                                                     25 26 27 28 29 30 31 u
01A      IP SERVER INTFC         TN2312BP HW28 FW044 01 02 03 04 05 06 07 08
```

Enter the CLAN **IP INTERFACES** configuration menu using the command **change ip-interface 01a02**. Enter the **Node Name**, verify **IP Address,** enter **Subnet Mask**, **Gateway Address**, and verify **Enable Ethernet Port** is set to **y.**

```
change ip-interface 01a02                                         Page   1 of   2
                                  IP INTERFACES


                  Type: C-LAN
                  Slot: 01A02
           Code/Suffix: TN799  D
             Node Name: clan
            IP Address: 10.255.255.4
           Subnet Mask: 255.255.255.0                              Link: 1
       Gateway Address: 10.255.255.254
  Enable Ethernet Port? y                       Allow H.323 Endpoints? y
        Network Region: 1                        Allow H.248 Gateways? y
                  VLAN: n                          Gatekeeper Priority: 5

 Target socket load and Warning level: 400
       Receive Buffer TCP Window Size: 8320
                                ETHERNET OPTIONS
                  Auto? y
```

Enter the Medpro **IP INTERFACES** configuration menu using the command **change ip-interface 01a03.** Enter the **Node Name**, verify **IP Address**, enter the **Subnet Mask**, **Gateway Address** and verify **Enable Ethernet Port** is set to **y**.

```
change ip-interface 01a03                                         Page   1 of   1
                                  IP INTERFACES


                  Type: MEDPRO
                  Slot: 01A03
           Code/Suffix: TN2302
             Node Name: medpro
            IP Address: 10 .255.255.5
           Subnet Mask: 255.255.255.0
       Gateway Address: 10 .255.255.254
  Enable Ethernet Port? y
        Network Region: 1
                  VLAN: n



                                ETHERNET OPTIONS
                  Auto? y
```

6.3. Administer DS1 card

In this sample configuration, an E1 ISDN trunk is used to link the Headquarters Avaya Communications Manager to the Public Switched Telephone Network. Use the command **change ds1 01a04** to enter the **DS1 CIRCUIT PACK** configuration menu, set the **Line Coding** as **hdb3**, set the **Signaling Mode** as **isdn-pri**, **Connect** as **network**, and **Idle code** as **01010100**. These settings match the provider end settings of the E1 Trunk.

```
change ds1 01a04                                                Page   1 of   1
                                DS1 CIRCUIT PACK

            Location: 01A04                          Name: Trunk 20
            Bit Rate: 2.048                   Line Coding: hdb3

      Signaling Mode: isdn-pri
             Connect: network
   TN-C7 Long Timers? n                  Country Protocol: etsi
Interworking Message: PROGress
Interface Companding: alaw                           CRC? y
           Idle Code: 01010100
                              DCP/Analog Bearer Capability: 3.1kHz

                                         T303 Timer(sec): 4
                                        Disable Restarts? n

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

6.4. Locations

The locations of each branch as well as Headquarters should be defined within Avaya Communication Manager using the **change locations** command. The values used in the sample configuration are shown below. The location number (**Loc No**), **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      Disp   Prefix   Proxy Sel
 No                   Offset             FAC  FAC      Parm              Rte Pat
 1:  Headquarters    + 00:00   0                        1
 2:  Branch1         + 00:00   0    609                 1
 3:  Branch2         + 00:00   0    709                 1
 4:  Branch3         + 00:00   0    809                 1
 5:                      :
 6:                      :
 7:                      :
 8:                      :
 9:                      :
10:                      :
11:                      :
12:                      :
13:                      :
14:                      :
```

6.5. IP Codec set

The voice codec to be used throughout the enterprise are defined in the **IP Codec Set** form. For this sample configuration, a single **codec set** is used with a single codec type 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:
```

6.6. 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: du.rnd.avaya.com
    Name: Headquarters
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 values used in the sample configuration for **Branch 1 IP Network Region 2** are shown below. The **Location**, **Name**, **Codec Set** and **Authoritative Domain** field values shown are specific to this 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 2                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 2
Location: 2       Authoritative Domain: du.rnd.avaya.com
    Name: Branch1
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 2**. 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 2** and **network region 1**.

```
 display ip-network-region 2                                    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
 2   1    1     y    NoLimit                                             n   all
 2   2    1                                                                  all
 2   3
 2   4
 2   5
 2   6
 2   7
```

6.7. 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
 10 .10 .10 .0      10 .10 .10 .255      24       2        n
 10 .255.255.0      10 .255.255.255      24       1        n
 11 .1  .1  .0      11 .1  .1  .255      24       3        n
 12 .1  .1  .0      12 .1  .1  .255      24       4        n
```

6.8. HQ Avaya Communication Manager Signaling and Trunk Group Configuration

The HQ Avaya Communications Manager requires Signaling groups/Trunk groups for connectivity to HQ Avaya SIP Enablement Services and also to the Public Switched Telephone Network.

6.8.1. Create SIP Signaling / Trunk Groups

Adding a SIP signaling/trunk group for communications between HQ Avaya SIP Enablement Services and the HQ Avaya Communication Manager. To enter the new signaling group, use the command **add signaling-group 10**. Set the **Group Type** as **sip**, **Transport Method** as **tls**. Set the **Near-end Node Name** as **clan** and **Far-end Node Name** as the **HQSES**. Set the **Far-end Domain** to the appropriate value for the environment.

```
add signaling-group 10
                               SIGNALING GROUP

Group Number: 10             Group Type: sip
                       Transport Method: tls


  Near-end Node Name: clan                   Far-end Node Name: HQSES
Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                        Far-end Network Region: 1
      Far-end Domain: du.rnd.avaya.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? y
Session Establishment Timer(min): 3              Alternate Route Timer(sec): 6
```

To add a corresponding trunk group, use the command **add trunk-group 10**. Set **Group Type** as **sip**. Choose value for the **TAC**. Set **Group Name** to an appropriate name, **Service Type** to **tie**, **Signaling Group** to **10,** and configure the desired **Number of Members.**

Note: When creating SIP user accounts for the Branch FXS ports, the Survivable Call Processor field is left blank. The Branch Cisco ISR provides survivable service for the FXS ports.

```
add trunk-group 10                                                  Page   1 of  21
                                TRUNK GROUP

Group Number: 10                   Group Type: sip           CDR Reports: y
  Group Name: To HQSES                    COR: 1        TN: 1        TAC: 110
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                       Signaling Group: 10
                                                     Number of Members: 10

```

For location based routing, another Signaling / Trunk Group is created to route the local Branch PSTN call invite back to the HQSES. Adding a SIP signaling/trunk group for communications from HQ Avaya Communication Manager back to the HQ Avaya SIP Enablement Services. To enter the new signaling group, use the command **change signaling-group 11**. Set the **Group Type** as **sip**, **Transport Method** as **tls**. Set the **Near-end Node Name** as **clan** and **Far-end Node Name** as the **HQSES**, Set the **Far-end Domain** to the appropriate value for the environment.

```
change signaling-group 11
                              SIGNALING GROUP

Group Number: 11              Group Type: sip
                        Transport Method: tls








  Near-end Node Name: clan                    Far-end Node Name: HQSES
Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                         Far-end Network Region: 1
      Far-end Domain: du.rnd.avaya.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
```

To add a corresponding trunk group, use the command **add trunk-group 11**. Set **Group Type** as **sip**. Choose value for the **TAC**. Set **Group Name** to an appropriate name, **Service Type** to **tie**, **Signaling Group** to **11,** and configure the desired **Number of Members.**

```
 change trunk-group 11                                            Page   1 of  21
                               TRUNK GROUP

Group Number: 11                   Group Type: sip          CDR Reports: y
  Group Name: ACMtoSES BranchLocal        COR: 1        TN: 1        TAC: 118
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

                                                      Signaling Group: 11
                                                    Number of Members: 10
```

6.8.2. Create Route Pattern

Adding a route-pattern for the signaling/trunk groups created in the previous steps. To enter the configuration menu, use the command **change route-pattern 10.** Choose a relevant **Pattern Name** e.g., **HQSES.** Set the **Grp No** as the number given to the matching signaling/trunk group. In this case **10** and set **FRL** to **0**.

```
change route-pattern 10                                           Page   1 of   3
                    Pattern Number: 10  Pattern Name: HQSES
                             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: 10   0                                                                 n   user
 2:                                                                        n   user
 3:                                                                        n   user
 4:                                                                        n   user
 5:                                                                        n   user
 6:                                                                        n   user

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

Use the command **change route-pattern 11 to associate a route pattern with trunk group 11.** Choose a relevant **Pattern Name** e.g**., Back to HQSES.** Set the **Grp No** as the number given to the matching signaling/trunk group, in this case **11** and set **FRL** to **0**. The routing back to the HQSES includes removing the leading digit, in this case just the first digit , **No. Del Dgts** (Digits) is set to **1**.

```
.  change route-pattern 11                                          Page   1 of   3
                    Pattern Number: 10  Pattern Name: Back to HQSES
                             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: 11   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
```

6.8.3. Create ISDN Signaling / Trunk Group

Adding a SIP signaling/trunk group for communications between HQ Avaya Communication Manager and the Public Switched Telephone Network.

Note: The field **Primary D-Channel:** refers to the DS1 card slot, **01A04** and the data channel for E1 ISDN is **16**.

Enter the **SIGNALING GROUP** configuration menu using the command **add signaling-group 20**. Set the **Group Type** as **isdn-pri**, set the **Primary D-Channel** as **01A0416**. Set **Trunk Group for Channel Selection** as **20**, and the **TSC Supplementary Service Protoco**l as **c**.

```
add signaling-group 20
                                SIGNALING GROUP

 Group Number: 20              Group Type: isdn-pri
                   Associated Signaling? y          Max number of NCA TSC: 0
                      Primary D-Channel: 01A0416     Max number of CA TSC: 0
                                                  Trunk Group for NCA TSC: 20
       Trunk Group for Channel Selection: 20
      TSC Supplementary Service Protocol: c          Network Call Transfer? n
                      ETSI CCBS Support: both-directions
```

Add a trunk group using the command **add trunk-group 20**. Set the **Group Type** as **isdn**, choose an appropriate **Group Name**, set **TAC** to an appropriate value, in this case the next available value, **111**. **Carrier Medium** is **PRI/BRI** and **Service Type** is **public-ntwrk**.

```
add trunk-group 20                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 20                  Group Type: isdn          CDR Reports: y
  Group Name: OUTSIDE CALL               COR: 1        TN: 1        TAC: 111
   Direction: two-way        Outgoing Display? y         Carrier Medium: PRI/BRI
 Dial Access? y              Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n            TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
```

Complete configuration changes to the relevant trunk group. Navigate to page 5. For each channel, enter the **Port 01A04xx** and **Sig Grp 20.** Repeat this for all 30 channels.

```
change trunk-group 20                                           Page   5 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/30
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  30

       Port    Code Sfx Name        Night           Sig Grp
  1: 01A0401  TN2464 C                                20
  2: 01A0402  TN2464 C                                20
  3: 01A0403  TN2464 C                                20
  4: 01A0404  TN2464 C                                20
  5: 01A0405  TN2464 C                                20
  6: 01A0406  TN2464 C                                20
  7: 01A0407  TN2464 C                                20
  8: 01A0408  TN2464 C                                20
  9: 01A0409  TN2464 C                                20
 10: 01A0410  TN2464 C                                20
 11: 01A0411  TN2464 C                                20
 12: 01A0412  TN2464 C                                20
 13: 01A0413  TN2464 C                                20
 14: 01A0414  TN2464 C                                20
 15: 01A0415  TN2464 C                                20
```

6.8.4. Create ISDN Route Pattern

Enter the configuration menu using the command **change route-pattern 20**. Choose a relevant **Pattern Name** e.g., **DialoutPSTN**. Set the **Grp No** as **20** and **FRL** as **0**.

```
change route-pattern 20                                        Page   1 of   3
                   Pattern Number: 20  Pattern Name: DialoutPSTN
                            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: 20   0                                                            n   user
 2:                                                                   n   user
 3:                                                                   n   user
 4:                                                                   n   user
 5:                                                                   n   user
 6:                                                                   n   user

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

It is assumed that the **dialplan** and **aar analysis** will be configured to suit the particular list of extensions and call routing parameters. See **Appendix 12.1 & 12.2** for details of the **dialplan** and **aar analysis** used in this sample configuration.

6.8.5. Automatic Route Selection (ARS)

The Automatic Route Selection (ARS) entries highlighted in this section focus on the local and long distance dialing from branch locations. 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   6
                               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:
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code:
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA:        All:        Deactivation:
   Call Forwarding Enhanced Status:        Act:        Deactivation:
```

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 11** is used when a match is made on any of these ARS entries, in this example the digits matched will be **1609**.

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

         Dialed           Total     Route     Call   Node  ANI
         String          Min  Max  Pattern    Type   Num   Reqd
   1609                  9    9     11        natl         n
```

6.8.6. Automatic Route Selection 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. In this sample configuration long distance calls, 1 + 10 digits, will match the Dialed String of 1 with 11 digits and select Route Pattern 20.

Route Pattern 20 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 20, the associated PSTN Trunk Group and the Avaya G650 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                    9    9     20        op           n
    101xxxx0             18   18    deny      op           n
    101xxxx01            16   24    deny      iop          n
    101xxxx011           17   25    deny      intl         n
    101xxxx1             18   18    deny      fnpa         n
    10xxx0               6    6     deny      op           n
    10xxx0               16   16    deny      op           n
    10xxx01              14   22    deny      iop          n
    10xxx011             15   23    deny      intl         n
    10xxx1               16   16    deny      fnpa         n
```

6.9. HQ Avaya CM Stations and Off PBX Station mapping

Along with the regular stations added, for example, HQ H323 endpoints, there is also a requirement for all remote branch endpoints i.e., SIP and FXS Analog phones, to have **stations** entries added to the HQ Avaya Communication Manager. These stations will also be configured with **off-pbx-telephone** details.

For this sample configuration, the set-up of station **7104** is used as an example. To enter the configuration menu use the command **add station 7104**. Set the **Extension** to **7104**, and **Security Code** as **710400**, the same credentials used when creating a user on the HQSES . Choose a relevant **Name** e.g., **Branch1 7104.** The rest of the setting can be left as default.

```
add station 7104                                             Page   1 of   6
                                     STATION

Extension: 7104                         Lock Messages? n              BCC: 0
     Type: 9620                         Security Code: 710400          TN: 1
     Port: S00016                     Coverage Path 1:                COR: 1
     Name: Branch1 7104        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: 7104
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                            IP SoftPhone? n



                                              Customizable Labels? y
```

Add this station as an **off-pbx-telephone**, to enter the off-pbx-telephone station-mapping configuration menu, use the command **change off-pbx-telephone station-mapping 7104**. Set the **Application** as **OPS**, **Phone Number** as **7104**, **Trunk Selection** as **10** and **Config Set** as **1**.

```
change off-pbx-telephone station-mapping 7104                    Page   1 of   2

                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                       Selection   Set
 7104             OPS          -     7104                 10         1
                               -
                               -
                               -
                               -
                               -
                               -
                               -
```

7. Cisco Integrated Services Router

This section illustrates the configuration for the Cisco ISR 2821 to support the Avaya Communication Manager Survivable SIP Gateway Solution in a Distributed Trunking scenario. It is assumed that the basic configuration of the router is already complete, see **Section 13, References [9]** for details on Cisco ISR 2821 basic setup.

7.1. Cisco ISR Check system hardware.

To view the hardware detected by the Cisco ISR, use the command **show diag**. Connect to the Cisco ISR using the standard cisco console cable, or network terminal if the device is already configured for such.

```
Branch1#
Branch1#show diag
Slot 0:
        C2821 Motherboard with 2GE and integrated VPN Port adapter, 2 ports
        Port adapter is analyzed
        Port adapter insertion time 1d00h ago
        EEPROM contents at hardware discovery:
        PCB Serial Number        : FOC1011169N
        Hardware Revision        : 1.0
        Top Assy. Part Number    : 800-26921-01
        Board Revision           : A0
        Deviation Number         : 0
        Fab Version              : 03
        RMA Test History         : 00
        RMA Number               : 0-0-0-0
        RMA History              : 00
        Processor type           : 87
        Hardware date code       : 20060316
        Chassis Serial Number    : FCZ101872P0
        Chassis MAC Address      : 0017.5a5e.d968
        MAC Address block size   : 32
        CLEI Code                : CNMJ6P0BRB
        Product (FRU) Number     : CISCO2821
        Part Number              : 73-8853-03
        Version Identifier       : V02
        EEPROM format version 4
        EEPROM contents (hex):
          0x00: 04 FF C1 8B 46 4F 43 31 30 31 31 31 36 39 4E 40
          0x10: 03 E8 41 01 00 C0 46 03 20 00 69 29 01 42 41 30
          0x20: 88 00 00 00 00 02 03 03 00 81 00 00 00 00 04 00
          0x30: 09 87 83 01 32 18 9C C2 8B 46 43 5A 31 30 31 38
          0x40: 37 32 50 30 C3 06 00 17 5A 5E D9 68 43 00 20 C6
          0x50: 8A 43 4E 4D 4A 36 50 30 42 52 42 CB 8F 43 49 53
          0x60: 43 4F 32 38 32 31 20 20 20 20 20 20 82 49 22 95
          0x70: 03 89 56 30 32 20 D9 02 40 C1 FF FF FF FF FF FF

        PVDM Slot 0:                                PVDM resource for Analog Ports
        16-channel (G.711) Voice/Fax PVDMII DSP SIMM PVDM daughter card
        Hardware Revision        : 4.0
        Part Number              : 73-8538-05
        Board Revision           : B0
        Deviation Number         : 0
        Fab Version              : 04
        PCB Serial Number        : FOC1238489P
```

```
        RMA Test History        : 00
        RMA Number              : 0-0-0-0
        RMA History             : 00
        Processor type          : 00
        Product (FRU) Number    : PVDM2-16
        Version Identifier      : V01
        EEPROM format version 4
        EEPROM contents (hex):
          0x00: 04 FF 40 03 EF 41 04 00 82 49 21 5A 05 42 42 30
          0x10: 88 00 00 00 00 02 04 C1 8B 46 4F 43 31 32 33 38
          0x20: 34 38 39 50 03 00 81 00 00 00 00 04 00 09 00 CB
          0x30: 88 50 56 44 4D 32 2D 31 36 89 56 30 31 20 D9 02
          0x40: 40 C1 FF FF FF FF FF FF FF FF FF FF FF FF FF FF
          0x50: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
          0x60: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
          0x70: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF

        WIC Slot 0:                                     Analog Ports
        FXS Voice daughter card (4 port)
        Hardware Revision       : 3.1
        Part Number             : 73-6918-02
        Board Revision          : E0
        Deviation Number        : 0
        Fab Version             : 02
        PCB Serial Number       : FOC10232BED
        RMA Test History        : 00
        RMA Number              : 0-0-0-0
        RMA History             : 00
        Top Assy. Part Number   : 800-17016-02
        Connector Type          : 01
        Product (FRU) Number    : VIC-4FXS/DID=
        EEPROM format version 4
        EEPROM contents (hex):
          0x00: 04 FF 40 00 3A 41 03 01 82 49 1B 06 02 42 45 30
          0x10: 88 00 00 00 00 02 02 C1 8B 46 4F 43 31 30 32 33
          0x20: 32 42 45 44 03 00 81 00 00 00 00 04 00 C0 46 03
          0x30: 20 00 42 78 02 05 01 FF FF FF FF FF FF FF FF FF
          0x40: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
          0x50: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
          0x60: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
          0x70: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF

Slot 1:
        High Density Voice Port adapter
        Port adapter is analyzed
        Port adapter insertion time 1d00h ago
        EEPROM contents at hardware discovery:
        Hardware Revision       : 1.1
        Top Assy. Part Number   : 800-03567-01
        Board Revision          : G0
        Deviation Number        : 0-0
        Fab Version             : 02
        PCB Serial Number       : JAD062601RH
        RMA Test History        : 00
        RMA Number              : 0-0-0-0
        RMA History             : 00
        Product (FRU) Number    : NM-HDV=
```

```
        EEPROM format version 4
        EEPROM contents (hex):
          0x00: 04 FF 40 00 CC 41 01 01 C0 46 03 20 00 0D EF 01
          0x10: 42 47 30 80 00 00 00 00 02 02 C1 8B 4A 41 44 30
          0x20: 36 32 36 30 31 52 48 03 00 81 00 00 00 00 04 00
          0x30: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
          0x40: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
          0x50: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
          0x60: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
          0x70: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF

        HDV SIMMs: Product (FRU) Number: PVDM-12=          PVDM Resources for E1 ISDN
         SIMM slot 0: PVDM-12 SIMM present.
         SIMM slot 1: PVDM-12 SIMM present.
         SIMM slot 2: PVDM-12 SIMM present.
         SIMM slot 3: PVDM-12 SIMM present.
         SIMM slot 4: PVDM-12 SIMM present.

        WIC Slot 0:                                         E1 ISDN interface
        E1 (2 Port) Multi-Flex Trunk (Drop&Insert) WAN Daughter Card
        Hardware revision 1.0           Board revision B0
        Serial number     28249531      Part number    800-04615-03
        FRU Part Number     VWIC-2MFT-E1-DI=
        Test history      0x0           RMA number     00-00-00
        Connector type    PCI
        EEPROM format version 1
        EEPROM contents (hex):
          0x20: 01 25 01 00 01 AF 0D BB 50 12 07 03 00 00 00 00
          0x30: 58 00 00 00 02 04 12 00 FF FF FF FF FF FF FF FF

        HDV firmware: Compiled Fri 19-Nov-04 14:23 by michen
        HDV memory size 524280 heap free 167869
Branch1#
```

7.2. Cisco ISR Configuration for Remote Branch Distributed Trunking

Cisco ISR configuration commands are applied with the same syntax as listed below. The configuration changes added to the basic setup of the ISR are highlighted in bold.
Use the command **config** to enter the configuration mode.

To set the **hostname** use the command as listed in the Cisco ISR Configuration file **Section 7.2.1**

Branch1#**config**	**Enter the Configuration mode**
Configuring from terminal, memory, or network [terminal]?	**Hit RETURN for default**
Enter configuration commands, one per line. End with CNTL/Z.	
Branch1(config)#**hostname Branch1**	**Set hostname to Branch1**
Branch1(config)#**exit**	**Exit Configuration mode**

To save the changes use the command **copy running-config startup-config.**

Branch1#**copy running-config startup-config**	**Saves the running config**
Destination filename [startup-config]?	**Hit RETURN for default**
Building configuration...	
[OK]	
Branch1#	

7.2.1. Cisco ISR Configuration File

To view the contents of the **running** configuration file, use the command **show configuration.** The configuration changes made to the ISR for this testing are highlighted below, which an explanation of what the command does to the ISR, listed opposite in blue highlight.

Use command : show configuration

```
version 12.4
no service pad
service tcp-keepalives-in
service tcp-keepalives-out
service timestamps debug datetime msec localtime show-timezone
service timestamps log datetime msec localtime show-timezone
service password-encryption
service sequence-numbers
!
hostname Branch1                                   Set the name of the ISR
!
boot-start-marker
boot system flash c2800nm-ipvoicek9-mz.124-20.YA2.bin
boot-end-marker
!
security authentication failure rate 10 log
security passwords min-length 6
logging message-counter syslog
logging buffered 4096
logging console critical
enable secret 5 $1$1p.4$/DL.s2Gi1tjmfqYgt0ie..
enable password 7 121E0403170009013A2E3679616676
!
aaa new-model
!
aaa authentication login local_auth local
!
aaa session-id common
no network-clock-participate slot 1
!
voice-card 0
 no dspfarm
!
voice-card 1
!!
ip cef
!
no ip bootp server
no ip domain lookup
ip domain name du.rnd.avaya.com                    Set the domain name
login block-for 100 attempts 100 within 10
no ipv6 cef
multilink bundle-name authenticated
!
isdn switch-type primary-5ess                      Set the PSTN switch type
!
voice dsp waitstate 0
!
```

```
voice service voip                                   Enable VOIP service on the ISR
 media statistics
 allow-connections sip to sip                        Allow SIP to SIP call control
 redirect ip2ip                                      Enable IP to IP calls
 sip                                                 Enter SIP configuration
  registrar server expires max 600 min 60
  redirect contact order best-match
!
voice class codec 1                                  Create a voice class codec
 codec preference 1 g711ulaw                         Set g711ulaw as preference 1
 codec preference 2 g711alaw                         Set g711ulaw as preference 1
!
voice register pool  1                               Create SIP registration pool
 id network 10.10.10.0 mask 255.255.255.0            Allow SIP registration from IP range
 application session                                 Enable application SIP
 preference 2                                        Set local branch proxy preference
 dtmf-relay rtp-nte
 voice-class codec 1                                 Use voice class codec 1
!!
crypto pki trustpoint TP-self-signed-759887484
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-759887484
 revocation-check none
 rsakeypair TP-self-signed-759887484
!
crypto pki certificate chain TP-self-signed-759887484
 certificate self-signed 01 nvram:IOS-Self-Sig#1.cer
!
username root privilege 15 secret 5 $1$EHll$oMIBXiWiM6VirCOp.htI7/
username SDM privilege 15 password 7 08324843584B5643
username ciscoHQ privilege 15 password 7 13061E0108030C3B
archive
 log config
  hidekeys
!
controller E1 1/0/0                                  Enter E1 Controller Configuration
 pri-group timeslots 1-31                            Set time-slots for E1 ISDN
!
controller E1 1/0/1
 shutdown
!
interface GigabitEthernet0/0                         Enter the GB Ethernet Configuration
 description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
 ip address 10.10.10.20 255.255.255.0                Set the Controller IP address
duplex auto                                          Set connection parameters
 speed auto                                          Set connection parameters
!
interface GigabitEthernet0/1
 no ip address
 no ip redirects
 no ip unreachables
 no ip proxy-arp
 shutdown
 duplex auto
 speed auto
 no mop enabled
!
```

interface Serial1/0/0:15	**Enter Serial Interface Configuration**
no ip address	**no IP address assigned**
isdn switch-type primary-5ess	**Set the switch type**
isdn incoming-voice voice	**Treat Incoming calls as voice**
isdn send-alerting	
isdn sending-complete	
!	
ip default-gateway 10.10.10.254	**Set default IP gateway**
ip route 10.0.0.0 255.0.0.0 10.10.10.254	**Set Static route**
ip route 10.10.10.0 255.255.255.0 10.10.10.254	**Set Static route**
!	
control-plane	
!	
call fallback active	**Enable SRST**
!	
voice-port 0/0/0	
!	
voice-port 0/0/1	
!	
voice-port 0/0/2	
!	
voice-port 0/0/3	
!	

voice-port 1/0/0:15	**Enter voice-port configuration**
playout-delay maximum 170	**Settings for packet jitter**
playout-delay nominal 80	**Settings for packet jitter**
playout-delay minimum low	**Settings for packet jitter**
no comfort-noise	
bearer-cap 3100Hz	**Information transfer capability**
!	
no ccm-manager fax protocol cisco	
!	
mgcp fax t38 ecm	
!	
dial-peer voice 7000 voip	**Create dial-peer 7000 voip**
description To HQSES 7... extensions pref 0	
destination-pattern 7...	**All 4 digit numbers starting with 7**
voice-class codec 1	
session protocol sipv2	
session target sip-server	**Send call to sip-server**
!	
dial-peer voice 10 pots	**Create dial-peer 10 pots**
destination-pattern 7110	**Extension 7110**
port 0/0/0	**FXS port 0**
forward-digits all	**Pass all digits**
!	
dial-peer voice 11 pots	**Create dial-peer 11 pots**
destination-pattern 7111	**Extension 7111**
port 0/0/1	**FXS port 1**
forward-digits all	**Pass all digits**
!	
dial-peer voice 30000 voip	**Create dial-peer 30000 pots**
description PSTN etxn's 3.... via HQCM	
destination-pattern 3....	**All 5 digit numbers starting with 3**
voice-class codec 1	
session protocol sipv2	
session target sip-server	**Send call to sip-server**
!	
dial-peer voice 7 pots	**Create dial-peer 7 pots**
description SurvMode HQ extn's 7... via Branch E1	
preference 1	**Set as 2nd Preference**
destination-pattern 7...	**All 4 digit numbers starting with 7**
port 1/0/0:15	
forward-digits all	
prefix 62	**Add digits 62 to the dialout number**
!	
dial-peer voice 3 pots	**Create dial-peer 3 pots**
description SurvMode PSTN extn 3.... via Branch E1	
preference 1	**Set as 2nd Preference**
destination-pattern 3....	**All 5 digit numbers starting with 3**
port 1/0/0:15	**Send call to Serial 1/0/0/:15 (PSTN)**
forward-digits all	**Pass all digits**
!	
dial-peer voice 63200 pots	**Create dial-peer 63200 pots**
description Incoming PSTN	
incoming called-number 7...	**All 4 digit incoming calls start with 7**
direct-inward-dial	**Incoming**
port 1/0/0:15	**Serial 1/0/0/:15 (PSTN)**
!	

```
!
dial-peer voice 609 pots                              Create dial-peer 609
 description Distributed Local PSTN
 destination-pattern 609.....                         All 8 digit numbers starting with 609
 port 1/0/0:15                                        Send call to Serial 1/0/0/:15 (PSTN)
 forward-digits 5                                     Pass trailing 5 digits, remove 609
!
dial-peer voice 91609 pots                            Create dial-peer 91609
 description Distributed Local PSTN Surviv Mode
 destination-pattern 91609.....                       All 10 digit numbers starting with 91609
 port 1/0/0:15                                        Send call to Serial 1/0/0/:15 (PSTN)
 forward-digits 5                                     Pass trailing 5 digits, remove 609
!
sip-ua                                                Enter the ISR SIP User Agent Config.
 registrar ipv4:10.255.255.100 expires 3600           Enable SIP Reg. for FXS Ports
 sip-server ipv4:10.255.255.100                       set IP of primary SIP Server
!
line con 0
 exec-timeout 5 0
 login authentication local_auth
 transport output telnet
line aux 0
 exec-timeout 15 0
 login authentication local_auth
 transport output telnet
line vty 0 4
 access-class 23 in
 privilege level 15
 password 7 0822455D0A165445
 no exec
 transport preferred none
 transport input none
!
scheduler allocate 20000 1000
end
```

7.2.2. SIP-UA Keep-Alive Feature

With regards to a **keep-alive** feature on the Cisco ISR configuration, there are two options. Standard icmp ping or a SIP message **keep-alive**. The SIP message keep-alive mechanism may be more suitable for production environments. This configuration is not listed in the **show configuration** output on the Cisco ISR shown in **Section 7.2.1**. The following command show hot to set up the **sip-ua keepalive** feature to contact the HQSES.

```
Branch1#config                                                  Enter Config menu
Configuring from terminal, memory, or network [terminal]?
Enter configuration commands, one per line.  End with CNTL/Z.
Branch1(config)#sip-ua                                          Enter sip-ua config menu
Branch1(config-sip-ua)#
Branch1(config-sip-ua)#keepalive target ipv4:10.255.255.100 tcp Enter the keepalive parameters
Branch1(config-sip-ua)#exit                                     Exit from sip-ua config menu
Branch1(config)#exit                                            Exit from config menu
```

The Branch Cisco ISR will send a **keepalive** request in the form of a SIP **options** message. HQSES simply responds with a **200 OK**. To save the ISR configuration use the command: **copy running-config startup-config**

8. 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 the Avaya one-X Deskphone SIP for 9600 Series IP Telephones Administrator Guide **[1]** before setting or changing any of the parameters.

46xxsettings.txt Parameter Name	Value Used in Sample Configuration	Description
SIP_CONTROLLER_LIST	10.255.255.100:5061;transport=tls	Avaya SIP Enablement Services includes the option Survivable Call Processor, see **Section 5.1**. In the scenario where there are multiple remote branches, this is the preferred option to use. In this set-up the HQ SIP Enablement Services (10.255.255.100) is primary and the Branch Cisco ISR (10.10.10.20) is secondary.
FAILBACK_POLICY	Auto	While in Survivable Mode, determines the mechanism to use to fail back to the centralised 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 centralised 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.
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 should be the same for all SIP controllers in the configuration. SIPDOMAIN is set to **du.rnd.avaya.com** in the sample configuration.
DIALPLAN	7xxx|91xxxxxxxxxx|6xxxxx|#[1-9]|*[1-9]	Enables the acceleration of dialing when the WAN is down and the Cisco ISR SRST 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 Cisco ISR dial-peers.

9. Verification Steps

Follow these steps to verify the core environment setup.

9.1. Timing Expectations for Fail-over to Branch Cisco ISR

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 Branch Cisco ISR, 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 90 seconds. With multiple identical idle phones in the same branch, it would not be unusual for some phones to register to the Branch Cisco ISR before others, with the earliest registering in approximately 30 seconds and the latest registering in approximately 90 seconds.

9.2. Timing Expectations for Fail-back to HQSES

This section is intended to set *approximate* expectations for the length of time before Avaya 9600 SIP Telephones registered to the Branch Cisco ISR 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 within 30 seconds. 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.

9.3. Network Connectivity

From a command terminal in the Branch1 location, verify network connectivity between the HQ and the remote Branch by issuing a ping command to the following IP Addresses:

HQCLAN 10.255.255.4
HQSES 10.255.255.100

```
C:\>ping 10.255.255.4

Pinging 10.255.255.4 with 32 bytes of data:

Reply from 10.255.255.4: bytes=32 time=1ms TTL=63
Reply from 10.255.255.4: bytes=32 time<1ms TTL=63
Reply from 10.255.255.4: bytes=32 time<1ms TTL=63
Reply from 10.255.255.4: bytes=32 time<1ms TTL=63

Ping statistics for 10.255.255.4:
    Packets: Sent = 4, Received = 4, Lost = 0 (0% loss),
Approximate round trip times in milli-seconds:
    Minimum = 0ms, Maximum = 1ms, Average = 0ms

C:\>
```

On the HQ Avaya Communication Server, enter the **sat** interface and issue ping command to verify network connectivity from HQCM to the various environment components. The example below show a ping test from the Avaya Communication Manager to the CLAN card.

```
ping ip-address 10.255.255.4

                          PING RESULTS

  End-pt IP        Port      Port Type  Result    Time(ms)  Error Code

  10.255.255.4     01A0217   ETH-PT     PASS      0
```

Another example testing the connectivity to the Branch1 Cisco ISR.

```
ping ip-address 10.10.10.20

                          PING RESULTS

  End-pt IP        Port      Port Type  Result    Time(ms)  Error Code

  10.10.10.20      01A0217   ETH-PT     PASS      1
```

9.4. Verify Signaling and Trunk Group Status

On the HQ Avaya Communication Server, use the sat interface to verify the **TRUNK GROUP** and **SIGNALING GROUP** status. Use the command **status trunk 10** and **status signaling-group 10,** as in the examples below, repeat this for any other signaling/trunk groups created.

```
status trunk 10

                             TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0010/001 T00001   in-service/idle    no
0010/002 T00002   in-service/idle    no
0010/003 T00003   in-service/idle    no
0010/004 T00004   in-service/idle    no
0010/005 T00005   in-service/idle    no
0010/006 T00006   in-service/idle    no
0010/007 T00007   in-service/idle    no
0010/008 T00008   in-service/idle    no
0010/009 T00009   in-service/idle    no
0010/010 T00010   in-service/idle    no
```

```
status signaling-group 10
                         STATUS SIGNALING GROUP

        Group ID: 10                              Active NCA-TSC Count: 0
      Group Type: sip                              Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```

9.5. Verify Dial routes on Avaya Communication Server

Check routing of calls to 31047 extension, digital phone extension at Headquarters.

```
list aar route-chosen 31047#

                          AAR ROUTE CHOSEN REPORT

    Location:  all                    Partitioned Group Number:  1

     Dialed            Total        Route    Call      Node
     String          Min    Max    Pattern   Type     Number    Location

31                   5     5        20       aar                  all

  Actual Outpulsed Digits by Preference (leading 35 of maximum 42 digit)

1:                                      9:
2:                                     10:
3:                                     11:
4:                                     12:
5:                                     13:
6:                                     14:
7:                                     15:
8:                                     16:
```

Check routing of calls to Branch1 **632104** via Headquarters PSTN.

```
list aar route-chosen 632104#

                          AAR ROUTE CHOSEN REPORT

    Location:  all                    Partitioned Group Number:  1

     Dialed            Total        Route    Call      Node
     String          Min    Max    Pattern   Type     Number    Location

632                  6     6        20       aar                  all

  Actual Outpulsed Digits by Preference (leading 35 of maximum 42 digit)

1:                                      9:
2:                                     10:
3:                                     11:
4:                                     12:
5:                                     13:
6:                                     14:
7:                                     15:
8:                                     16:
```

Check location based routing of calls to Branch1 **9160931047** routed back to HQSES.

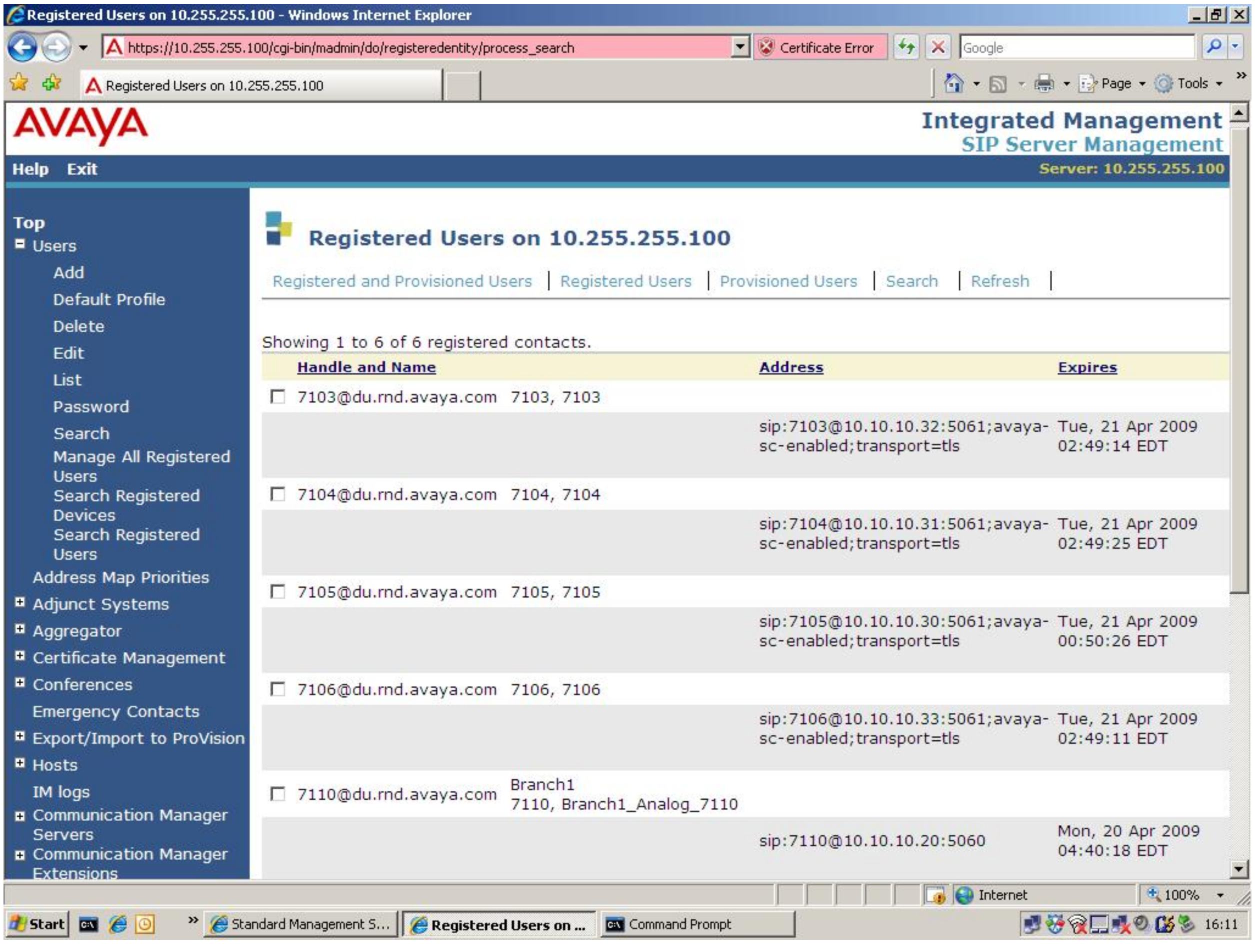
```
list ars route-chosen 9160931047#

                           ARS ROUTE CHOSEN REPORT

    Location:  all                    Partitioned Group Number:  1

     Dialed            Total        Route    Call      Node
     String          Min    Max    Pattern   Type     Number    Location

 9                    6     6        2       hpna                  all

  Actual Outpulsed Digits by Preference (leading 35 of maximum 42 digit)

 1:                                      9:
 2:                                     10:
 3:                                     11:
 4:                                     12:
 5:                                     13:
 6:                                     14:
 7:                                     15:
 8:                                     16:
```

In this sample configuration, all SIP phones and Branch1 analog extensions are configured as off-pbx stations, to list these use the command **off-pbx-telephone station-mapping**. Calls to these extensions via the HQ Avaya Communication Manager, are passed onto the HQ Avaya SIP Enablement Services.

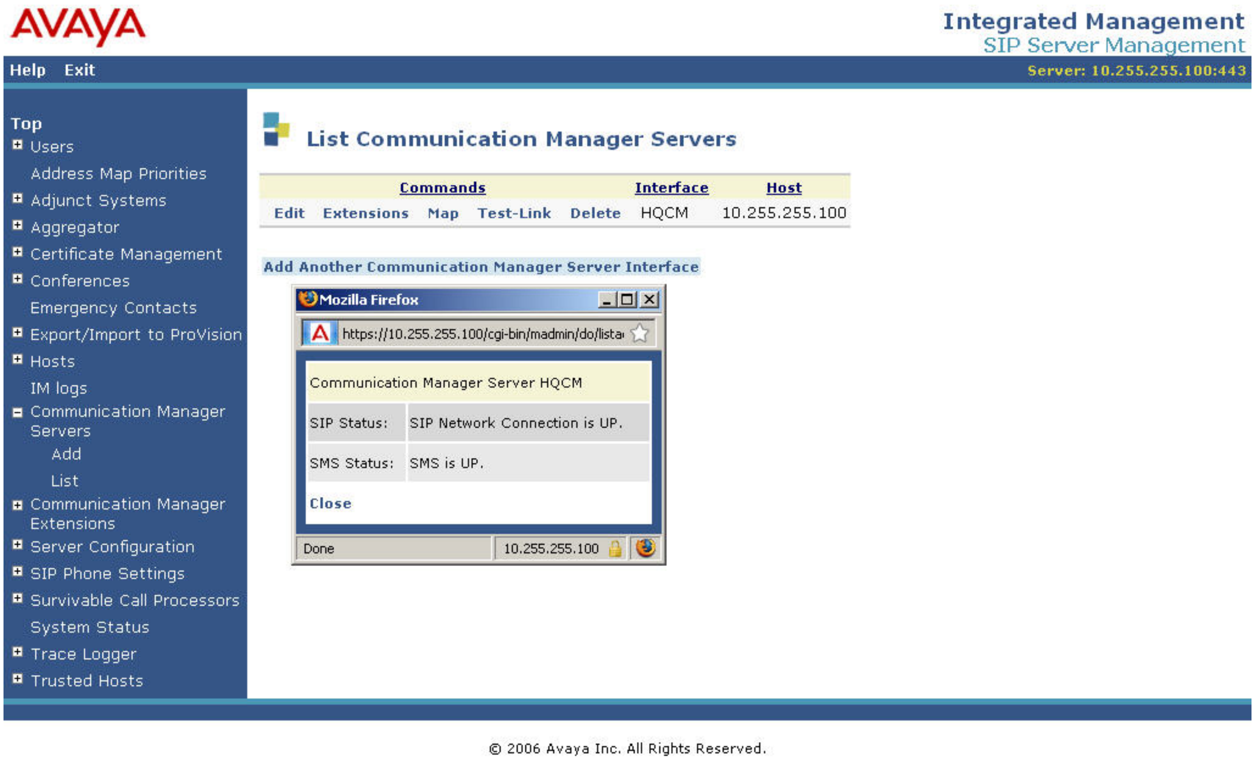
```
display off-pbx-telephone station-mapping                       Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                       Selection   Set
 2001              OPS          -      2001               10         1
 2002              OPS          -      2002               10         1
 2003              OPS          -      2003               10         1
 3001              OPS          -      3001               10         1
 7101              OPS          -      7101               10         1
 7102              OPS          -      7102               10         1
 7103              OPS          -      7103               10         1
 7104              OPS          -      7104               10         1
 7105              OPS          -      7105               10         1
 7106              OPS          -      7106               10         1
 7108              OPS          -      7108               10         1
 7109              OPS          -      7109               10         1
 7110              OPS          -      7110               10         1
 7111              OPS          -      7111               10         1
```

9.6. Verify SIP Enablement Services

On the HQ Avaya SIP Enablement Services List registered users. Using the web interface, **Launch SES Administration Interface**. Select **Users→Search Registered Users→ Search**. All the HQ and Branch1 SIP phones registered should be listed here, along with any FXS analog stations at the Branch1 location.



Verify connectivity to HQCM. In the **SES Administration Interface**, select **Communications Manager Servers→List** and use **Test-Link.** A second window will pop up with the test result, as shown below.



9.7. Verify Branch1 Cisco ISR

On the Branch1 Cisco ISR carry out the following command to verify set-up using a terminal interface.

To ping the HQSES from the Branch Cisco ISR, use the command **ping ip 10.255.255.100**

```
Branch1#ping ip 10.255.255.100

Type escape sequence to abort.
Sending 5, 100-byte ICMP Echos to 10.255.255.100, timeout is 2 seconds:
!!!!!
Success rate is 100 percent (5/5), round-trip min/avg/max = 1/1/4 ms
Branch1#
```

To verify dial-peer's, use the command **show dial-peer voice summary.** Any phones which are in **simultaneous** proxy registration, should have dial-peers automatically created, one for each proxy control, HQSES (10.255.255.100) and Branch1 Cisco ISR (10.10.10.20).

```
Show dial-peer voice summary                                  [Normal Mode]

TAG    TYPE   MIN    OPER       PREFIX    DEST   PREF PTHRU    SESS-TAR      STAT   PORT
7000   voip   up     up                   7...    0      syst  sip-server
10            pots   up     up            7110    0                          up     0/0/0
11            pots   up     up            7111    0                          up     0/0/1
30000         voip   up     up            3....   0      syst  sip-server
7             pots   up     up     62      7...   1                          up   1/0/0:15
3             pots   up     up            3....   1                          up   1/0/0:15
63200         pots   up     up                    0                        down  1/0/0:15
40001   voip  up     up                   7104    2      syst  ipv4:10.10.10.31:506
40002   voip  up     up                   7104    1      syst  ipv4:10.255.255.100:
40003   voip  up     up                   7105    2      syst  ipv4:10.10.10.30:506
40004   voip  up     up                   7105    1      syst  ipv4:10.255.255.100:
40005   voip  up     up                   7106    2      syst  ipv4:10.10.10.33:506
40006   voip  up     up                   7106    1      syst  ipv4:10.255.255.100
```

In Survivable mode, the Cisco ISR creates dial-peers for each of the phones switching over to survivable mode, these phones were set with **alternative** proxy registration.

```
Show dial-peer voice summary                                  [Survivable Mode]

TAG    TYPE   MIN    OPER       PREFIX    DEST   PREF PTHRU    SESS-TAR      STAT   PORT
7000   voip   up     up                   7...    0      syst  sip-server
10            pots   up     up            7110    0                          up     0/0/0
11            pots   up     up            7111    0                          up     0/0/1
30000         voip   up     up            3....   0      syst  sip-server
7             pots   up     up     62      7...   1                          up   1/0/0:15
3             pots   up     up            3....   1                          up   1/0/0:15
63200         pots   up     up                    0                        down  1/0/0:15
40001   voip  up     up                   7104    2      syst  ipv4:10.10.10.31:506
40002   voip  up     up                   7104    1      syst  ipv4:10.255.255.100:
40003   voip  up     up                   7105    2      syst  ipv4:10.10.10.30:506
40004   voip  up     up                   7105    1      syst  ipv4:10.255.255.100:
40005   voip  up     up                   7106    2      syst  ipv4:10.10.10.33:506
40006   voip  up     up                   7106    1      syst  ipv4:10.255.255.100
40007   voip  up     up                   7103    2      syst  ipv4:10.10.10.32:506
40008   voip  up     up                   7103    1      syst  ipv4:10.255.255.100:
40009   voip  up     up                   7108    2      syst  ipv4:10.10.10.36:506
40010   voip  up     up                   7108    1      syst  ipv4:10.255.255.100:
40011   voip  up     up                   7109    2      syst  ipv4:10.10.10.35:506
40012   voip  up     up                   7109    1      syst  ipv4:10.255.255.100:
```

Verify FXS phone registration with the HQSES, using the command **show sip register status.** The branch1 analog endpoints should display **yes** for registered status.

```
Branch1#show sip register status
Line                                peer        expires(sec)  registered
================================  ==========  ============  ==========
3....                               3           73            no
609.....                            609         73            no
7...                                7           73            no
7110                                10          2239          yes
7111                                11          2384          yes
91609.....                          91609       73            no
```

Check the status of the PSTN interface using the commands **show isdn status.** Check that it is in an **ACTIVE** state.

```
Branch1#show isdn status
Global ISDN Switchtype = primary-5ess
ISDN Serial1/0/0:15 interface
        dsl 0, interface ISDN Switchtype = primary-5ess
    Layer 1 Status:
        ACTIVE
    Layer 2 Status:
        TEI = 0, Ces = 1, SAPI = 0, State = MULTIPLE_FRAME_ESTABLISHED
    Layer 3 Status:
        0 Active Layer 3 Call(s)
    Active dsl 0 CCBs = 0
    The Free Channel Mask:  0xFFFF7FFF
    Number of L2 Discards = 0, L2 Session ID = 1
    Total Allocated ISDN CCBs = 0
```

Display channel status using the command **show isdn service.** Verify that the **State** is **0** for all channels except the data channel 16.

```
Branch1#show isdn service
PRI Channel Statistics:
ISDN Se1/0/0:15, Channel [1-31]
  Configured Isdn Interface (dsl) 0
   Channel State (0=Idle 1=Proposed 2=Busy 3=Reserved 4=Restart 5=Maint_Pend)
    Channel :  1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
    State   :  0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 3 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
   Service State (0=Inservice 1=Maint 2=Outofservice 8=MaintPend 9=OOSPend)
    Channel :  1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
    State   :  0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 2 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
Branch1#
```

10. Troubleshooting

10.1. Avaya one-X IP phone 9600, doesn't switch to Survivable mode

There are several reasons why the phone may not switch over to survivable mode and register with the Branch Cisco ISR. The following are a few key points to check.

- Verify firmware release installed on the phone. On the Avaya one-X IP phone, select **menu**, scroll down to **About Avaya one-X** and select **OK.** Check the **Version** number displayed, it should be **2.4.8.24** or later.
- Check the configuration file **46xxsettings.txt** on the file server, verify the main parameters with **Section 7** above.
- Check the configuration on the phone. On the Avaya one-X IP phone, enter the **Administrator Procedures** menu. Scroll **down to the option SIP….** Choose **Select**, and **then choose SIP Proxy Settings**, Verify that th**e cor**rec**t IP** a**ddress**es **of HQSES and Branch Cisco ISR are listed.** Verify that HQ**SES is** t**he act**i**ve pro**xy **in normal mode, i.e. the tick marked.**
- **V**erify that the **featu**re **call fallback** is **Active** on the **Cisco ISR. Using** th**e** comman**d show call fallback stats**

```
Branch1#show call fallback stats
VoIP Fallback Statistics:
Fallback Mode : Active
Total accepted calls: 0
Total rejected calls: 0
Total cache overflows: 0
Branch1#
```

10.2. No audio in local branch call

If a call can be establish but the audio is not present, it maybe related to the **ip route** set-up on the Cisco ISR. Check the current setup using **show ip route.**

```
Branch1#show ip route
Gateway of last resort is not set

     10.0.0.0/8 is variably subnetted, 2 subnets, 2 masks
C       10.10.10.0/24 is directly connected, GigabitEthernet0/0
S       10.0.0.0/8 [1/0] via 10.10.10.254
Branch1#
```

Also verify the default gateway using **show default-gateway.**

```
Branch1#show ip default-gateway
10.10.10.254
Branch1#
```

If these points check out ok, capture an ethereal trace of the SIP signaling at the ethernet port of the Cisco ISR. Create a mirrored port on the switch being used. Check the SIP Invites Session Description Protocol SDP information and also capture an ethernet trace at the Cisco ISR, filter for RTP data to see where that data is being sent.

10.3. WAN connection is restored, the phones do not switch back

If the branch phones do not switch back to the HQSES after the WAN connection is restored, check the settings on the phone.
Enter the 9600 IP phone **Admin Procedures**, navigate to the **SIP** option in the menu. Select OK and choose **SIP Global Settings**. Check the **Failback Policy**, it should be set to **Auto**.

10.4. Response from HQSES min-se timer too small

In this instance the session timer on the HQSES is smaller than the Branch Cisco ISR.
To check the value of the minimum session timer on the Cisco ISR use the command **show sip-ua min-se.**

```
Branch1#show sip-ua min-se
SIP UA MIN-SE Value (seconds)
Min-SE: 1800
Branch1#
```

If the HQSES indicates that the minimum session timer is too small, the value can be changed on the Cisco ISR within the **sip** configuration mode

```
Branch1#configure
Configuring from terminal, memory, or network [terminal]?
Enter configuration commands, one per line.  End with CNTL/Z.
Branch1(config)#voice service voip
Branch1(conf-voi-serv)#sip
Branch1(conf-serv-sip)#min-se 1200
Branch1(conf-serv-sip)#exit
Branch1(conf-voi-serv)#exit
Branch1(config)#exit
Branch1#copy run
Branch1#copy running-config startup-config
Destination filename [startup-config]?
Building configuration...
[OK]
Branch1#
Branch1#show sip-ua min-se
SIP UA MIN-SE Value (seconds)
Min-SE: 1200
Branch1#
```

11. Conclusion

SIP endpoints deployed at remote branch locations risk a loss of service if a break in connectivity to the centralised SIP call control platform occurs. Connectivity loss can be caused by WAN access problems being experienced at the branch or network problems at the central 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, incorporating a Cisco Integrated Service Router at the remote branch to minimize service disruptions to remote branch SIP endpoints.

12. Appendix

12.1. HQ Avaya Communications Manager Dialplan

To enter the **dialplan** configuration menu use the command **change dialplan analysis**. In this sample configuration, extensions **711x** and **710x** were used for all test phones. For this the configuration **Dialed String 711** and **710**, **Total Length 4**, **Call Type ext** were used. Also a dial out to the PSTN circuit, used **632xxx**. The configuration for this, **Dialed String 632**, **Total Length 6**, **Call Type aar**.

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

       Dialed   Total  Call    Dialed   Total  Call      Dialed   Total  Call
       String   Length Type    String   Length Type      String   Length Type
     100          4    ext
     11           3    dac
     20           4    ext
     22           5    aar
     30           4    ext
     31           5    aar
     632          6    aar
     710          4    ext
     711          4    ext
     9            1    fac
```

12.2. HQ Avaya Communications Manager Automatic Alternate Routing (AAR)

Automatic Alternate Routing is set-up using the command **change aar analysis 1.** In this sample configuration, two possible dialout strings were setup to gain access to the PSTN circuit. **632xxx** was used to dial to the Branch1 via PSTN and 31xxx was used as the access number for dialing to digital stations via PSTN.

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

         Dialed           Total     Route    Call   Node  ANI
         String          Min  Max  Pattern   Type   Num   Reqd
   3                     7    7     999      aar          n
   31                    5    5     20       aar          n
   4                     7    7     999      aar          n
   5                     7    7     999      aar          n
   6                     7    7     999      aar          n
   632                   6    6     20       aar          n
   7                     7    7     999      aar          n
   8                     7    7     999      aar          n
   9                     7    7     999      aar          n
                                                          n
```

13. Additional 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 Cisco ISR in a Centralised Trunking Configuration*

[5] *Avaya Communication Manager Survivable SIP Gateway Solution using the AudioCodes MP-114 in a Centralised Trunking Configuration – Issue 1.0*

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

[7] *The following Cisco ISR 2821 references are relevant to these Application Notes:*

[8] *Cisco Unified Survivable Remote Site Telephony Version 4.1*

[9] *Dial Peer Configuration on Voice Gateway Routers 12.4T*

[10] *Cisco 2800 Series Integrated Services Routers Quick Start Guide*

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com