Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunks among Ingate SIParator, Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager - Issue 1.0

Abstract

These Application Notes describe a sample configuration for a network that uses Avaya Aura™ Session Manager to connect Ingate SIParator and Avaya Aura™ Communication Manager using SIP trunks.

The Ingate SIParator is a SIP Session Border Controller (SBC) that manages and protects the flow of SIP signaling and related media across an untrusted IP network. The compliance testing focused on telephony scenarios between an enterprise site, where the Ingate SIParater, Session Manager and Communication Manager were located, and a second site simulating a service provider service node.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe a sample configuration for a network that uses Avaya Aura™ Session Manager to connect Ingate SIParator and Avaya Aura™ Communication Manager using SIP trunks.

The compliance testing focused on telephony scenarios between an enterprise site, where the Ingate SIParater, Session Manager and Communication Manager were located, and a second site simulating a service provider service node.

1.1. Interoperability Compliance Testing

The compliance testing focused on interoperability between Ingate SIParator and Session Manager / Communication Manager by making calls between the enterprise site and a second site simulating a service provide service node that were connected through the SIParator using direct SIP trunks. The following functions and features were tested in the compliance test:

- Calls from both SIP and non-SIP endpoints between sites
- G.711u and G.729A codec support
- Proper recognition of DTMF transmissions by navigating voicemail menus
- Proper operation of voicemail with message waiting indicators (MWI)
- PBX features including Multiple Call Appearances, Hold, Transfer, and Conference
- Extended telephony features using Communication Manager Feature Name Extensions (FNE) such as Call Forwarding, Conference On Answer, Call Park, Call Pickup, Automatic Redial and Automatic Call Back and Send All Calls.
- Proper system recovery after SIParator restart and/or re-establishment of broken IP connectivity.

1.2. Support

Technical support for Ingate SIParator can be obtained by contacting Ingate at

- EMEA Phone: +46-13-21 08 52
- NA Phone: +1-866-809-0002
- Email: support@ingate.com
- Web: http://www.ingate.com

2. Configuration

Figure 1 illustrates the test configuration. The test configuration shows two sites connected via a SIP trunk across an untrusted IP network: the main enterprise site and a second site that simulates a service provider service node. The main site has a Juniper Networks Netscreen-50 firewall at the edge of the network restricting unwanted traffic between the untrusted network and the main enterprise site. Also connected to the edge of the main site is a SIParator Session Border Controller (SBC). The public side of the SIParator is connected to the untrusted network and the private side is connected to the trusted corporate LAN.

All SIP traffic between sites flows through the SIParator. In this manner, the SIParator can protect the main site's infrastructure from any SIP-based attacks. The voice communication across the untrusted network uses SIP over TCP and RTP for the media streams. All non-SIP traffic bypasses the SIParator and flows directly between the untrusted network and the private LAN of the enterprise if permitted by the data firewall.

Also connected to the corporate LAN at the main site are:

- A Session Manager and its companion Avaya Aura™ System Manager. The Session Manager serves as a SIP routing hub and System Manager provides management functions for Session Manager.
- An Avaya S8300B Server running Communication Manager in an Avaya G700 Media Gateway. Avaya IA 770 Intuity Audix is also running on the Avaya S8300B Server to provide Voice Mail functionality.
- An Avaya S8500 Server running Avaya Aura™ SIP Enablement Services that provides SIP registrar and proxy server functions for SIP endpoints in the enterprise IP telephony network.
- An HTTP server for SIP phones at the enterprise site to obtain their configuration information.

The Session Manager connects the SIParator and Communication Manager using SIP trunks. Endpoints include both SIP and non-SIP endpoints. An ISDN-PRI trunk connects the media gateway to the PSTN.

Located at the 2nd site simulating a service provider service node is a SIP Enablement Services server and a Communication Manager with both SIP and non-SIP endpoints.

The SIP endpoints located at both sites are registered to the local SIP Enablement Services. Each site has a separate SIP domain: ***business.com*** for the main site and ***bigtime.com*** for the 2nd site. SIP and H.323 telephones at both sites use the local HTTP server to obtain their configuration files.

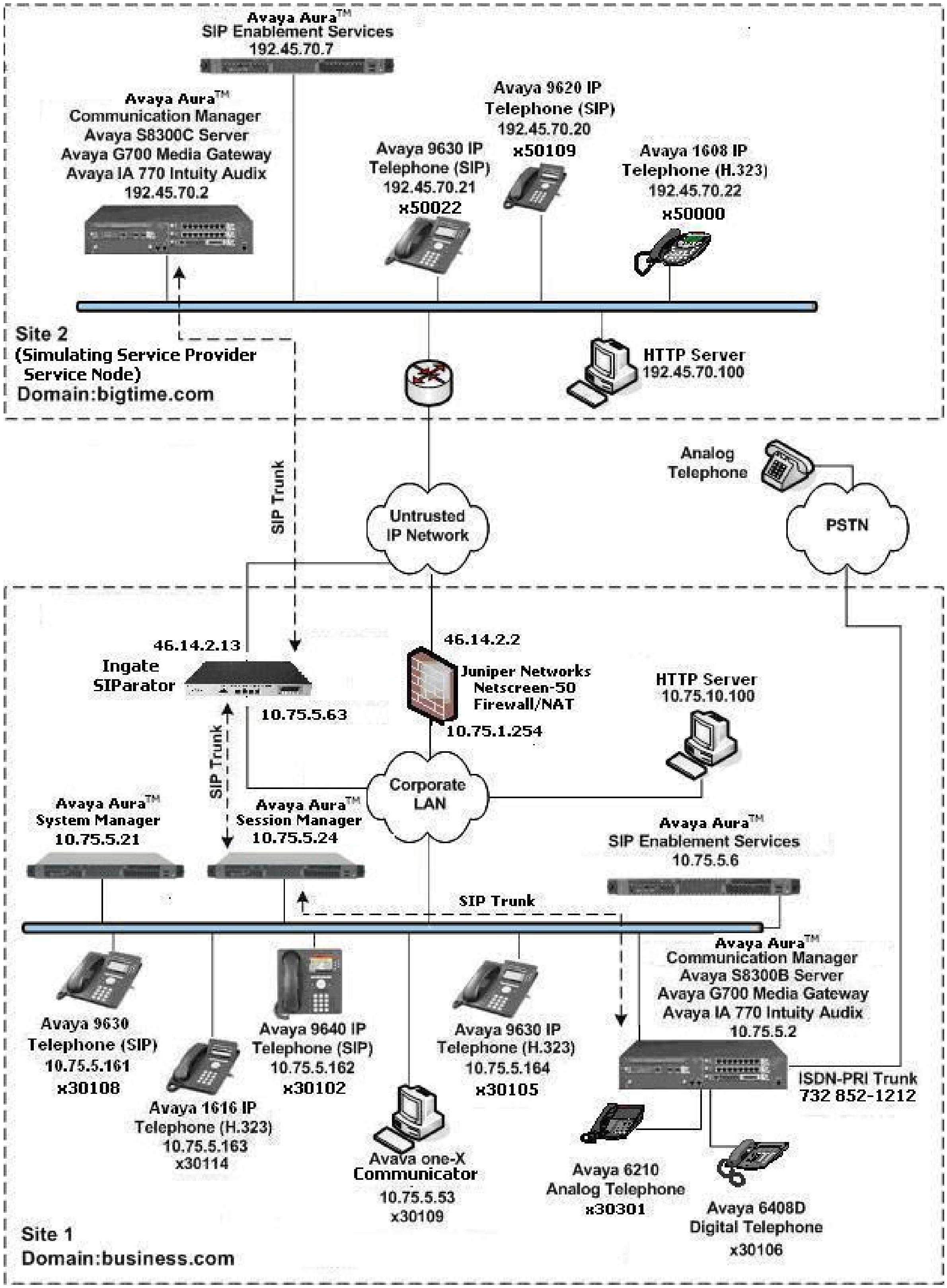


Figure 1: SIParator SIP Trunking Test Configuration

3. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8300B/C Server with Avaya G700 Media Gateway Avaya IA 770 Intuity Audix	Avaya Aura™ Communication Manager 5.2 (R015x.02.0.947.3 with update 17294)
Avaya S8500 Server	Avaya Aura™ SIP Enablement Services 5.2 (SES-5.2.0.0-947.3b with update SES-2.0.947.3-SP1)
Avaya 9600 Series IP Telephones (SIP)	Avaya one-X™ Deskphone Edition SIP 2.2
Avaya 9600 Series IP Telephones (H.323)	Avaya one-X™ Deskphone Edition H.323 Release 3.0
Avaya 1616 IP Telephone (H.323)	Avaya one-X™ Deskphone Value Edition Release 1.100
Windows PC (Soft Phone)	Windows XP Professional SP2 Avaya one-X™ Communicator (SIP) R1.030-SP3-16918
Avaya 6408D Digital Telephone	-
Avaya 6210 Analog Telephone	-
Analog Telephone	-
Windows Server (HTTP Server)	Windows Server 2003 SP2
Juniper Networks Netscreen-50	5.4.0r9.0
Ingate SIParator with installed modules: • Standard SIP features • SIP Trunking • Remote SIP Connectivity (NAT Traversal) • Failover • VPN (IPsec and PPTP)	4.7.1

4. Configure Communication Manager

This section describes the Communication Manager configuration at the main enterprise site to support the network shown in **Figure 1**. It assumes the procedures necessary to support SIP and connectivity to SIP Enablement Services have been performed as described in [3] and [5]. It also assumes that an off-PBX station (OPS) has been configured on Communication Manager for each SIP endpoint in the configuration as described in [3] and [4].

This section is divided into two parts. **Section 4.1** summarizes the user-defined parameters used in the installation procedures that are important to understanding the solution as a whole. It will not attempt to show the installation procedures in their entirety. It also describes any deviations from the standard procedures, if any.

Section 4.2 will describe procedures beyond the initial SIP installation procedures that are necessary for connecting Communication Manager to Avaya Aura™ Session Manager.

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

Note that in the case of the compliance test, a second site comprised of an Communication Manager and SIP Enablement Services was set up to simulate a service provider service node, therefore the configuration described in this section must be repeated for the Communication Manager at the 2nd site using values appropriate from **Figure 1**. This includes but is not limited to the IP addresses, SIP domain and user extensions. The specific differences will be called out in the configuration details in this section. A complete set of the key configuration screens on Communication Manager at site 2 is included as an appendix.

4.1. Summary of Initial SIP Installation

This section summarizes the applicable user-defined parameters used during the SIP installation procedures.

Step	Description
1.	**IP network region** The Avaya S8300B Server, SIP Enablement Services and IP (H.323/SIP) endpoints were located in a single IP network region (IP network region 1) using the parameters described below. Use the **display ip-network-region** command to view these settings. The example below shows the values used for the compliance test. ▪ **Authoritative Domain**: ***business.com*** This field was configured to match the domain name configured on SIP Enablement Services. This name will appear in the “From” header of SIP messages originating from this IP region. ▪ **Name**: ***Default*** Any descriptive name may be used. ▪ **Intra-region IP-IP Direct Audio**: ***yes*** **Inter-region IP-IP Direct Audio**: ***yes*** By default, IP-IP direct audio (media shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. Shuffling can be further restricted at the trunk level on the **Signaling Group** form. ▪ **Codec Set**: ***1*** The codec set contains the set of codecs available for calls within this IP network region. This includes SIP calls since all necessary components are within the same region.

```
display ip-network-region 1                                    Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: business.com
    Name: Default
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
```

Step	Description
2.	**Codecs** IP codec set 1 was used for the compliance test. Multiple codecs were listed in priority order to allow the codec used by a specific call to be negotiated during call establishment. The list includes the codecs the enterprise wishes to support within the normal trade-off of bandwidth versus voice quality. The example below shows the values used in the compliance test. It should be noted that when testing the use of each individual codec, only the codec under test was included in the list.

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

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n          2        20
 2: G.729A            n          2        20
 3:
 4:
 5:
 6:
 7:


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

Step	Description
3.	**Signaling Group** For the compliance test, signaling group 1 was used for the signaling group associated with the SIP trunk group between Communication Manager and SIP Enablement Services. Signaling group 1 was configured using the parameters highlighted below. All other fields were set as described in [3]. ▪ **Near-end Node Name**: ***procr*** This node name maps to the IP address of the Avaya S8300 Server. Node names are defined using the **change node-names ip** command. ▪ **Far-end Node Name**: ***SES*** This node name maps to the IP address of SIP Enablement Services. ▪ **Far-end Network Region**: ***1*** This defines the IP network region which contains SIP Enablement Services. ▪ **Far-end Domain**: ***business.com*** This domain is sent in the "To" header of SIP messages of calls using this signaling group. ▪ **Direct IP-IP Audio Connections**: ***y*** This field must be set to ***y*** to enable media shuffling on the SIP trunk.

```
display signaling-group 1
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
                        Transport Method: tls
  IMS Enabled? n





   Near-end Node Name: procr                 Far-end Node Name: SES
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
Far-end Domain: business.com

                                             Bypass If IP Threshold Exceeded? n


         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n                      Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 6
```

Step	Description

4.	**Trunk Group**
	For the compliance test, trunk group 1 was used for the SIP trunk group between Communication Manager and SIP Enablement Services. Trunk group 1 was configured using the parameters highlighted below. All other fields were set as described in [3].
	▪ **Signaling Group**: ***1*** This field is set to the signaling group shown in the previous step.
	▪ **Number of Members: *24*** This field represents the number of trunks in the SIP trunk group. It determines how many simultaneous SIP calls can be supported by the configuration. Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. Thus, a call from a SIP telephone to another SIP telephone will use two SIP trunks. A call between a non-SIP telephone and a SIP telephone will only use one trunk.
	<pre>display trunk-group 1 Page 1 of 21
	                                TRUNK GROUP
	
	Group Number: 1 Group Type: sip CDR Reports: y
	  Group Name: SES Trk Grp COR: 1 TN: 1 TAC: 101
	   Direction: two-way Outgoing Display? y
	 Dial Access? n Night Service:
	Queue Length: 0
	Service Type: tie Auth Code? n
	
	                                                    Signaling Group: 1
	                                                  Number of Members: 24</pre>

5.	**Trunk Group – continued**
	On **Page 3**:
	▪ Verify the **Numbering Format** field is set to ***public***. This field specifies the format of the calling party number sent to the far-end.
	▪ The default values may be retained for the other fields.
	<pre>display trunk-group 1 Page 3 of 21
	TRUNK FEATURES
	          ACA Assignment? n Measured: none
	                                                   Maintenance Tests? y
	
	                     Numbering Format: public
	                                                 UUI Treatment: service-provider
	
	                                                  Replace Restricted Numbers? n
	                                                 Replace Unavailable Numbers? n
	
	 Show ANSWERED BY on Display? y</pre>

Step	Description
6.	**Public Unknown Numbering** Public unknown numbering defines the calling party number to be sent to the far-end. An entry was created for the trunk group defined in **Step 4**. In the example shown below, all calls originating from a 5-digit extension beginning with 3 and routed across any trunk group (**Trk Grp(s)** setting is blank) including trunk group1 will be sent as a 5 digit calling number. This calling party number is sent to the far-end in the SIP "From" header.

```
display public-unknown-numbering 0                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                         Total
Ext Ext           Trk       CPN           CPN
Len Code          Grp(s)    Prefix        Len
                                                   Total Administered: 14
 5  3                                     5           Maximum Entries: 240
```

4.2. Configure SIP Trunks to Session Manager

To connect to Session Manager, 2 SIP trunk groups were configured on Communication Manager, one for sending outgoing calls from Communication Manger to Session Manager, the other for receiving incoming calls from Session Manager.

Step	Description
1.	**Node Names** Use the **change node-names ip** command to create a node name for the IP address of Session Manager. Enter a descriptive name in the **Name** column and the IP address assigned to Session Manager in the **IP address** column. The example below shows the values used in the compliance test at site 1. At site 2, since a direct SIP trunk needs to be established between the Communication Manager and the SIParater at the main enterprise site, the SIParator and its public side IP address should be configured in the **IP Node Names** form instead of the entry for Session Manager.

```
change node-names ip
                                  IP NODE NAMES
    Name              IP Address
ASMeast             10.75.5.24
SES                 10.75.5.6
default             0.0.0.0
myaudix             10.75.5.7
procr               10.75.5.2
```

Step	Description
2.	**Signaling Group (for outgoing calls)** For the compliance test, signaling group 27 was used for the SIP trunk group defined for sending outgoing calls to Session Manager (see **Step 3**). Signaling group 27 was configured using the same parameters as signaling group 1 in **Section 4.1, Step 3** with the exception of the **Far-end Node Name**. The **Far-end Node Name** field was set to the node name for Session Manager. At site 2, this signaling group was used for the trunk group connecting the Communication Manager to the SIParator at the main enterprise site. So the **Far-end Node Name** field should be set to the node name for the SIParator and the Far-end Domain field should be set to ***bigtime.com***.

```
display signaling-group 27
                                SIGNALING GROUP

 Group Number: 27             Group Type: sip
                        Transport Method: tcp
  IMS Enabled? n





   Near-end Node Name: procr                 Far-end Node Name: ASMeast
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 1
Far-end Domain: business.com

                                             Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n                     Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

Step	Description

3.	**Trunk Group (for outgoing calls)**
	For the compliance test, trunk group 27 was used for the SIP trunk group defined for connecting Communication Manager to Session Manager. Trunk group 27 was configured using the same parameters as trunk group 1 in **Section 4.1, Step 4** except that the ***Group Name*** field was named differently and the **Signaling Group** field was set to ***27***. This includes the settings on **Page 3** of the trunk group form (not shown). Similar changes should be made for this trunk group form at site 2.
	
	`display trunk-group 27                                          Page   1 of  21`
	`                                TRUNK GROUP`
	
	`Group Number: 27                  Group Type: sip           CDR Reports: y`
	`  Group Name: To ASMeast                   COR: 1        TN: 1        TAC: 127`
	`   Direction: two-way       Outgoing Display? n`
	` Dial Access? n                                 Night Service:`
	`Queue Length: 0`
	`Service Type: tie                  Auth Code? n`
	
	`                                                        Signaling Group: 27`
	`                                                      Number of Members: 24`

4.	**Signaling Group (for incoming calls)**
	For the compliance test, signaling group 26 was used for the SIP trunk group defined for receiving incoming calls from Session Manager (see **Step 5**). Signaling group 26 was configured using the same parameters as signaling group 27 in **Step 2** with the exception of the **Far-end Domain** set to blank.
	
	At site 2, this signaling group was used for the trunk group connecting the Communication Manager to the SIParator at the main enterprise site. So the **Far-end Node Name** field should be set to the node name for the SIParator.
	
	`display signaling-group 26`
	`                                SIGNALING GROUP`
	
	` Group Number: 26             Group Type: sip`
	`                        Transport Method: tcp`
	`  IMS Enabled? n`
	
	`   Near-end Node Name: procr                 Far-end Node Name: ASMeast`
	` Near-end Listen Port: 5060                Far-end Listen Port: 5060`
	`                                       Far-end Network Region: 1`
	`Far-end Domain:`
	
	`                                            Bypass If IP Threshold Exceeded? n`
	
	`         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y`
	`Session Establishment Timer(min): 3                    IP Audio Hairpinning? n`
	`         Enable Layer 3 Test? n                   Direct IP-IP Early Media? n`
	`H.323 Station Outgoing Direct Media? n          Alternate Route Timer(sec): 6`

Step	Description
5.	**Trunk Group (for incoming calls)** For the compliance test, trunk group 26 was used for the SIP trunk group defined for receiving incoming calls from Session Manager. Trunk group 26 was configured using the same parameters as trunk group 27 in **Step 3** except that the ***Group Name*** field was named differently and the **Signaling Group** field was set to ***26***. This includes the settings on **Page 3** of the trunk group form (not shown). Similar changes should be made for this trunk group form at site 2.

```
display trunk-group 26                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 26                  Group Type: sip           CDR Reports: y
  Group Name: From ASMeast               COR: 1        TN: 1        TAC: 126
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

                                                      Signaling Group: 26
                                                    Number of Members: 24
```

Step	Description
6.	**Automatic Alternate Routing** Automatic Alternate Routing (AAR) was used to route calls to Session Manager (for onward routing to the 2nd site through the SIParator). In the example shown, numbers that begin with 50 and are 5 digits long use route pattern 27. Route pattern 27 routes calls to the SIP trunk group defined for sending outgoing calls to Session Manager (see **Step 7**).

```
display aar analysis 5                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    3

          Dialed           Total     Route    Call   Node   ANI
          String          Min  Max  Pattern   Type   Num    Reqd
    50                     5    5     27      aar           n
    500                    5    5     27      aar           n
    501                    5    5     27      aar           n
```

Step	Description
7.	**Route Pattern** For the compliance test, route pattern 27 was used for calls destined for the 2nd site through Session Manager and the SIParator. Route pattern 27 was configured using the parameters highlighted below. ▪ **Pattern Name**: Any descriptive name. ▪ **Grp No**: ***27*** This field is set to the trunk group number defined in **Step 3**. ▪ **FRL**: ***0*** This field is the Facility Restriction Level of the trunk. It must be set to an appropriate level to allow authorized users to access the trunk. The level of 0 is the least restrictive.

```
display route-pattern 27                                        Page   1 of   3
                    Pattern Number: 27  Pattern Name: To ASMeast
                             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: 27   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
```

5. Configure Session Manager

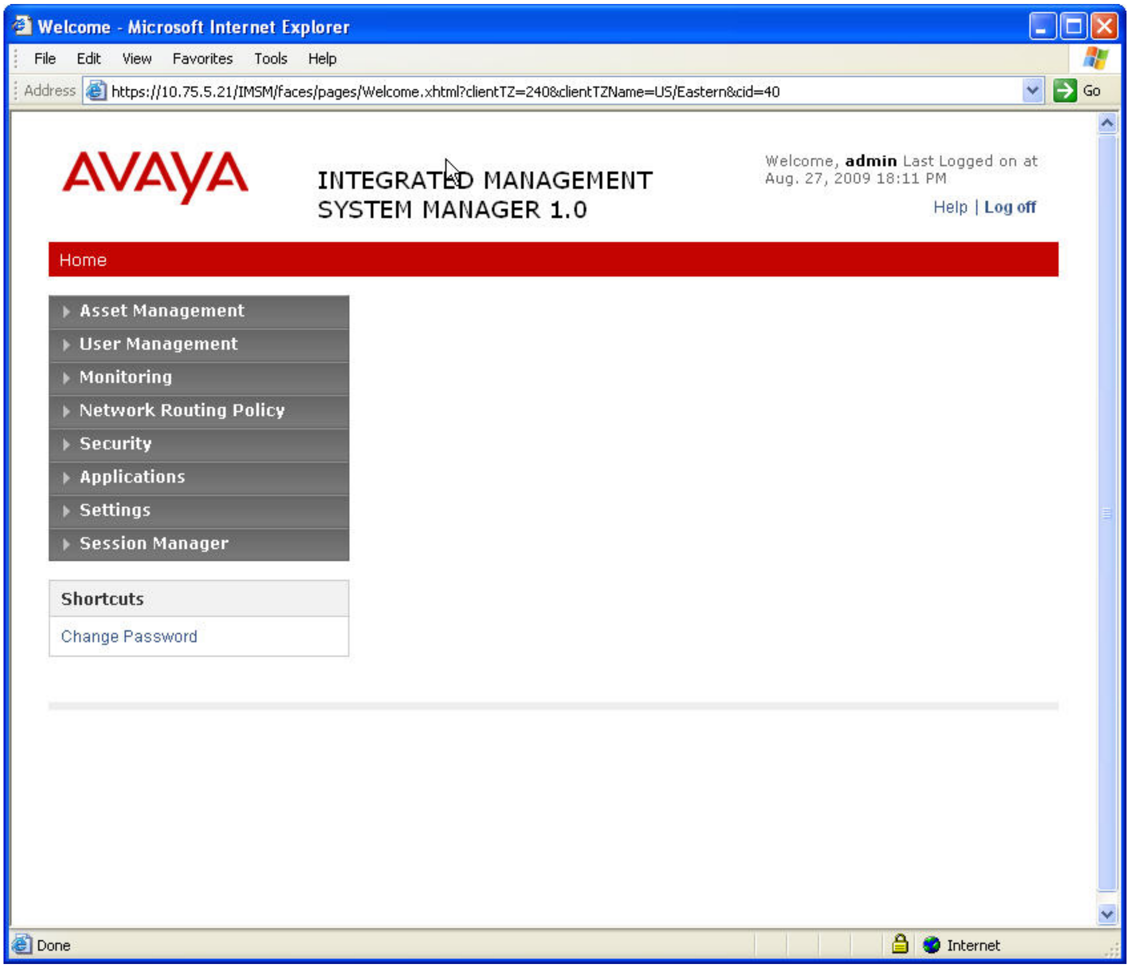
This section provides the procedures for configuring Session Manager as provisioned in the reference configuration. Session Manager is comprised of two functional components: the Session Manager server and the System Manager management server. All SIP call provisioning for Session Manager is performed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

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

The Session Manager server contains an SM-100 security module that provides the network interface for all inbound and outbound SIP signaling and media transport to all provisioned SIP entities. For the Session Manager used for the compliance test, the IP address assigned to the SM-100 interface is 10.75.5.24 as specified in **Figure 1**. The Session Manager server has a separate network interface used for connectivity to System Manager for managing/provisioning Session Manager. For the compliance test, the IP address assigned to the Session Manager management interface is 10.75.5.22. In the configuration for the compliance test, the SM-100 interface and the management interface were both connected to the same IP network. If desired, the SM-100 interface for real-time SIP traffic can be configured to use a different network than the management interface. For more information on Session Manager and System Manager, see [8] and [9].

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

- **SIP domain**
- Logical/physical **Locations** that can be occupied by SIP Entities
- **SIP Entities** corresponding to the SIP telephony systems (including Communication Manager and Session Border Controller) and Session Manager itself
- **Entity Links** which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- **Time Ranges** during which routing policies are active
- **Routing Policies** which control call routing between the SIP Entities
- **Adaptations** which specifies any digit conversions or domain modifications needed in SIP Request URI before routing the call to a SIP Entity
- **Dial Patterns** which govern to which SIP Entity a call is routed
- **Session Manager** corresponding to the Session Manager Servers managed by System Manager

1.	**Login** Access the Session Manager administration web interface by entering http://<*ip-addr*>/IMSM as the URL in an Internet browser, where <*ip-addr*> is the IP address of the System Manager server. Log in with the appropriate credentials. The main page for administrative interface is shown below. 

2. **Add SIP Domain**

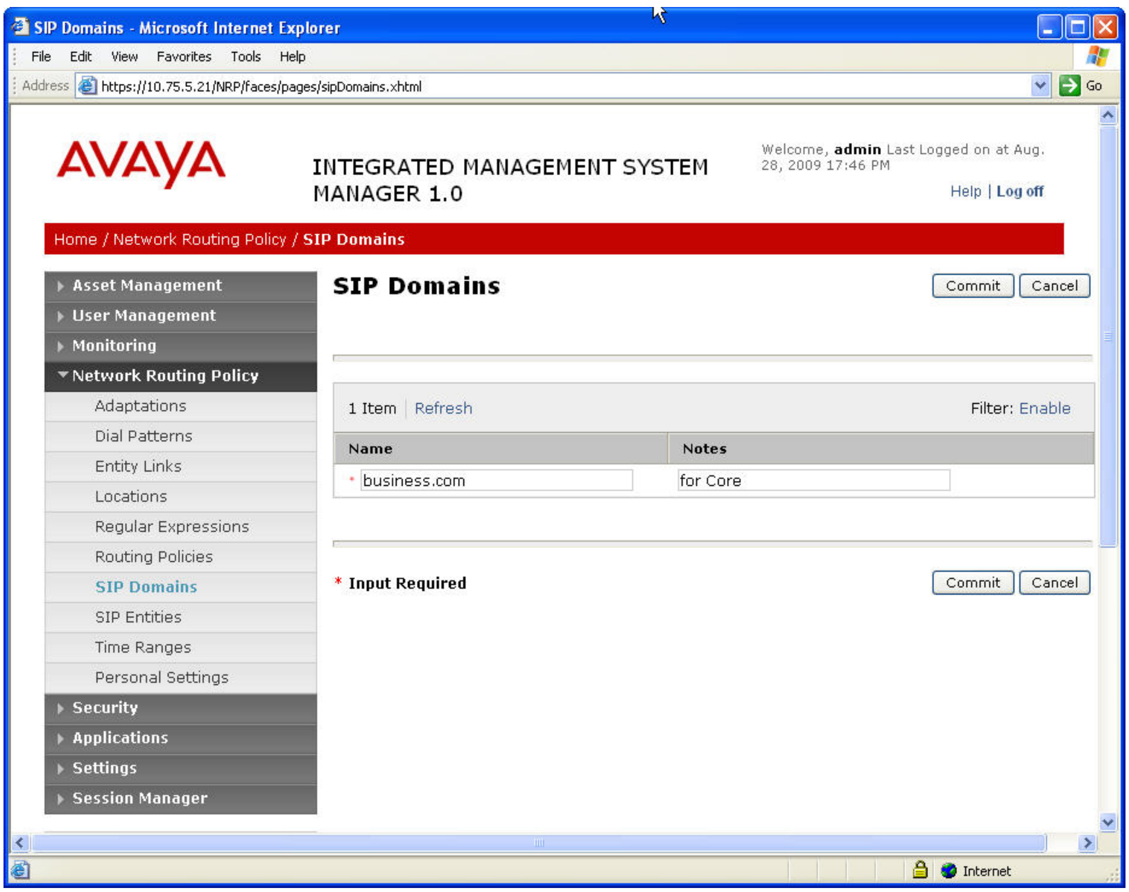
The **Network Routing Policy** sub menus contain all configuration tasks (except the last one) listed at the beginning of this section.

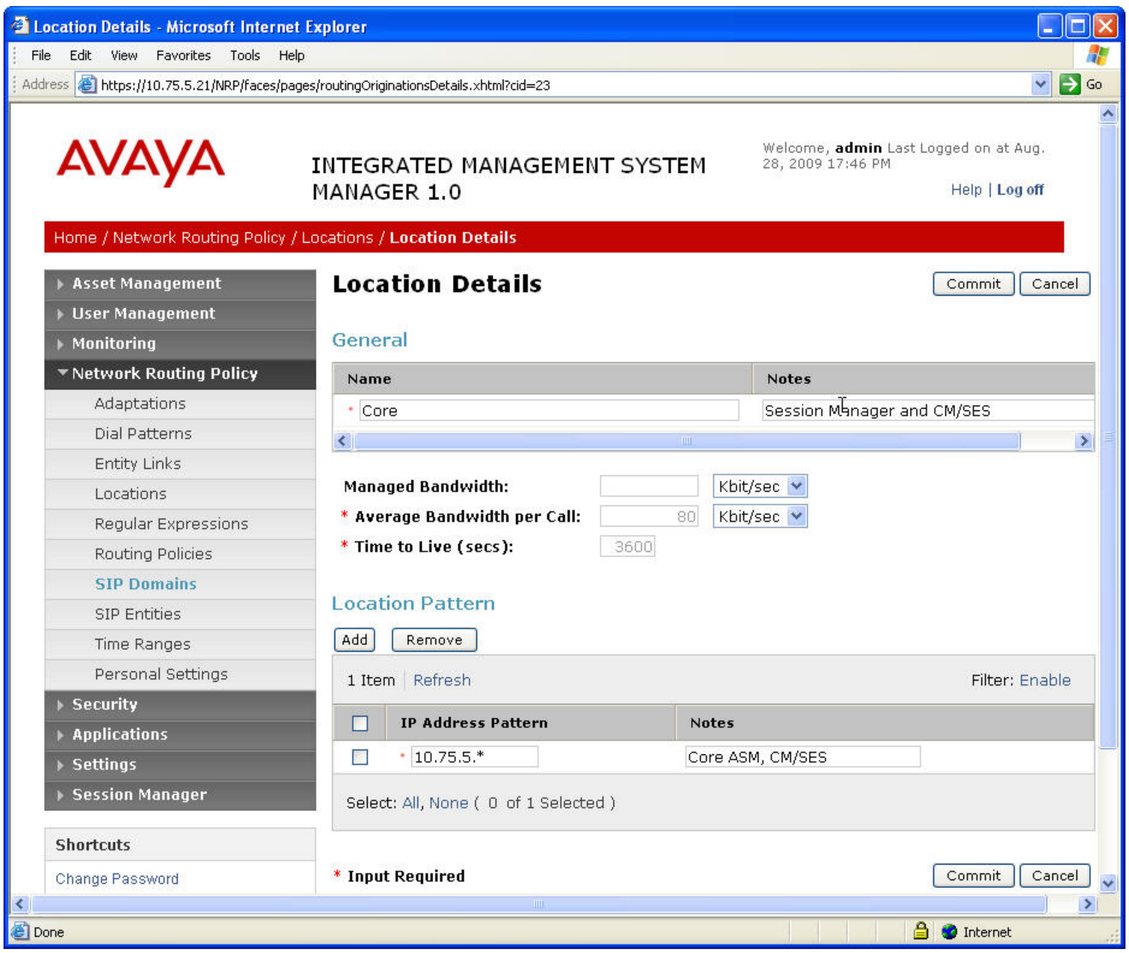
In the compliance test, only one SIP Domain was configured – all Session Manager SIP entities were located in the same authoritative domain.

Navigate to **Network Routing Policy→SIP Domains** to add the SIP domain with

- **Name**: ***business.com*** (as set in **Section 4.2, Step 2**)
- **Notes**: optional descriptive text

Click **Commit** to save the configuration.



3.	**Add Location** Locations identify logical and/or physical locations where SIP entities reside. In the compliance test, only one Location was configured – all Session Manager SIP entities were located in the same Location. Navigate to **Network Routing Policy→Locations** to add the Location. Under **General**: • **Name**: a descriptive name • **Notes**: optional descriptive text Under **Location Pattern**: • **IP Address Pattern**: ***10.75.5.**** • **Notes**: optional descriptive text Click **Commit** to save the configuration. 

4.

Add Adaptations

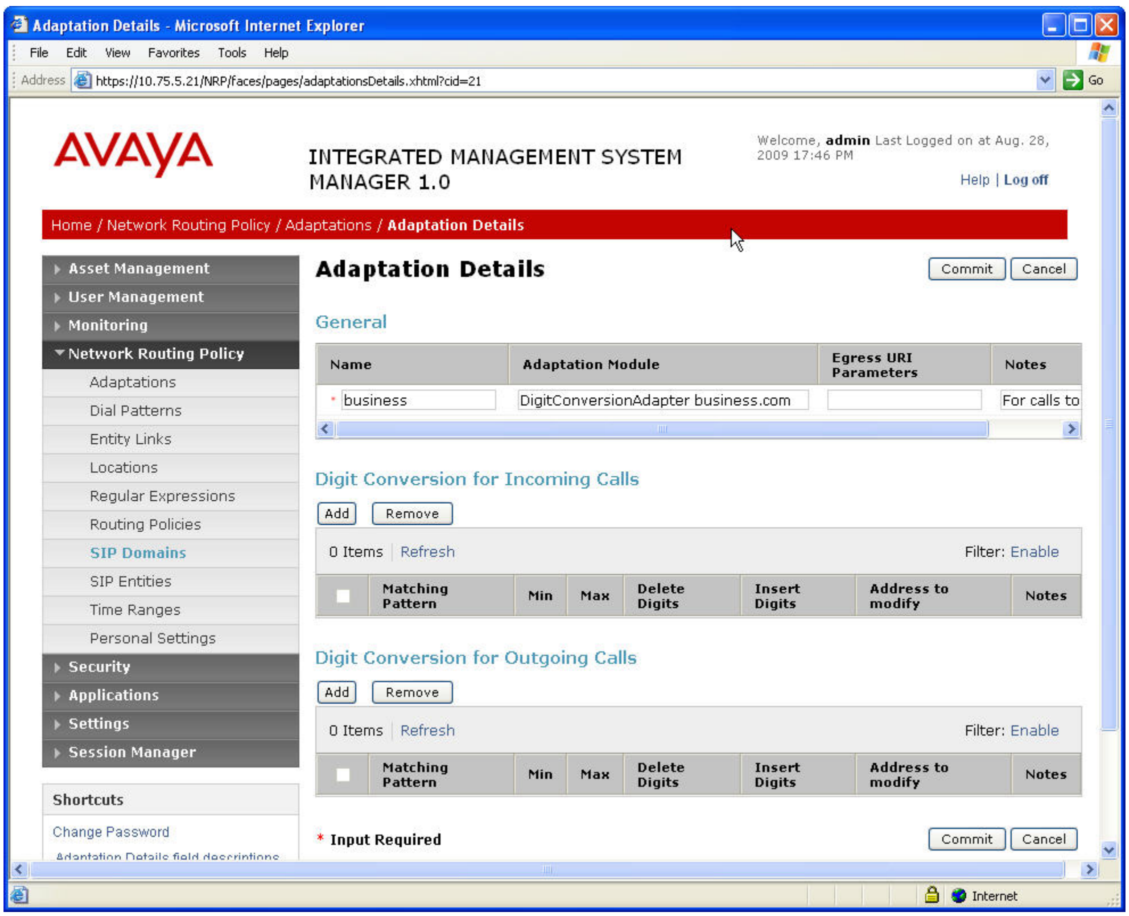
Session Manager provides for specialized code modules, called Adaptations, to process specific call processing requirements. In the compliance test, 2 Adaptations were used to update the domain as contained in the SIP Request-URI based on the SIP Entities to which this adaptation is defined. The screen below shows the configuration details of the Adaptation (when associated with the Communication Manager SIP Entity in **Step 7**) that will replace domain in the SIP Request-URI for all calls to Communication Manager to ***business.com***.

Navigate to **Network Routing Policy→Adaptations** to add Adaptation.

Under **General**:

- **Name**: a descriptive name
- **Adaptation Module**: enter ***DigiConversionAdapter business.com***
- **Notes**: optional descriptive text

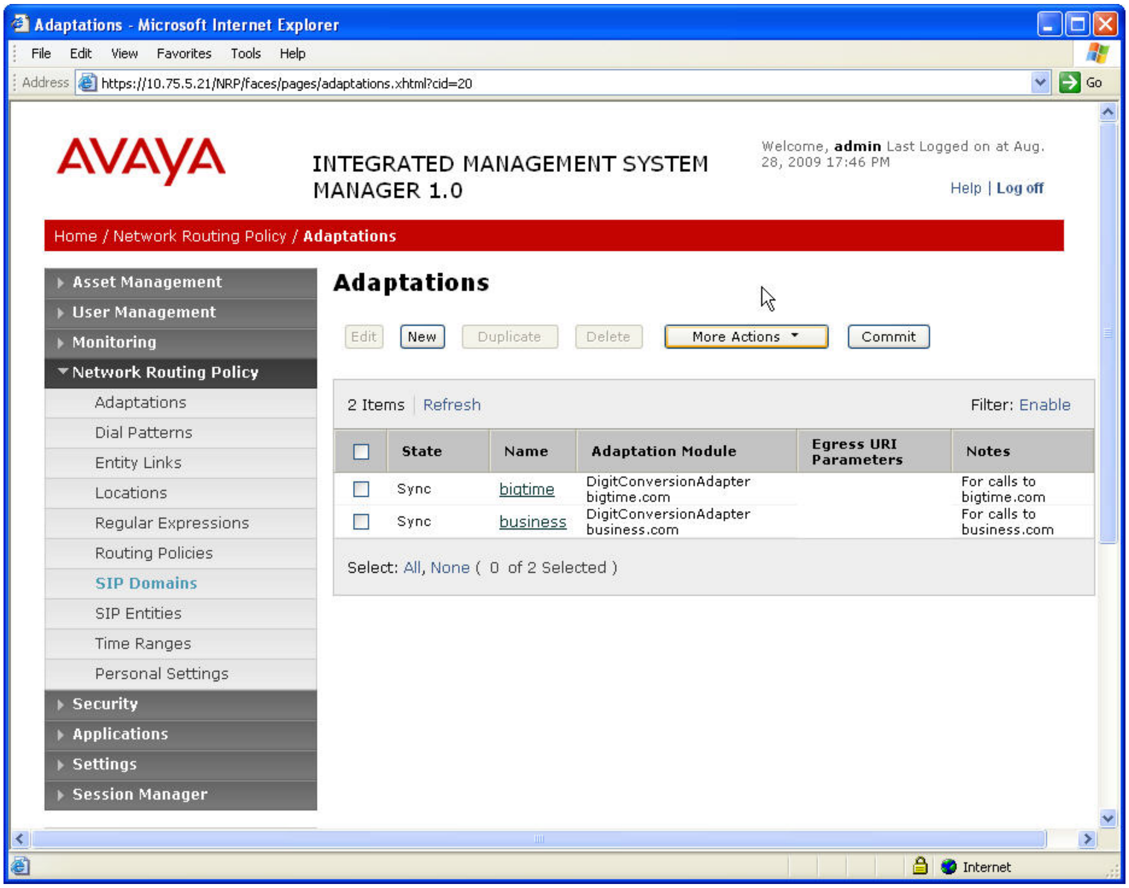
Click **Commit** to save the configuration.



5. **Add Adaptations (Continued)**

Add a second Adaptation that will replace domain in the SIP Request-URI for all calls to the Ingate SIParator SIP Entity (for onward routing to the 2nd site simulating a service provider service node) to ***bigtime.com***.

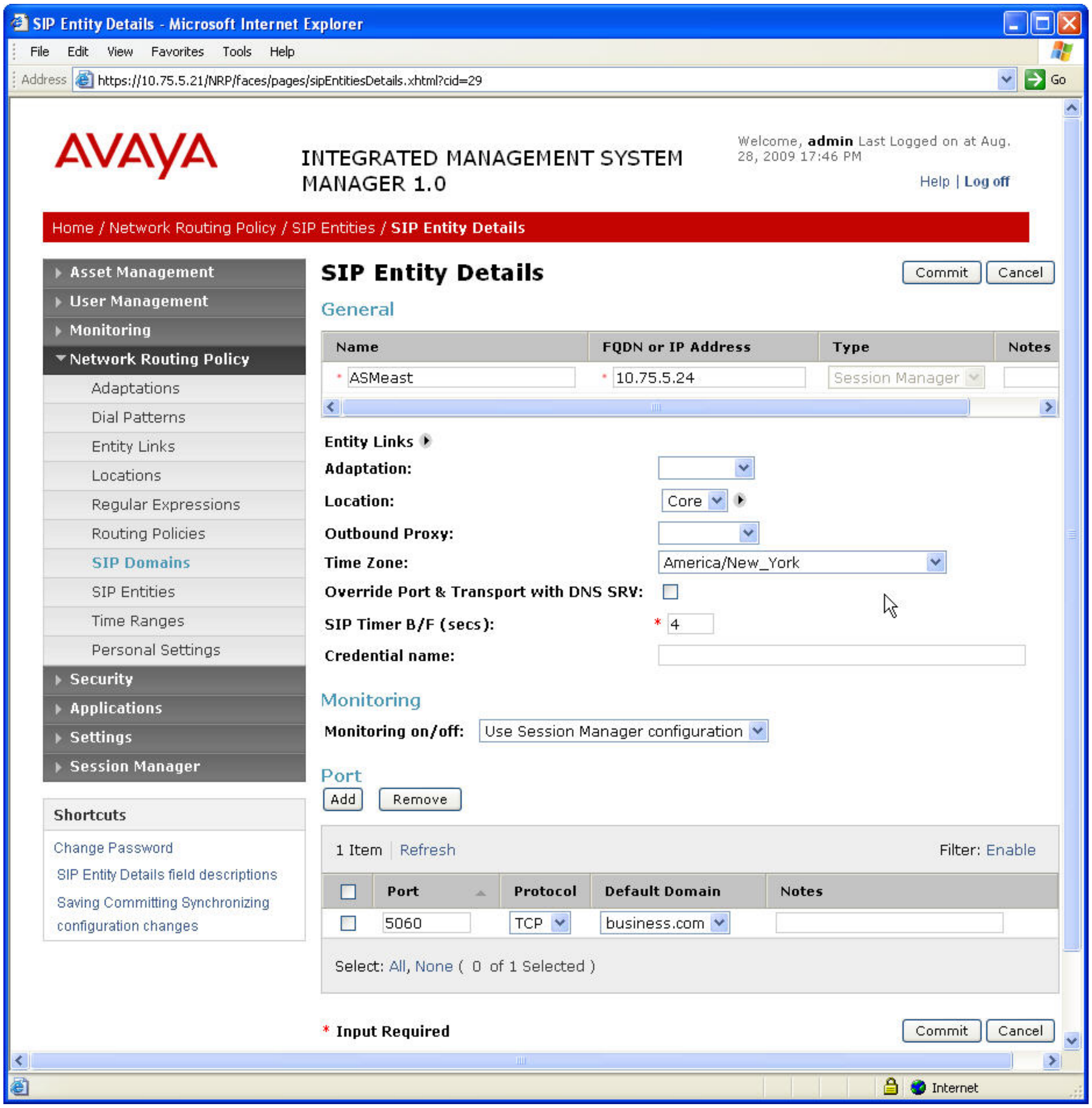
The Adaptations summary screen as shown below list the 2 Adaptations used in the compliance test:



6.	**Add SIP Entities – Session Manager** A SIP Entity must be added for Session Manager and for each SIP-based telephony system supported by it using SIP trunks. In the compliance test, a SIP Entity was added for the Session Manager itself, the Communications Manager, and the Ingate SIPrator. Navigate to **Network Routing Policy→SIP Entities** to add SIP Entities. The configuration details for the SIP Entity defined for Session Manager are as follows: Under **General**: • **Name**: a descriptive name • **FQDN or IP Address**: ***10.75.5.24*** as specified in **Figure 1**. This is the IP address assigned to the SM-100 security module installed in the Session Manager. • **Type**: select ***Session Manager*** • **Adaptation**: leave blank • **Location:** select the Location created in **Step 3** • **Time Zone:** select the proper time zone for this installation Under **Port**, click **Add**, then edit the fields in the resulting new row as shown below: • **Port**: ***5060***. This is the port number on which the system listens for SIP requests. • **Protocol**: ***TCP***. The TCP transport protocol was used in the compliance test to send SIP requests. • **Default Domain**: select the SIP Domain created in **Step 2**. Default settings can be used for the remaining fields. Click **Commit** to save the SIP Entity definition.

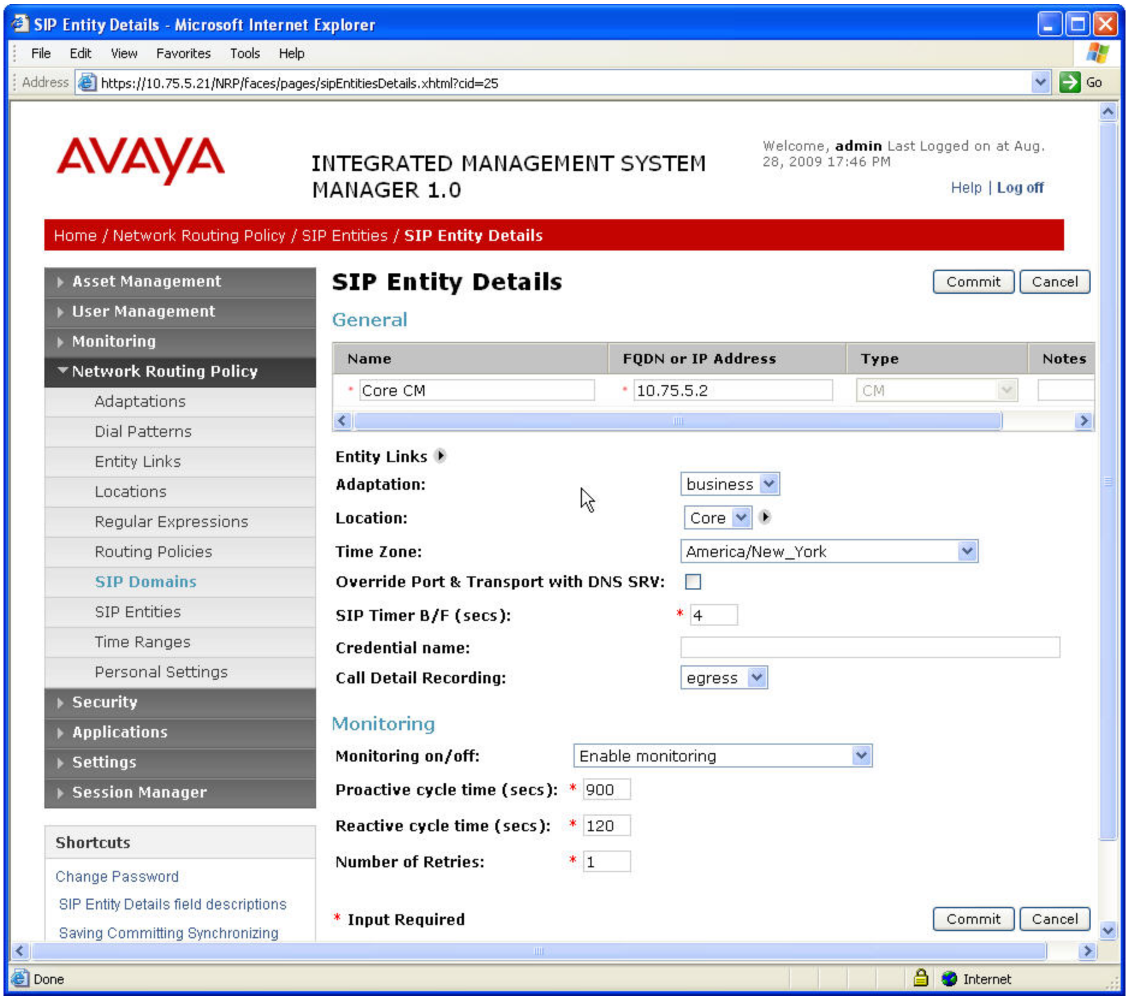
6. **Add SIP Entities – Session Manager (Continued)**

The screen below shows the SIP Entity configuration details for the Session Manager.



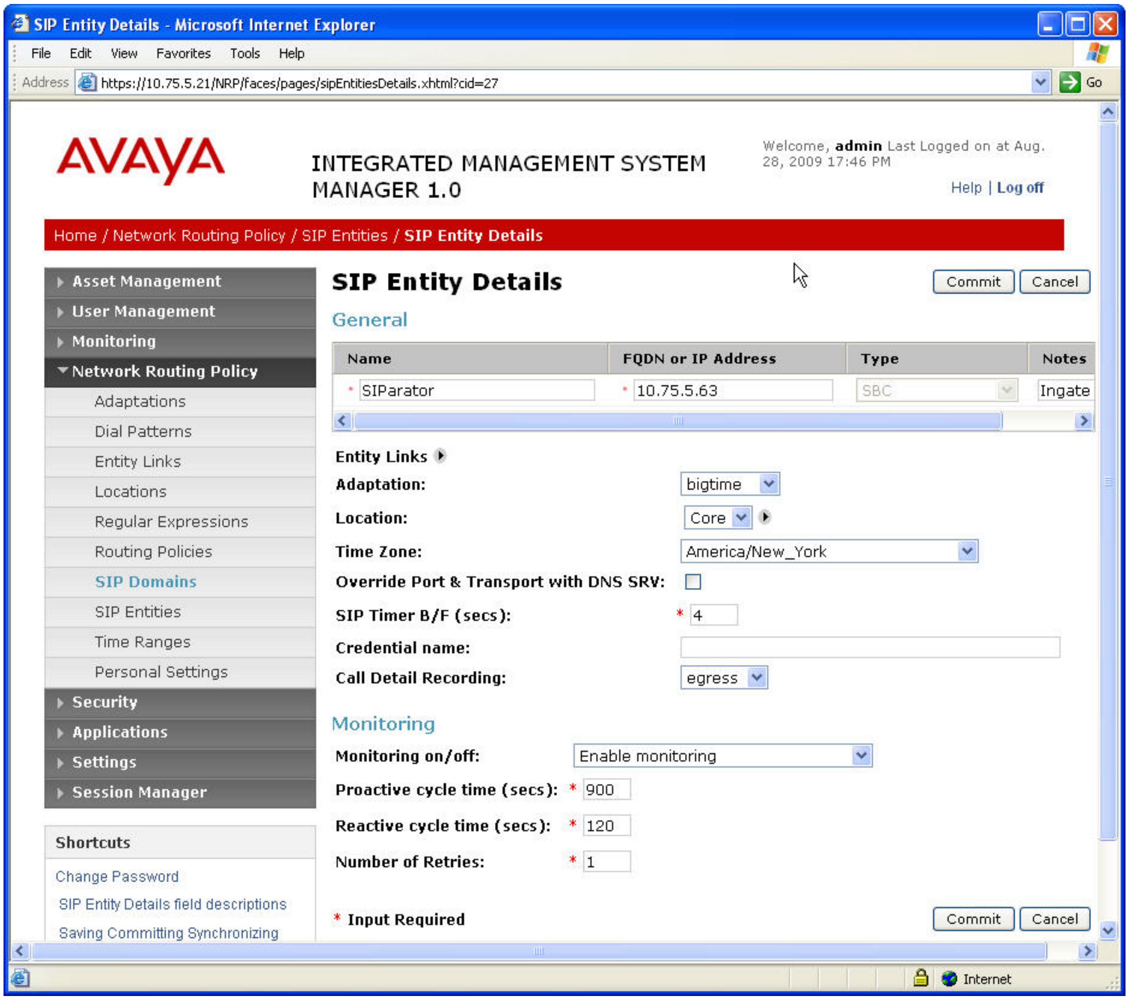
7. **Add SIP Entities – Communication Manager**

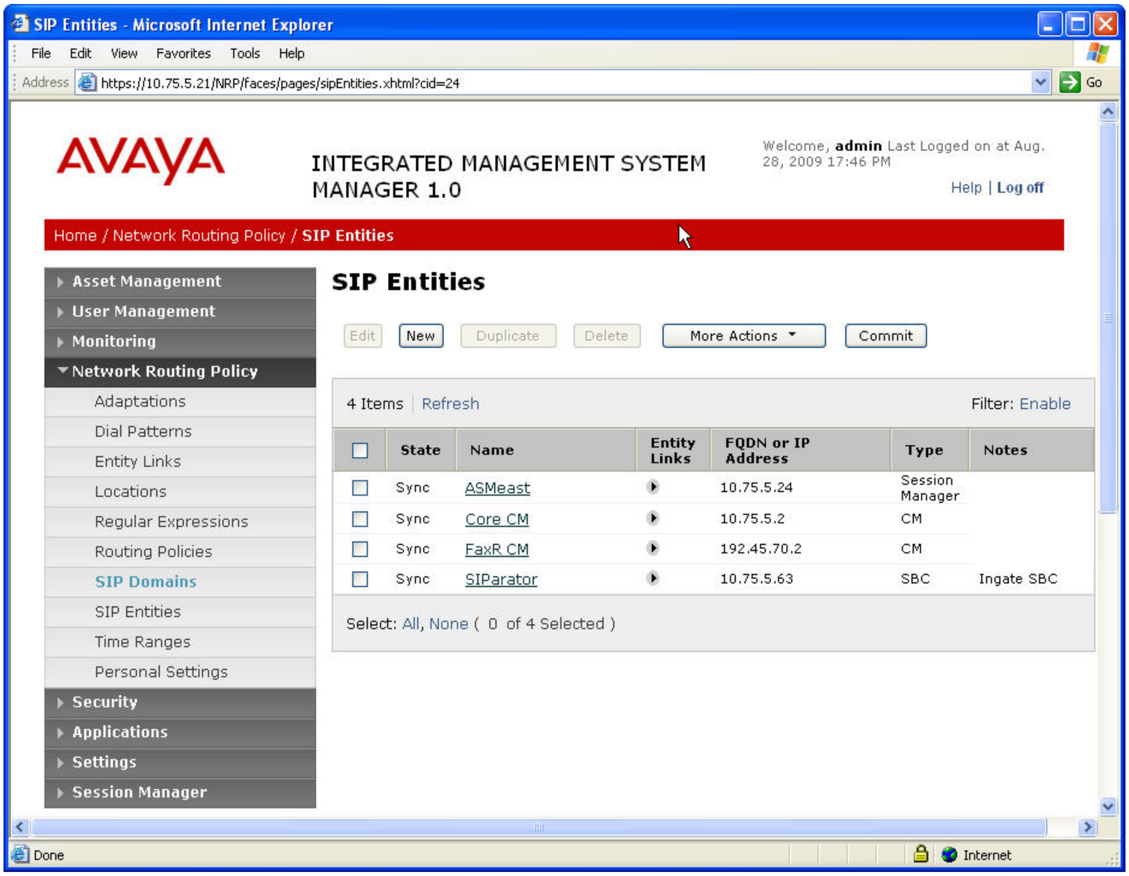
The screen below shows the SIP Entity configuration details for the Communication Manager. Note the ***CM*** selection for **Type** and the ***business*** **Adaptation** selection created in **Step 4**.



8. **Add SIP Entities – Ingate SIParator**

The screen below shows the SIP Entity configuration details for the Ingate SIParator. Note the ***SBC*** selection for **Type** and the ***bigtime*** **Adaptation** selection created in **Step 5**.



9.	**SIP Entities Summary List** The screen below shows the SIP Entities summary list displayed after the 3 SIP Entities have been added in **Steps 6**, **7** and **8**. Note that the SIP Entity named **FaxR CM** was configured for other purposes; it was not used in the compliance test. 

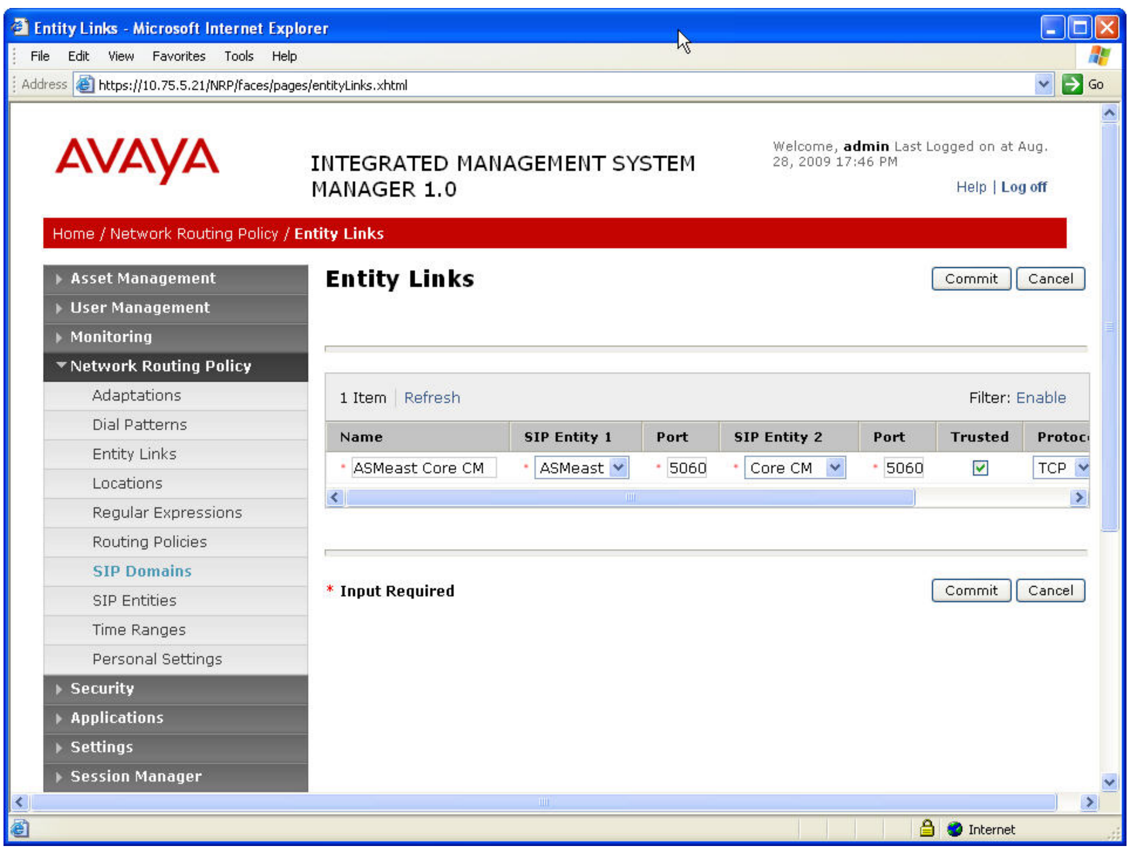
10. **Add Entity Links**

A SIP trunk between Session Manager and a telephony system is described by an Entity link. In the compliance test 2 Entity Links were created: one between Session Manager and Communication Manger; the other between Session Manager and Ingate SIParator.

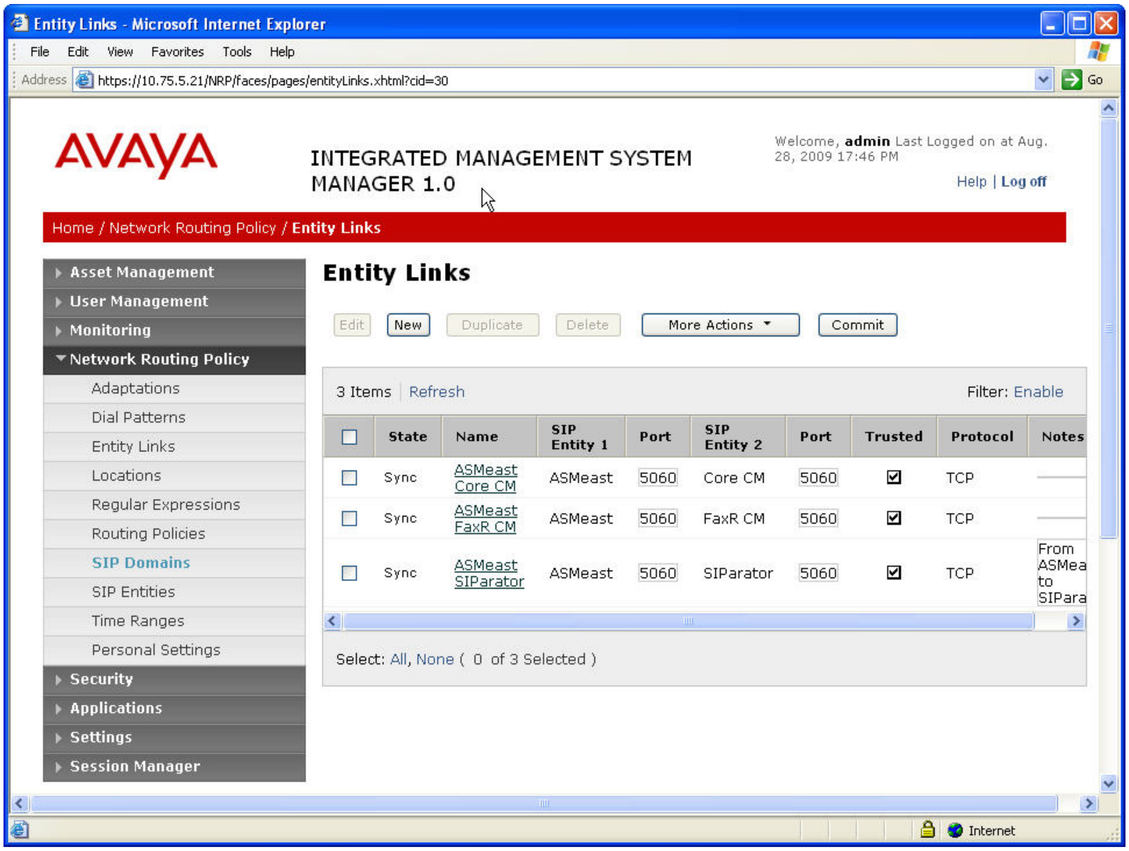
Navigate to **Network Routing Policy→Entity Links** to add a new Entity Link. The screen below shows the configuration details for the Entity Link connecting Session Manager to Communication Manager.

- **Name**: a descriptive name
- **SIP Entity 1**: select the Session Manager SIP Entity created in **Step 6**.
- **Port**: ***5060***. This is the port number to which the other system sends SIP requests.
- **SIP Entity 2**: select the Communication Manager SIP Entity created in **Step 7**.
- **Port**: ***5060***. This is the port number on which the other system receives SIP requests.
- **Trusted**: check this box
- **Protocol**: select ***TCP*** as the transport protocol.
- **Notes**: optional descriptive text

Click **Commit** to save the configuration.



11.	**Add Entity Links (Continued)** The Entity Link for connecting Session Manager to Ingate SIParator was similarly defined. The screen below shows the SIP Entity Links summary list displayed after the 2 SIP Entity Links have been configured. Note that the SIP Entity Link named **ASMeast FaxR CM** was configured for other purposes; it was not used in the compliance test.



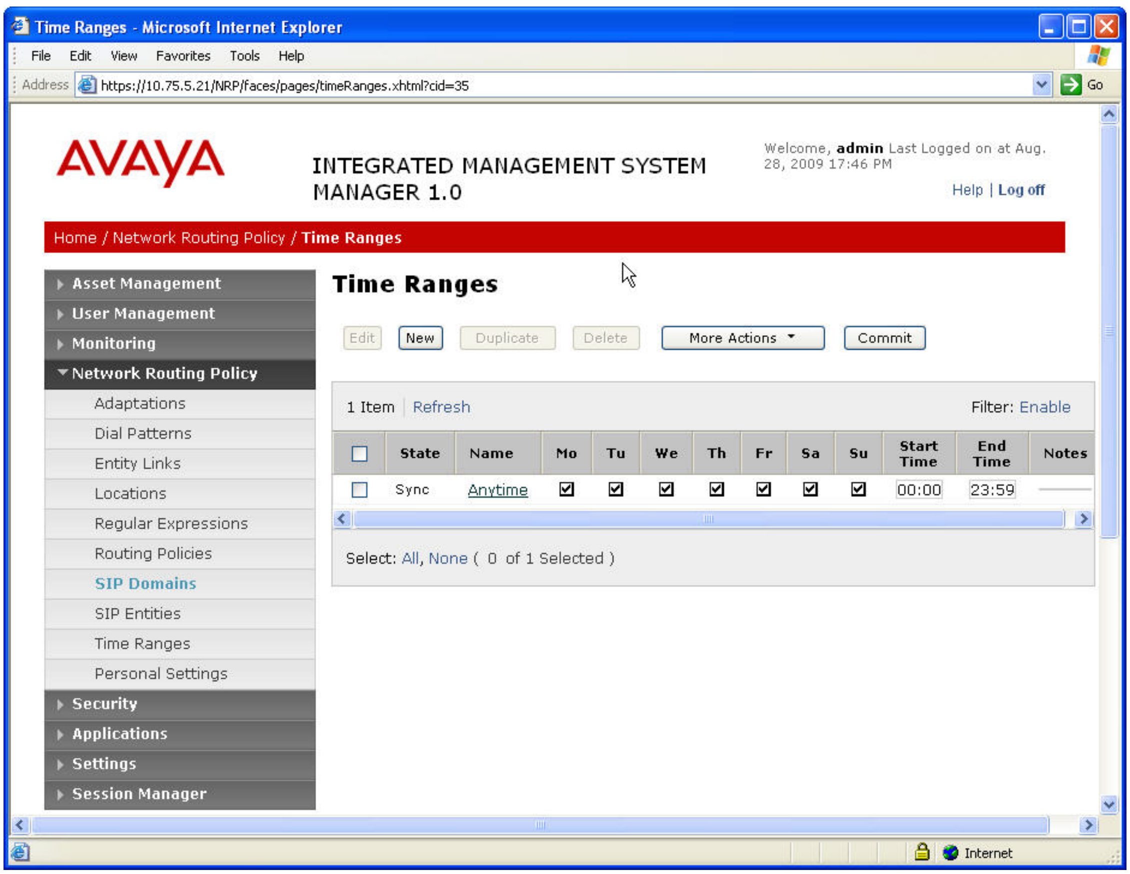
12. **Add Time Ranges**

Before adding routing policies (configured in next step), time ranges must be defined during which the policies will be active. For the compliance test, one Time Range was defined that would allow routing to occur at anytime.

Navigate to **Network Routing Policy→Time Ranges** to add a new Time Range:

- **Name**: a descriptive name
- **Mo** through **Su**: check the box under each of these headings
- **Start Time**: enter ***00:00***
- **End Time**: enter ***23:59***

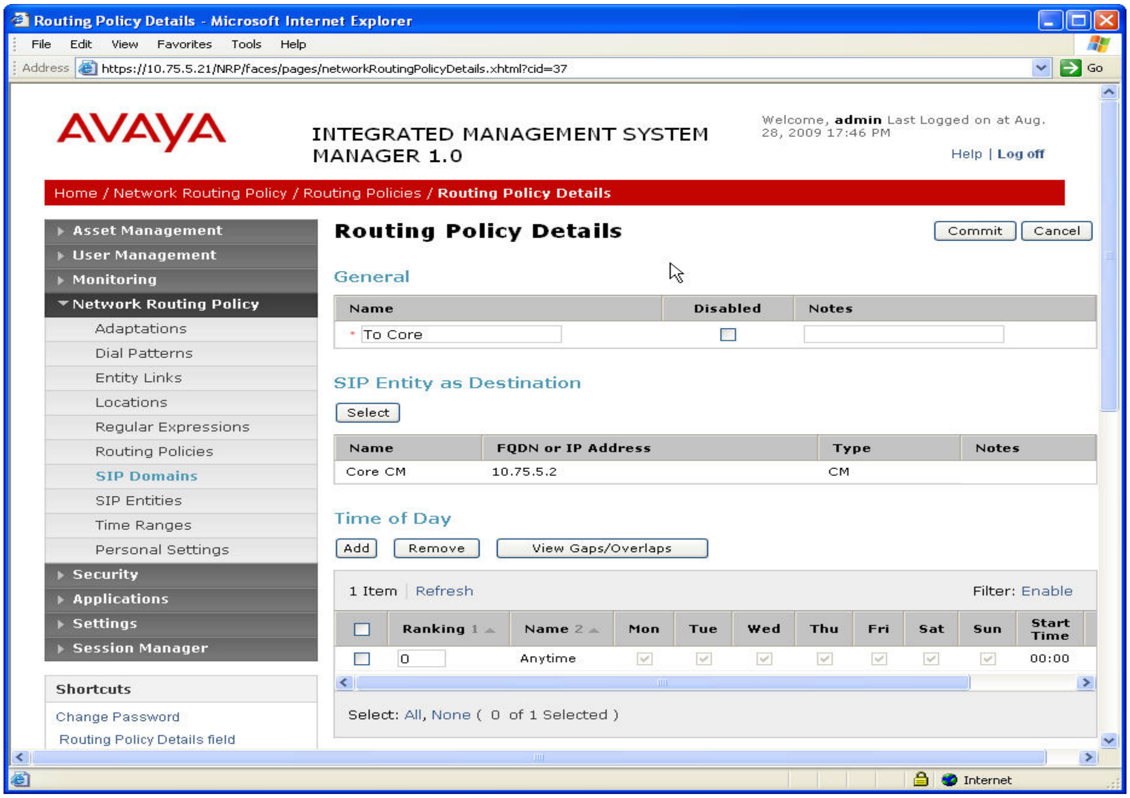
Click **Commit** to save this time range.

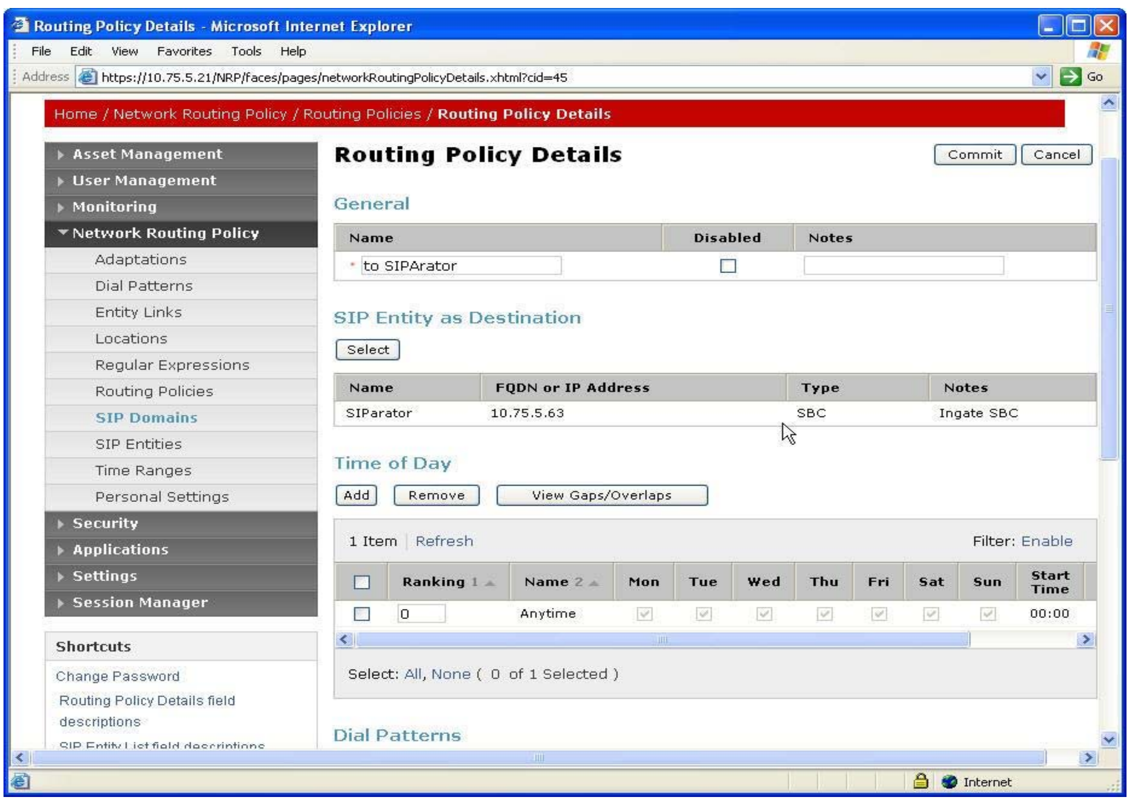


13.	**Add Routing Policies** Routing policies describe the conditions under which calls will be routed to the SIP Entities connected to the Session Manager. For the compliance test, 2 routing policies were added – one for routing calls to Communication Manager, the other for routing calls to Ingate SIParator. Navigate to **Network Routing Policy→Routing Policies** to add a new Routing Policy. Under **General**: • **Name**: a descriptive name • **Notes**: optional descriptive text Under **SIP Entity as Destination** Click **Select** to select the appropriate SIP Entity to which the routing policy applies. Under **Time of Day** Click **Add** to select the Time Range configured in **Step 12**. Default settings can be used for the remaining fields. Click **Commit** to save the configuration.

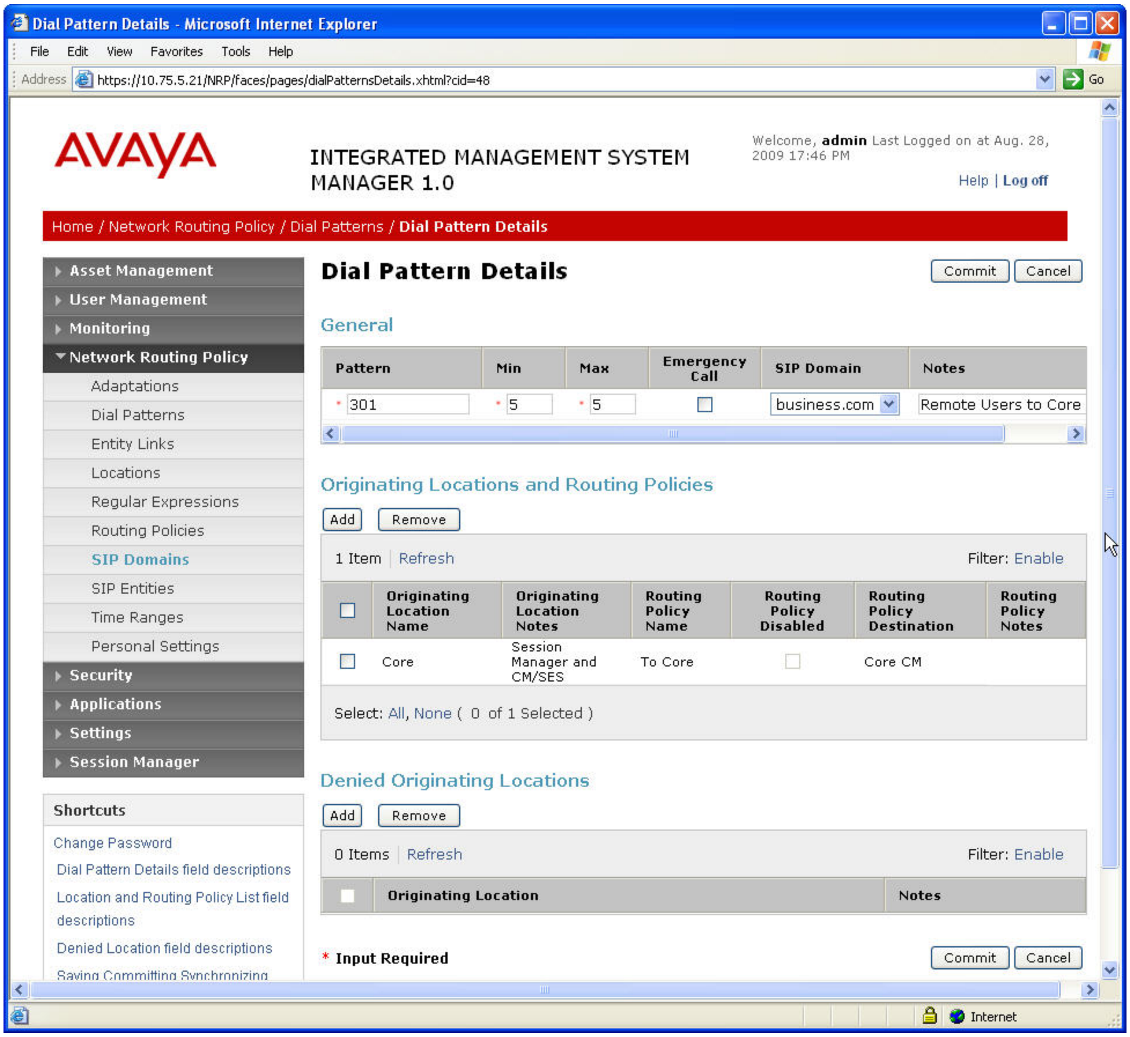
13. **Add Routing Policies (Continued)**

The screens below show the configuration details for the 2 Routing Policies defined for the compliance test.





14.	**Add Dial Patterns** Dial Patterns define digit strings to be matched against dialed numbers for directing calls to the appropriate SIP Entities. In the compliance test, 5-digit extensions beginning with “301” resided on Communication Manager in the main enterprise site; and 5-digit extensions beginning with “50” should be routed to Ingate SIPrator for onward routing to the 2nd site. Therefore 2 Dial Patterns were created accordingly. Navigate to **Network Routing Policy→Dial Patterns** to add a new Dial Pattern. Under **General**: • **Pattern**: dialed number or prefix • **Min**: minimum length of dialed number • **Max**: maximum length of dialed number • **SIP Domain**: select the SIP Domain created in Step 2 • **Notes**: optional descriptive text Under **Originating Locations and Routing Policies** Click **Add** to select the appropriate originating Location and Routing Policy from the list. Under **Time of Day** Click **Add** to select the time range configured in **Step 12**. Default settings can be used for the remaining fields. Click **Commit** to save the configuration.

14.	**Add Dial Patterns (Continued)** The screen below shows the configuration details for the Dialed Pattern defined for matching dialed numbers beginning with “301” destined for the main enterprise site. The Dialed Pattern defined for matching dialed numbers beginning with “50” destined for the Ingate SIParator (for onward routing to the 2nd site simulating a service provider service node) is similarly defined (not shown) with ***50*** specified for **Pattern** and the **to SIParator** selection for **Routing Policy Name** as defined in **Step 13**. 

15. **Add Session Manager**

To complete the configuration, adding the Session Manager provided the linkage between System Manager and Session Manager. This configuration procedure should have already been properly executed but is included here for reference and completeness.

Navigate to **Session Manager→Session Manager Administration** to add a new Session Manager:

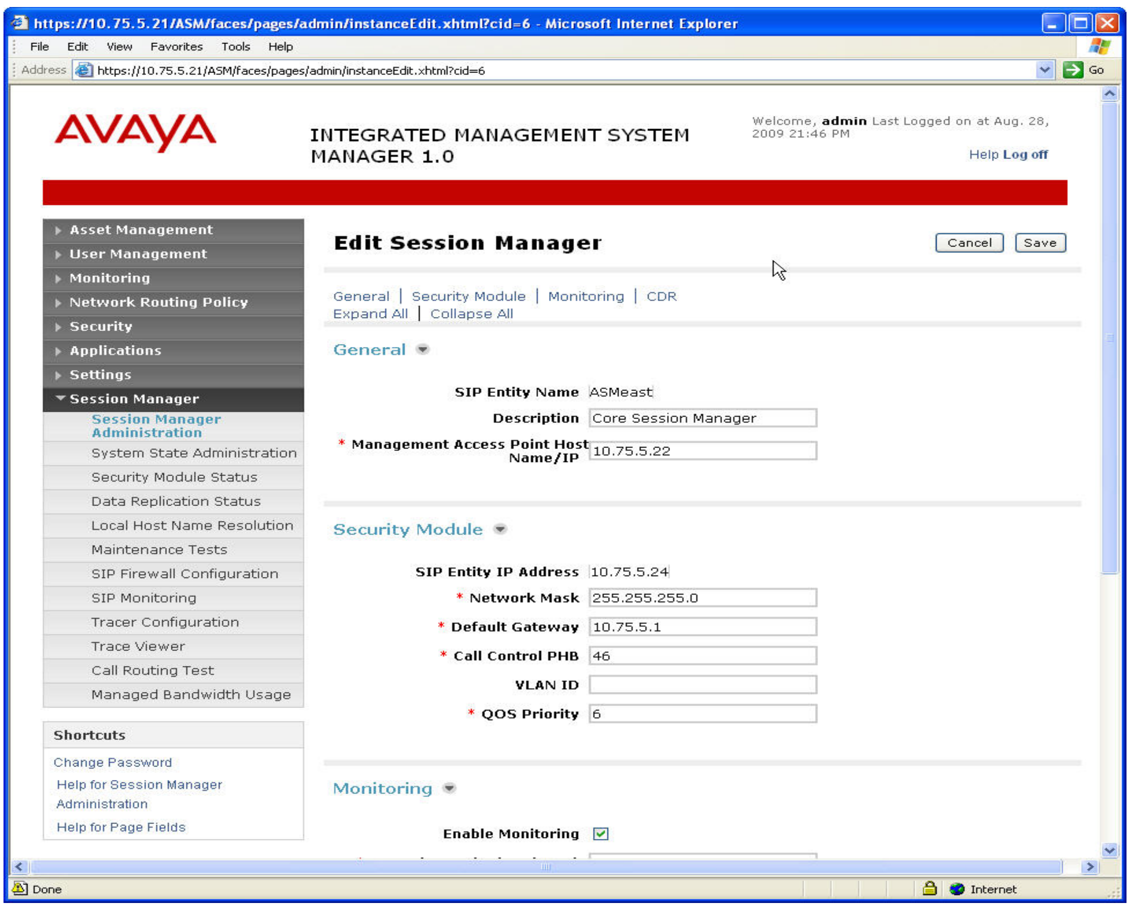
Under **Identity**:

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

Under Security Module:

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

Accept default settings for the remaining fields. Click **Save** to add this Session Manager..



6. Configure Avaya SIP Telephones

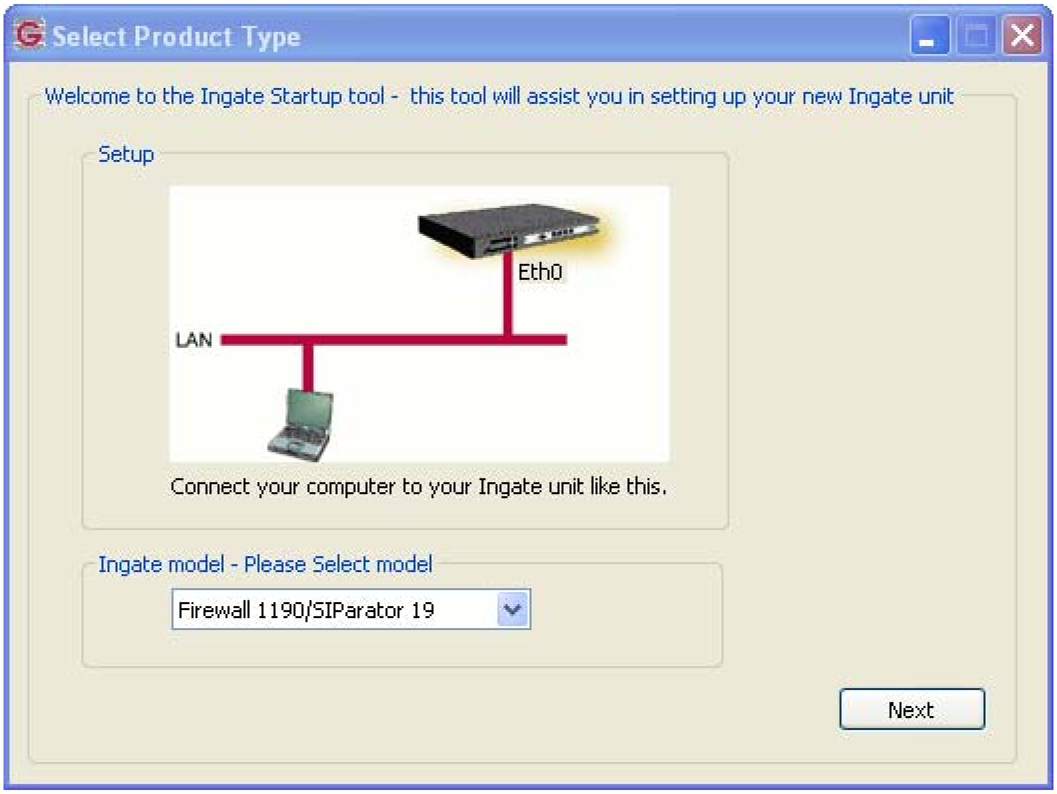
The SIP telephones at each site will use the local SIP Enablement Services as the call server. The table below shows an example of the SIP telephone network settings for each site.

	Main Site	**2nd Site**
Extension	30102	50022
IP Address	10.75.5.162	192.45.70.21
Subnet Mask	255.255.255.0	255.255.255.0
Router	10.75.5.1	192.45.70.1
File Server	10.75.10.100	192.45.70.100
DNS Server	0.0.0.0	0.0.0.0
SIP Domain	business.com	bigtime.com
Call Server or SIP Proxy Server	10.75.5.6	192.45.70.7

7. Configure the Ingate SIParator

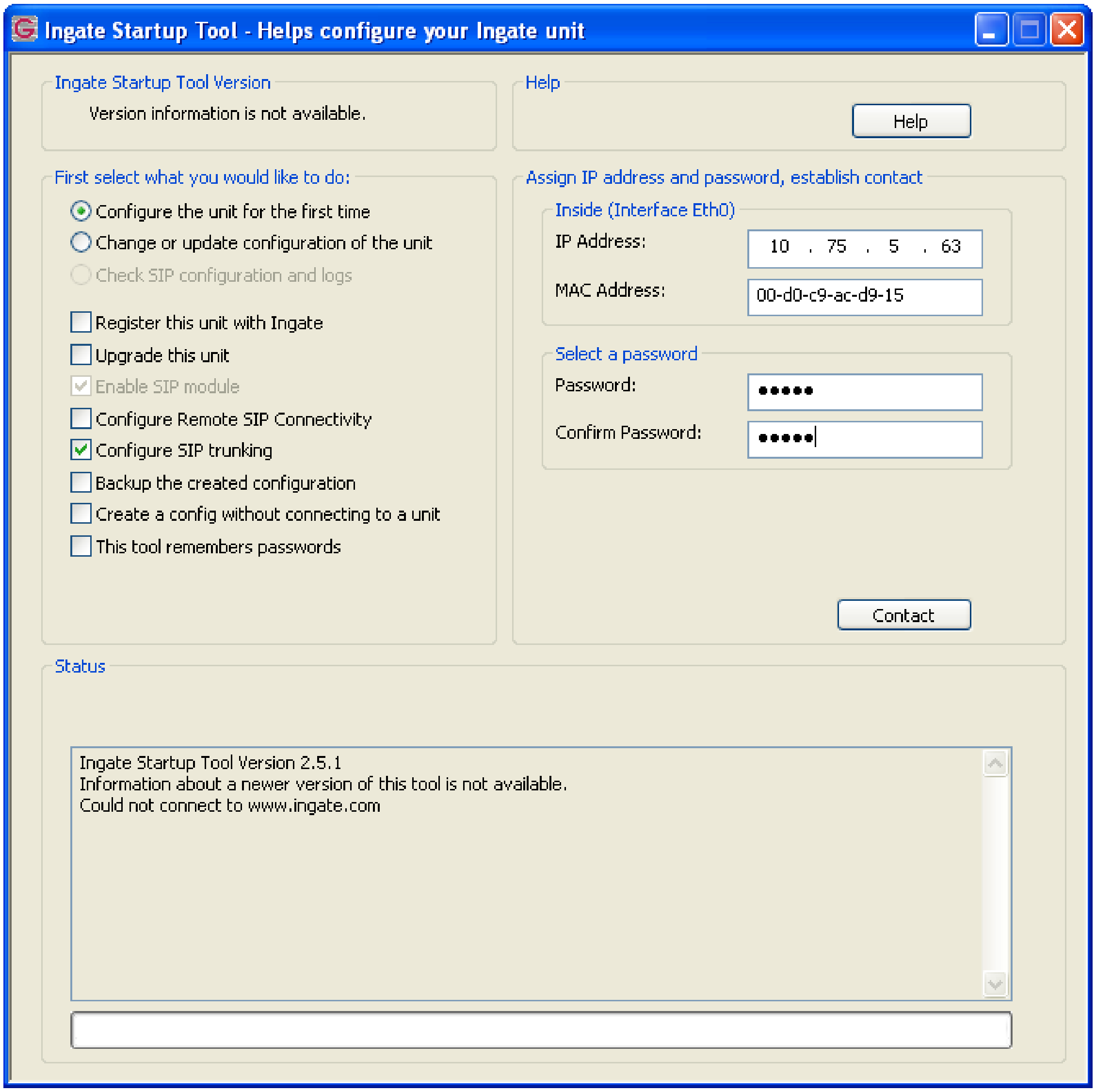
The Ingate SIParator is configured initially with the Ingate Startup Tool. Based on the provided input, the Startup Tool will create an initial configuration that can be uploaded to the SIParator. The results of this configuration can then be viewed or expanded using the SIParator web interface. To access the web interface, enter the IP address of the SIParator as the destination address in a web browser. When prompted for login credentials, enter an appropriate user name and password.

Step	Description
1.	**Launch Startup Tool** The Ingate Startup Tool is a windows application which is launched from the Windows Start Menu by navigating to **Start→All Programs→Shortcut to StartupTool.exe**.

Step	Description
2.	**Select Product Type** The initial Ingate Startup Tool screen is shown below. Verify the PC is running on the same LAN subnet as the SIParator as shown in the diagram. This is necessary in order to assign the initial IP address to the SIParator from the Startup Tool. Select the SIParator model from the **Ingate model** drop-down menu. Click the **Next** button. 

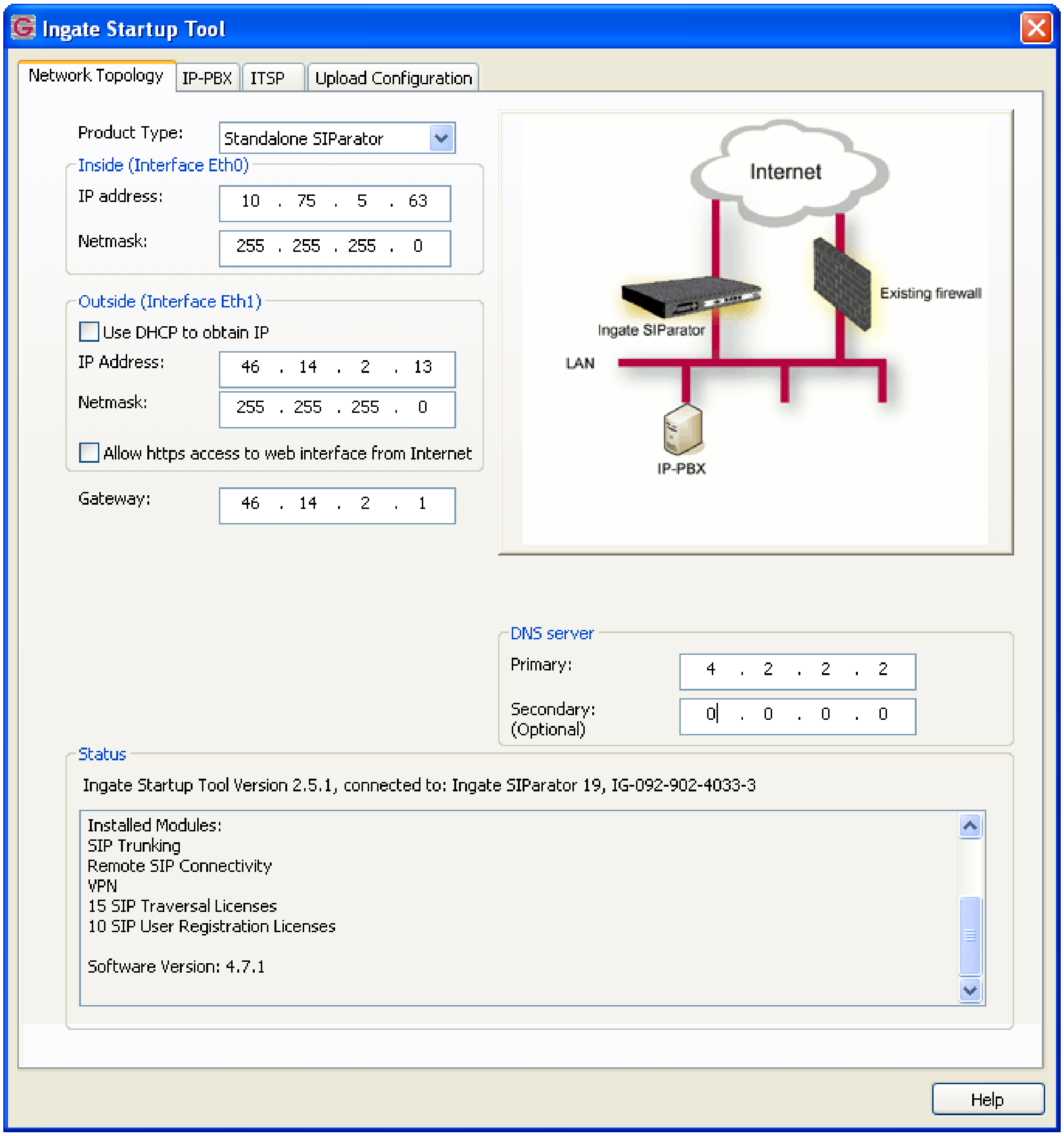
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>

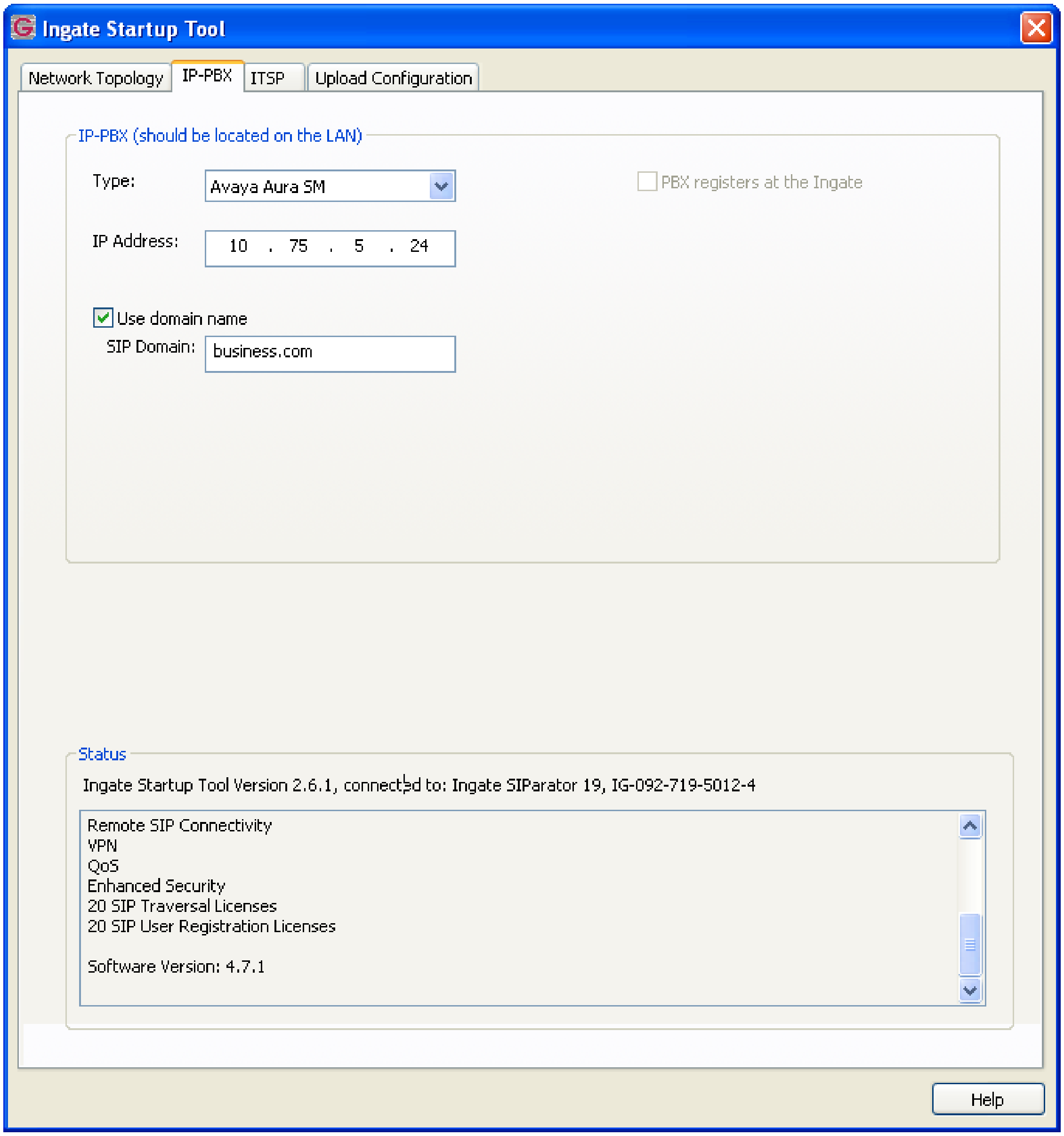
Select Configuration Options and Assign Private IP
Select options for **Configure the unit for the first time** and **Configure SIP trunking**. Enter the inside IP address, MAC address and a password. Click the **Contact** button to establish a connection to the SIParator. For future updates, click the option - **Change or update configuration of the unit**

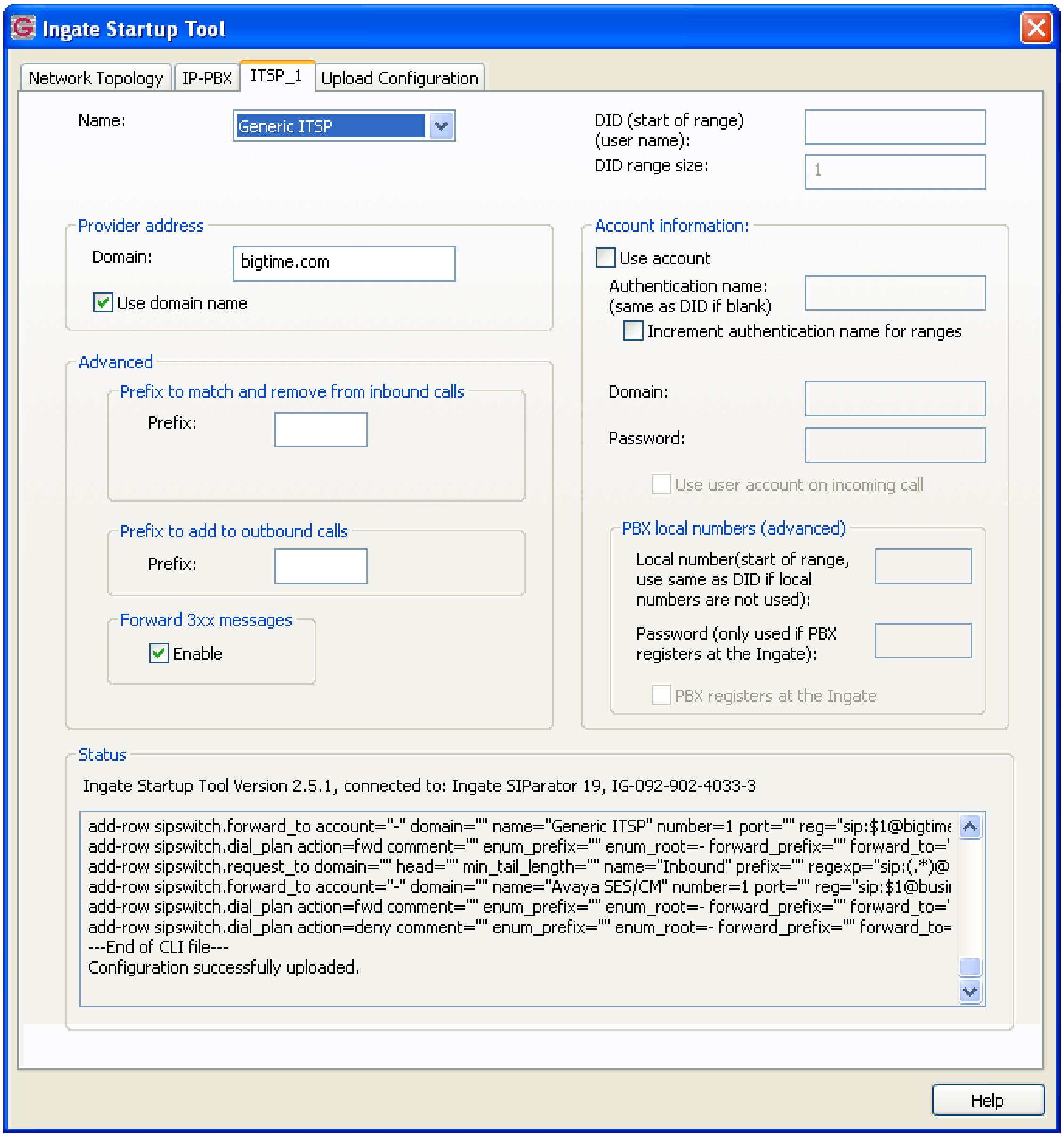


</td></tr>
</table>

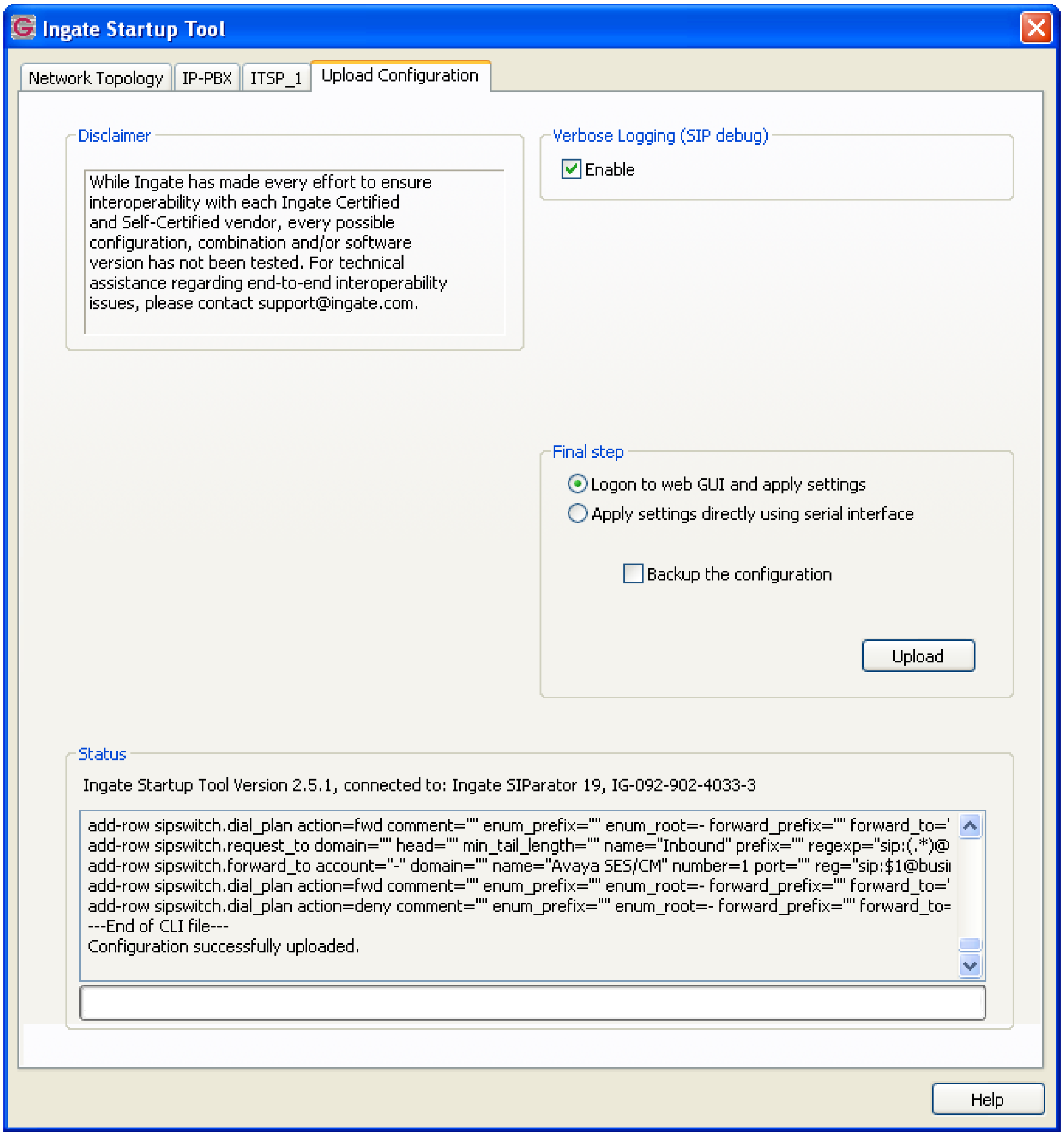
Step	Description
4.	**Network Topology** After connecting to the SIParator, the following page appears. Select the **Network Topology** tab. Select ***Standalone SIParator*** from the **Product Type** drop-down menu. Enter an IP address and subnet mask for both the inside and outside interfaces as shown in **Figure 1**. The **Gateway** field is set to the IP address of the default gateway on the public side of the SIParator. A DNS server was not used for the compliance test.

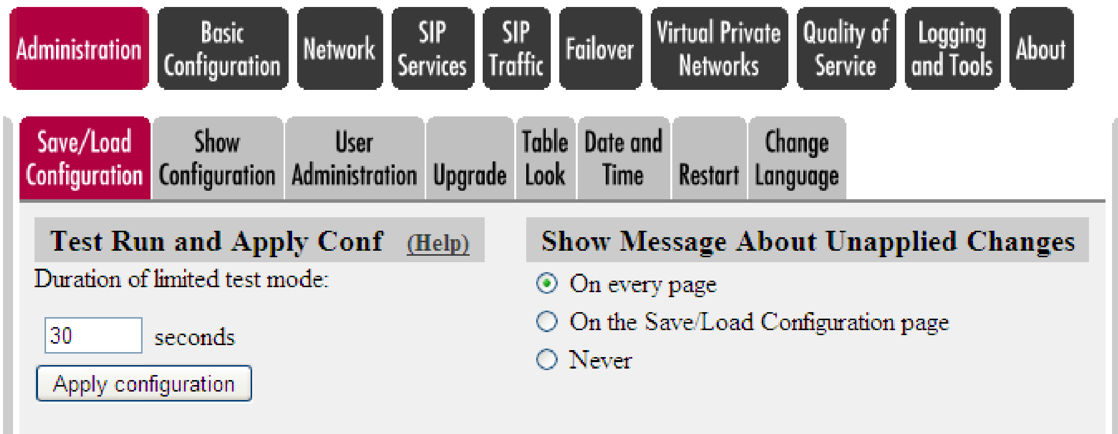


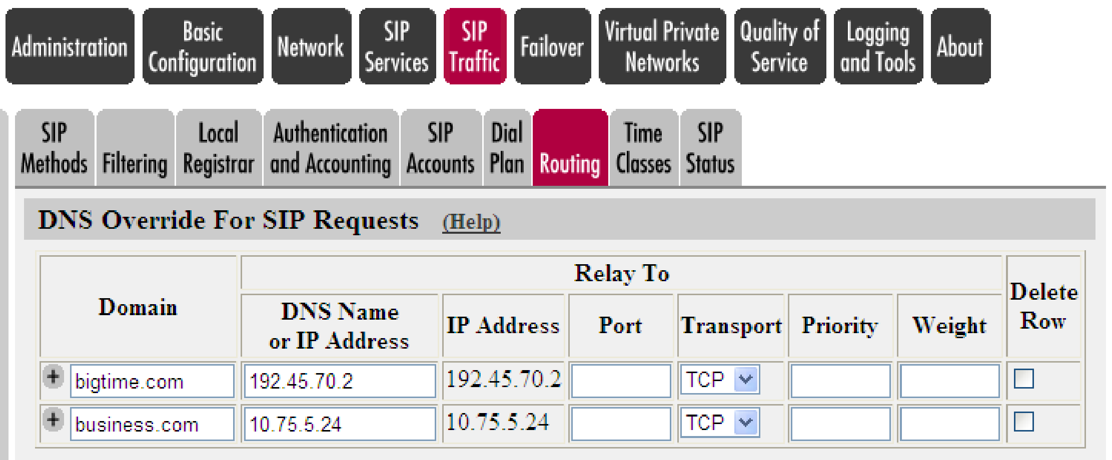
Step	Description
5.	**IP-PBX Settings** Select the **IP-PBX** tab. Select ***Avaya Aura SM*** from the **Type** drop-down menu (this selection was available from Startup Tool version 2.6.0 or greater). This will instruct the Startup Tool to configure the SIP parameters on the internal interface to be compatible with the Avaya component (Session Manager in this case) connected to it through direct SIP trunking interface. Enter the Session Manager IP address in the **IP Address** field. Also check the option to **use domain name**, then specify the domain name as set on Session Manager (see **Section 5 Step 2**) 

Step	Description
6.	**Service Provider Settings** Select the **ITSP_1** tab. Select ***Generic ITSP*** from the **Name** drop-down menu. This will instruct the Startup Tool to configure the SIP parameters on the external interface to be compatible with a generic SIP service provider. In the **Domain** field in the **Provide address** section, enter the domain for the 2nd site simulating a service provider service node and check the **Use domain name** option box. 

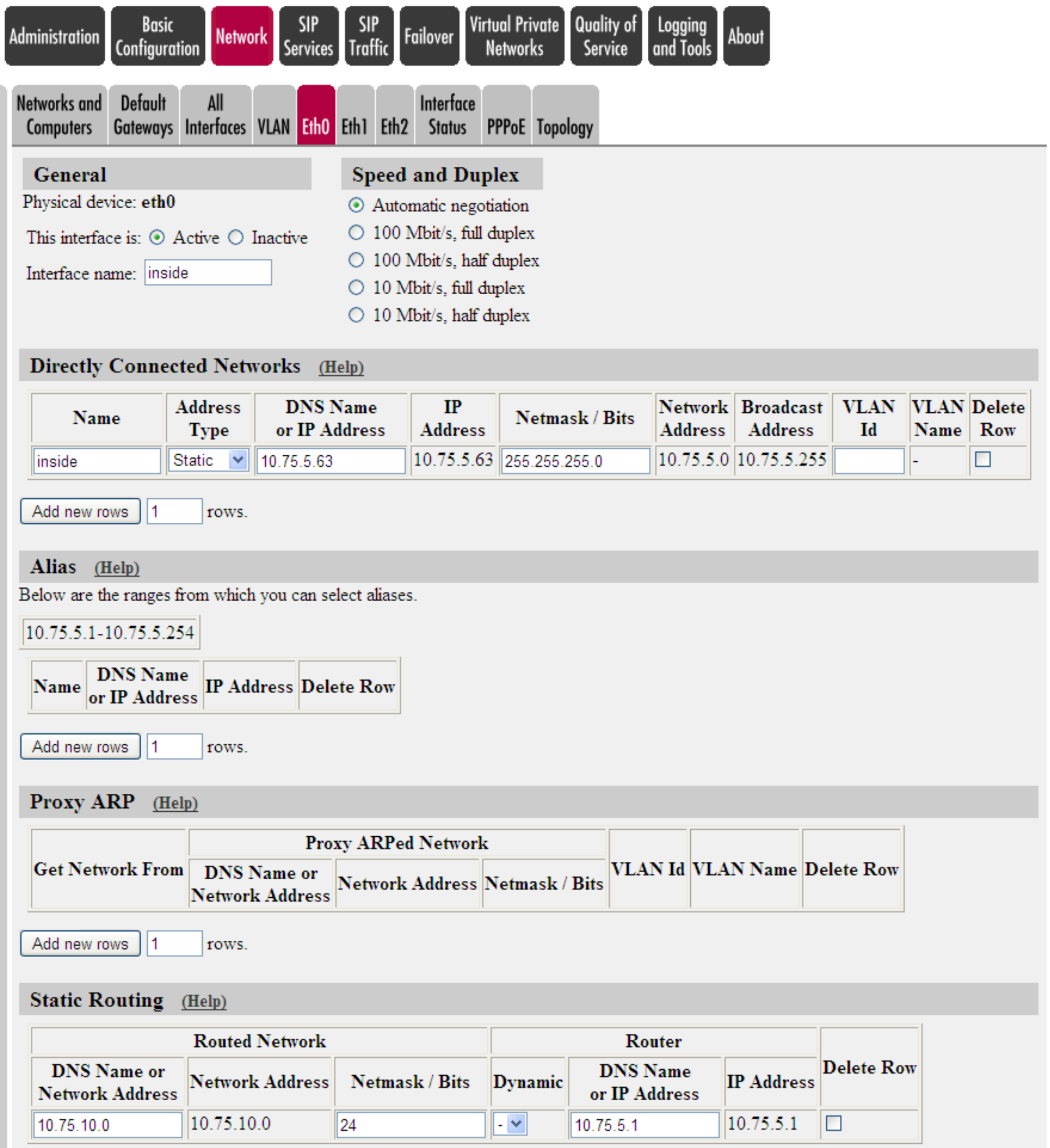
Step	Description
7.	**Upload Configuration** Select the **Upload Configuration** tab to upload the configuration to the SIParator. Click the **Upload** button to begin the upload.



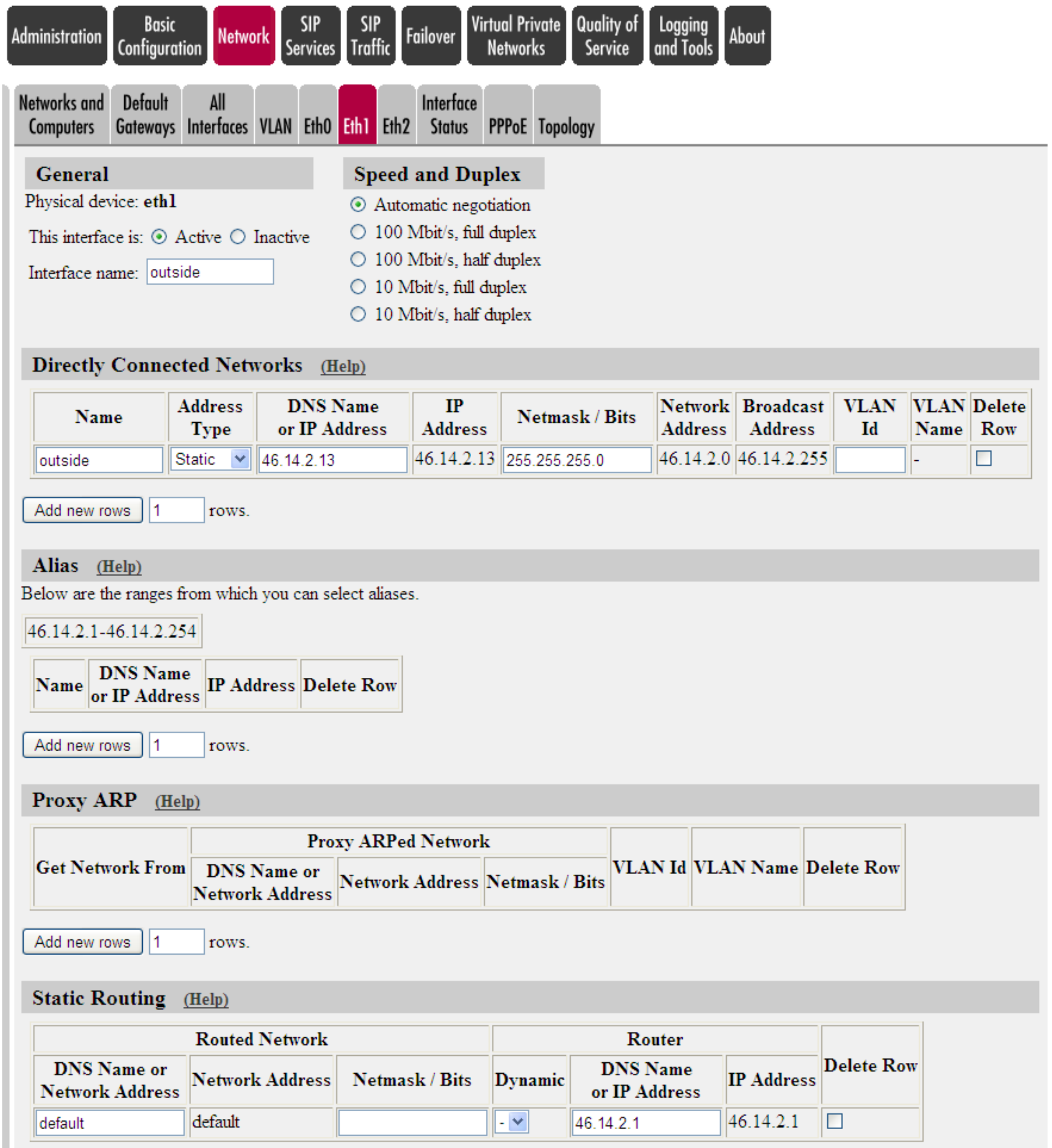
Step	Description
8.	**Apply Configuration** After uploading the configuration, the Startup Tool opens a web browser to the **Administration→Save/Load Configuration** page of the SIParator. Click the **Apply configuration** button to apply the configuration. The Startup Tool configuration is complete at this point. However, additional configuration was required to support all the test cases in the compliance test. This configuration is performed using the SIParator web interface and is covered in the remaining steps. 

Step	Description
9.	**Configure Routing** Navigate to **SIP Traffic→Routing** to add entries for DNS override for SIP requests. Add one entry for the outside interface and one entry for the inside interface as shown below. The configured parameters are: • **Domain**: domain names for the main enterprise site (***business.com***) and the 2nd site simulating a service provider service node (***bigtime.com***) • **DNS Name or IP Addresss**: IP addresses for the Avaya components connected to the SIParator on the outside (2nd site Communication Manager IP address ***192.45.70.2***) and on the inside (Session Manager IP address ***10.75.5.24***) • Transport: select ***TCP*** 

Step	Description
10.	**Configure Eth0 Inside Interface** In order to support endpoints on networks within the enterprise other than the subnet to which the SIParator is directly connected, a static route must be configured on the internal interface. In the case of the compliance test, one endpoint was located on the 10.75.10.0/24 network. Thus, to view the static route configured for this network, navigate to **Network→Eth0**. Scroll down to the **Static Routing** section. In this case, the routed network with **Network Address** ***10.75.10.0*** and **Netmask** of ***255.255.255.0*** is reached using **Router IP address** ***10.75.5.1***.



Step	Description
11.	**Configure Eth1 Outside Interface** The Eth1 outside interface is shown below for reference and completeness.



8. General Test Approach and Test Results

This section describes the compliance testing used to verify the interoperability of the Ingate SIParator with Session Manager and Communication Manager using SIP trunking. This section covers the general test approach and the test results.

8.1. General Test Approach

The general test approach was to make calls between the main enterprise site and the 2nd site simulating a service provider service node using various codec settings and exercising common PBX features.

8.2. Test Results

The Ingate SIParator passed compliance testing. The following features and functionality were verified. Any observations related to these tests are listed at the end of this section.

- Calls from both SIP and non-SIP endpoints between sites.
- G.711MU and G.729A codec support
- Proper recognition of DTMF transmissions by navigating voicemail menus.
- Proper operation of voicemail with message waiting indicators (MWI).
- PBX features including Hold, Transfer, Call Waiting, Call Forwarding and Conference.
- Extended telephony features using Communication Manager Feature Name Extensions (FNE) such as Conference On Answer, Call Park, Call Pickup, Automatic Redial and Send All Calls. For more information on FNEs, please refer to [4].
- Proper system recovery after a SIParator restart and loss of IP connection.

The following observation was made during the compliance test.

- When the SIParator was hard-reset to simulate the adverse condition of power outage, the SIP trunk between the SIParator and the Session Manger would not come back to the normal in-service state unless the Session Manager was restarted too.

9. Verification Steps

The following steps may be used to verify the configuration:

- Using System Manager **Monitoring** (from left navigation pane), verify that Entity Links to the SIParator and Communication Manager are up.
- From the Communication Manager SAT, use the **status signaling-group** command to verify that the SIP signaling group is in-service.
- From the Communication Manager SAT, use the **status trunk-group** command to verify that the SIP trunk group is in-service.
- Verify that calls can be placed from both SIP and non-SIP endpoints between sites.

10. Conclusion

The Ingate SIParator passed compliance testing. These Application Notes describe the procedures required to configure the Ingate SIParator to interoperate with Session Manager and Communication Manager to support the network shown in **Figure 1** where Session Manger connects the SIParator to Communication Manager using SIP trunking interface.

11. Additional References

[1] *Avaya Aura™ Communication Manager Feature Description and Implementation*, Doc # 555-245-205, May 2009.

[2] *Administering Avaya Aura™ Communication Manager*, Doc # 03-300509, May 2009.

[3] *SIP support in Avaya Aura™ Communication Manager Running on the Avaya S8xxx Servers*, Doc # 555-245-206, May 2009.

[4] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.0*, version 6.0, Doc # 210-100-500, Issue 9, June 2005

[5] *Administering Avaya Aura™ SIP Enablement Services on the Avaya S8300 Server*, Doc # 03-602508, May 2009.

[6] *Avaya IA770 INTUITY AUDIX Messaging Application Release 5.1 Administering Communication Manager Servers To Work with IA770*, June 2008.

[7] *Avaya Aura™ Session Manager Manage Overview*, Doc # 03-603323

[8] *Installing and Administering Avaya Aura™ Session Manager*, Doc # 03-603324

[9] *Maintaining and Troubleshooting Avaya Aura™ Session Manager*, Doc # 03-603325

[10] *Ingate SIParator Getting Started Guide*

[11] *Ingate SIParator Reference Guide.*

Product documentation for Avaya products may be found at http://support.avaya.com.

Product documentation for the SIParator can be obtained from Ingate. Contact Ingate using the contact link at http://www.ingate.com.

Appendix A: Communication Manager Configuration at 2nd Site

This section contains specific configuration screens that are important to the Communication Manager at the 2nd site simulating a service provider service node.

The **node-names ip** form: note the SIParator and its public side IP address.

```
display node-names ip
                                  IP NODE NAMES
    Name              IP Address
SES                 192.45.70.7
SIParator           46.14.2.13
procr               192.45.70.2
```

The **signaling-group** form (for outgoing calls): note the **Far-end Node Name** and **Far-end Domain** settings.

```
display signaling-group 36
                                SIGNALING GROUP

 Group Number: 36             Group Type: sip
                         Transport Method: tcp
  IMS Enabled? n




   Near-end Node Name: procr                Far-end Node Name: SIParator
 Near-end Listen Port: 5060               Far-end Listen Port: 5060
                                       Far-end Network Region: 1
Far-end Domain: 46.14.2.13

                                            Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                    IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

The **trunk-group** form (for outgoing calls): note the **Group Name** and **Signaling Group** settings.

```
display trunk-group 36                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 36                  Group Type: sip          CDR Reports: y
  Group Name: ToSIParator                COR: 1        TN: 1        TAC: *036
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

                                                        Signaling Group: 36
                                                      Number of Members: 10
```

The **signaling-group** form (for incoming calls): note the **Far-end Node Name** and **Far-end Domain** settings.

```
display signaling-group 37
                                SIGNALING GROUP

 Group Number: 37             Group Type: sip
                        Transport Method: tcp
  IMS Enabled? n






   Near-end Node Name: procr                   Far-end Node Name: SIParator
 Near-end Listen Port: 5060                  Far-end Listen Port: 5060
                                          Far-end Network Region: 1
Far-end Domain:

                                             Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                       Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

The **trunk-group** form (for incoming calls): note the **Group Name** and **Signaling Group** settings.

```
display trunk-group 37                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 37                    Group Type: sip          CDR Reports: y
  Group Name: FromSIParator                COR: 1        TN: 1        TAC: *037
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n

                                                       Signaling Group: 37
                                                     Number of Members: 10
```

The **public-unknown-numbering** form:

```
display public-unknown-numbering 0                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                          Total
Ext Ext           Trk       CPN            CPN
Len Code          Grp(s)    Prefix         Len
                                                     Total Administered: 1
 5  5                                      5            Maximum Entries: 240
                                                     Number of Members: 10
```

The **aar analysis** form:

```
display aar analysis 3                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    2

          Dialed          Total     Route     Call   Node  ANI
          String         Min  Max  Pattern    Type   Num   Reqd
    30                   5    5     36        aar          n
```

The **route-pattern** form: note that trunk group 36 was defined for routing outgoing calls (to the SIParator for onward routing to the main enterprise site).

```
display route-pattern 36                                        Page   1 of   3
                    Pattern Number: 36  Pattern Name: ToSIPArator
                             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: 36   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
```

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