Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura™ Communication Manager 6.0, Avaya Aura™ Session Manager 6.0, Avaya Aura™ Session Border Controller 6.0 with Skype Connect 1.3 – Issue 1.0

Abstract

These Application Notes describe the steps to configure the Avaya Aura™ R6 SIP reference architecture with the Skype Connect SIP trunking service.

Skype Connect allows Skype's 600 million registered user community to contact business users through click-to-call applications or by calling the Skype Names of business users associated with phone extensions on Avaya Aura™ without the need for TDM media gateways and the associated maintenance costs. Skype Connect also provides a low-cost inbound and outbound PSTN calling service with DID and Caller ID support.

Testing was conducted in the Avaya Solution and Interoperability Test Lab, utilizing a Skype Manager account on the Skype Connect production service.

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1. Introduction

These Application Notes describe the steps to configure the Avaya Aura™ R6 SIP reference architecture with the Skype Connect SIP trunking service. Skype Connect allows Skype's 600 million registered user community to contact business users through click-to-call applications or by calling the Skype Names of business users associated with phone extensions on Avaya Aura™ without the need for TDM media gateways and the associated maintenance costs. Skype Connect also provides a low-cost inbound and outbound PSTN calling service with DID and Caller ID support.

The Avaya Aura™ R6 SIP reference architecture consists of Avaya Aura™ Communication Manager, Avaya Aura™ Session Manager, Avaya Aura™ System Manager and Avaya Aura™ Session Border Controller. Avaya Aura™ Communication Manager controls the Avaya H.323, digital, and analog endpoints, normalizes the called and calling numbers for both incoming and outgoing calls to/from Skype Connect and provides telephony features such as Call Forward, Transfer and Call Pickup. The role of the Avaya Aura™ Session Manager in the reference architecture is to act as a Registrar for Avaya SIP endpoints, SIP Proxy for outbound/inbound trunk calls while providing a centralized dial-plan for least-cost and time-of-day based routing. Avaya Aura™ System Manager provides a web-based interface for the provisioning and maintenance of Avaya Aura™ Session Manager while the Avaya Aura™ Session Border Controller provides topology hiding without the need for Network Address Translation (NAT), SIP header manipulation and SIP signaling and media channel conversion services. While not the focus of this testing, a SIP-integrated Avaya Modular Messaging (version 5.2) system was used to provide enterprise voicemail call coverage for Avaya telephones. For an illustrative example of configuring Avaya Modular Messaging as a SIP-based centralized voicemail system see **Reference [1]**.

1.1. Design Principles and Assumptions

The service offer described in these Application Notes is designed for business customers using Avaya Aura™ Communication Manager and Avaya Aura™ Session Manager on a private protected enterprise network who opt for routing their voice calls over the public Internet using the Skype Connect service. It is assumed that Skype Connect is used as first choice for outbound calls and that business customers will keep some of their existing TDM connections to the PSTN as Skype Connect does not support calls to emergency services.

It is also assumed that the Avaya Aura™ Session Border Controller (AASBC) acts as a peering host between the public Internet and the private enterprise network and provides Denial-of-Service (DoS), packet filtering and topology hiding without the need for an additional firewall or intrusion prevention system (IPS) on either the public or private side of the Avaya Aura™ Session Border Controller. Although the Avaya Aura™ Session Border Controller can be configured to provide intelligent call routing decisions, no dial-plan was provisioned on the Avaya Aura™ Session Border Controller in the sample configuration as all the call routing and number modification logic is achieved by Avaya Aura™ Communication Manager and Avaya Aura™ Session Manager. Hardware, software resilience and failover between the various Avaya components is not covered in these Application Notes.

1.2. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound and outbound call flows (see **Section 3** for examples) and basic supplementary features between Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager, Avaya Aura™ Session Border Controller and the Skype Connect service.

The compliance testing was based on a test plan provided by TekVizion, for the functionality required for certification as a solution supported on the Skype Connect network. The following features were tested as part of this effort:

- SIP trunking.
- Passing of DTMF events and their recognition by navigating automated menus.
- Supplementary features such as hold, resume, conference and transfer.

1.3. Abbreviations

The abbreviations used in this document include the following:

Abbreviation	Description
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
HQ	Headquarters
B2BUA	Back-to-back User Agent
PE	Processor Ethernet
P2P	Peer-to-peer
AOR	Address of record
DNIS	Dialed Number Identification Service

1.4. Known Limitations

The following limitations are noted for the reference configuration described in these Application Notes:

- Skype Connect is currently U.S. only. The service will be introduced in other regions at a later stage.
- Skype Connect does not support calls to the emergency service. Another PSTN trunk must be provisioned in Avaya Aura™ Communication Manager to route calls to the emergency service.
- Porting of existing PSTN numbers (DIDs) to Skype Connect is not supported.
- Access to a broadband internet connection is required.
- Maximum of 300 simultaneous calls per SIP Profile. A company may have multiple SIP Profiles.
- SIP over TLS is not currently supported by Skype Connect.
- Call processing tones are locally generated by Avaya Aura™ Communication Manager.
- Premium-rated numbers (1900, 1976) are blocked.
- This solution does not currently support outbound SIP calls to Skype names.
- DNS A records are supported for Skype Connect service node name resolution, while DNS SRV records will be introduced at a later stage.
- The SIP REFER request is not supported for call redirection/transfer.
- SIP 3xx Redirect Responses are not supported.
- SRTP is not supported.
- IP TOS or DiffServ QoS markings are neither set nor honored, therefore Skype Connect cannot guarantee the end-to-end voice quality. Service Level Agreements (SLAs) are not available.
- Maximum 99 Online Numbers per SIP Profile. Sequential number block (DID) purchases will be introduced at a later stage.
- G.711A/mu-law, G.729 codecs are supported.
- For outbound calls (local, national and international) via Skype Connect the E.164 or International numbering format (00 + <country code>) must be used.
- For inbound calls Skype Connect delivers the called/calling number in E.164 format
- T.38 fax is not supported.
- RTCP and RTCP XR are not supported.
- Skype Connect calls are limited to 4 hours.
- High Availability with two physically separate Avaya Aura™ Session Border Controllers is not supported in R6.0.
- The SIP REGISTER method is not currently supported by Avaya Aura™ Session Border Controller R6.0
- Skype Connect is not guaranteed to work with credit card machines, franking (stamping) machines and alarm systems or other services which use a regular phone line with a modem connection.

1.5. Reference Configuration

Figure 1 illustrates Avaya Aura™ R6 SIP reference architecture used for Interoperability testing. The reference configuration is comprised of a sample enterprise HQ site connected via a Metro Ethernet link to the Internet. At the edge of the test HQ site an Avaya Aura™ Session Border Controller acts as a B2BUA for SIP calls. The Avaya Aura™ Session Border Controller terminates and re-originates calls using its own IP addresses thereby hiding the IP address range (topology) of the private network. Network security is provided by the DoS and packet filtering module of the Avaya Aura™ Session Border Controller. The Avaya Aura™ Session Border Controller converts the SIP signaling channel from UDP to TCP for inbound and vice-versa for outbound calls.

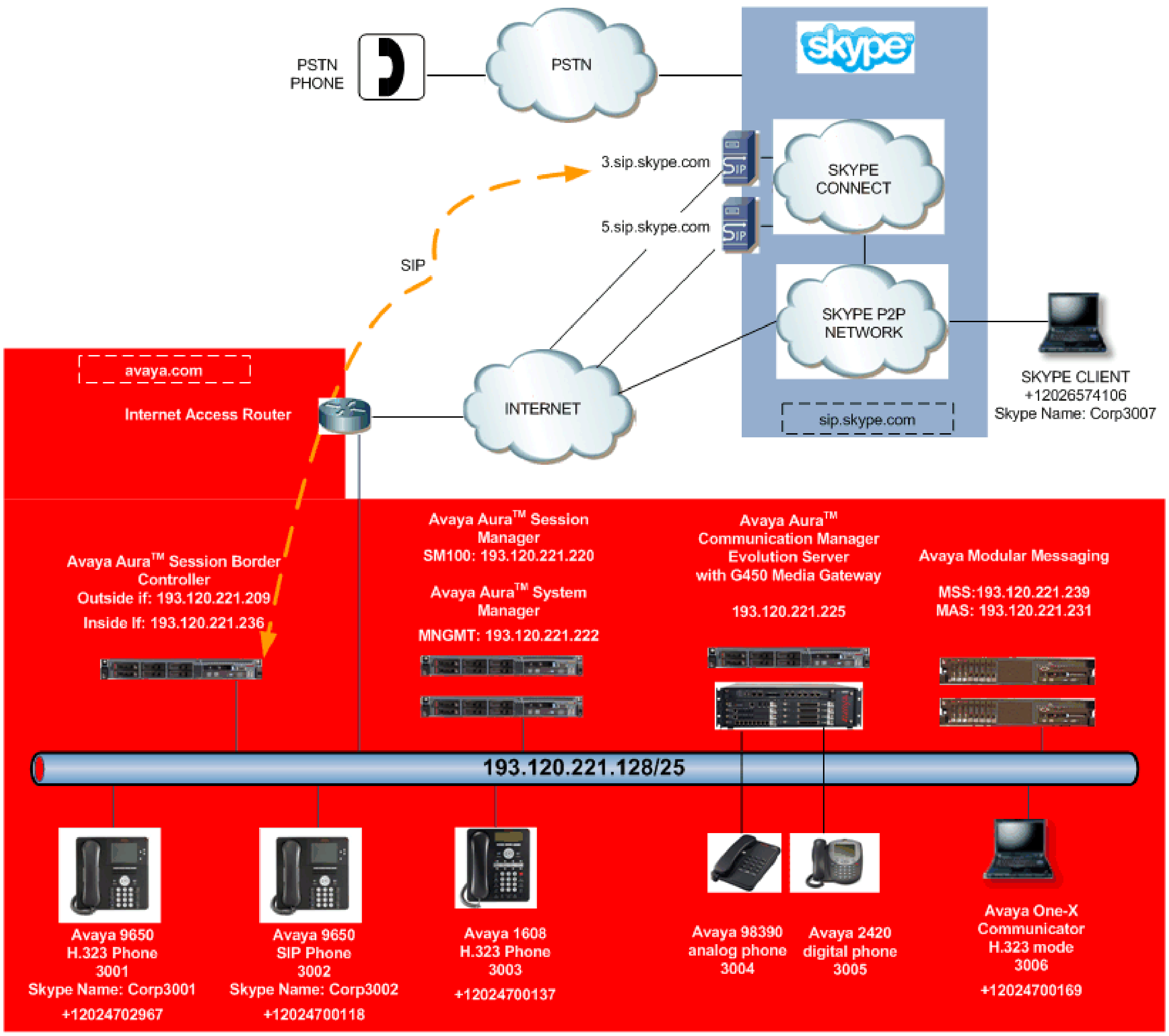


Figure 1: Avaya Interoperability Test Lab Reference Configuration

1.6. Support

For technical support on Avaya products described in these Application Notes visit http://support.avaya.com

For technical support on the Skype Connect service visit http://www.skype.com/support

2. Equipment and Software Validated

The following equipment and software were used in the reference configuration.

Equipment	Software Version
Avaya S8800 Server	Avaya Aura™ Session Manager - R6.0.0.0.600020
Avaya S8800 Server	Avaya Aura™ System Manager - R6.0.0.0.600020
Avaya S8800 Server	Avaya Aura™ Session Border Controller – R6.0.0.2.4
Avaya S8800 Server	Avaya Aura™ Communication Manager – Evolution Server - R016x.00.0.345.0
Avaya S8730 Server	Avaya Messaging Application Server – R5.2
Avaya S8730 Server	Avaya Message Store Server – R5.2
Avaya G450 Media Gateway	R30.12.1
Avaya 9650 H.323 Phone	R3.1.1
Avaya 9650 SIP Phone	R2.6.2.21
Avaya 1608 H.323 Phone	R1.2
Avaya 2420 Digital Phone	N/A
Avaya 98390 Analog Phone	N/A
Test PC1	Microsoft Windows Vista with Avaya One-X Communicator (H.323) R6.0.0.26
Test PC2	Skype Client v4.2.0.169
Skype Connect	Version 1.3

Table 1: Equipment and Software Used in the Reference Configuration

3. Call Flows

To understand how inbound and outbound Skype Connect service calls are handled by Avaya Aura™ Session Border Controller, Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager, four general call flows are described in this section.

3.1. Inbound from PSTN

The first call scenario illustrated in **Figure 2** is an inbound call from a PSTN phone to the Online Number of a phone registered to Avaya Aura™ Communication Manager on the enterprise network.

1. PSTN user dials the Skype Connect Online Number of the enterprise user (12024703313).
2. The call is routed to the Skype Connect service network.
3. The Skype Connect service sends the call to the publicly routable IP address of the outside interface of the Avaya Aura™ Session Border Controller.
4. The call is delivered to the Avaya Aura™ Session Border Controller using SIP over UDP.
5. The Avaya Aura™ Session Border Controller performs SIP header modifications, transport protocol conversion (UDP to TCP) and routes the call to Avaya Aura™ Session Manager.
6. Avaya Aura™ Session Manager performs call routing based on the called number and the configured Network Routing Policies. In the sample configuration the Online Number is associated with the Avaya Aura™ Communication Manager SIP Entity.
7. Avaya Aura™ Communication Manager routes the call to a phone.

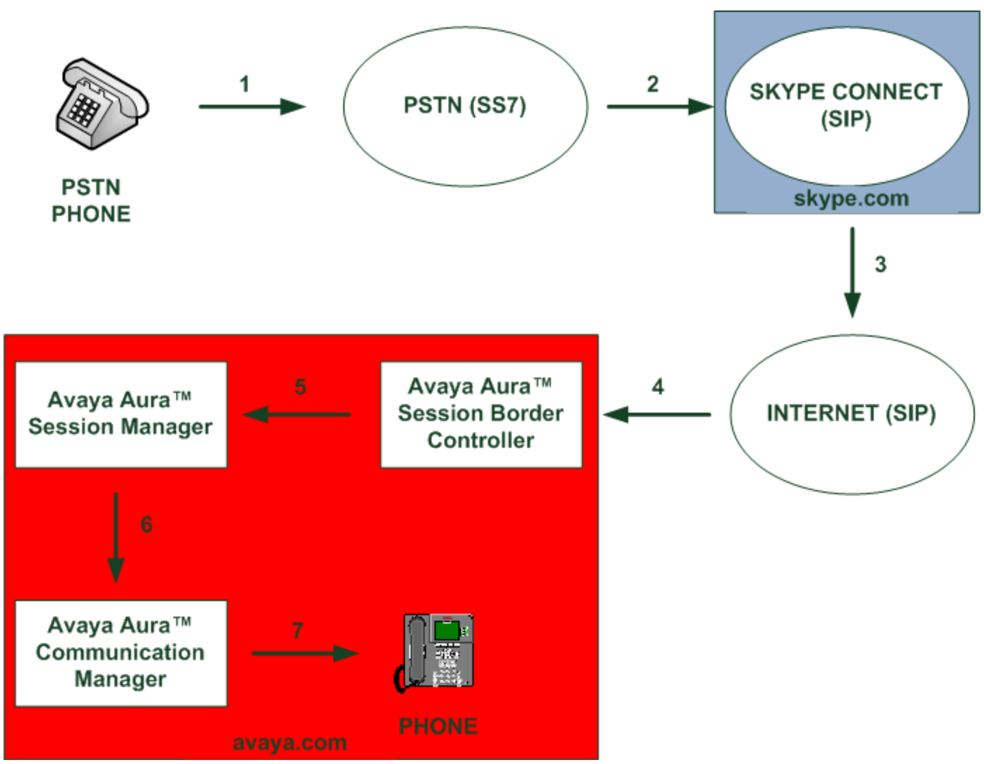


Figure 2 - Inbound PSTN to Skype Connect Call

Note: A single Avaya Aura™ Session Border Controller was used in the sample configuration as High Availability is not supported in Release 6.0.

3.2. Outbound to PSTN

The second call scenario illustrated in **Figure 3** is an outbound call with CLIP, that originates from a phone registered to Avaya Aura™ Communication Manager on the enterprise network and terminates at the PSTN phone.

1. Avaya phone originates a call to a PSTN number.
2. Avaya Aura™ Communication Manager converts the calling and called numbers to an E164 format before it routes the call to Avaya Aura™ Session Manager using SIP over TCP.
3. Based on the called number Avaya Aura™ Session Manager sends the call to Avaya Aura™ Session Border Controller.
4. Avaya Aura™ Session Border Controller queries the public DNS server for "sip.skype.com". The public DNS server returns the IP address of the geographically closest Skype Connect peer in an A record. The call is sent using SIP over UDP through the Internet.
5. The SIP INVITE arrives to the public interface of the Skype Connect peer.
6. The Skype Connect network compares the contents of the PAI header with the Online Number of the caller's SIP Profile. If the two E164 numbers match the Caller ID gets set before the call breaks out to the PSTN. Otherwise the call is delivered to the PSTN with no Caller ID or a default Caller ID displayed for all outbound calls if one has been pre-configured in Skype Manager.
7. The call is routed to the PSTN phone.

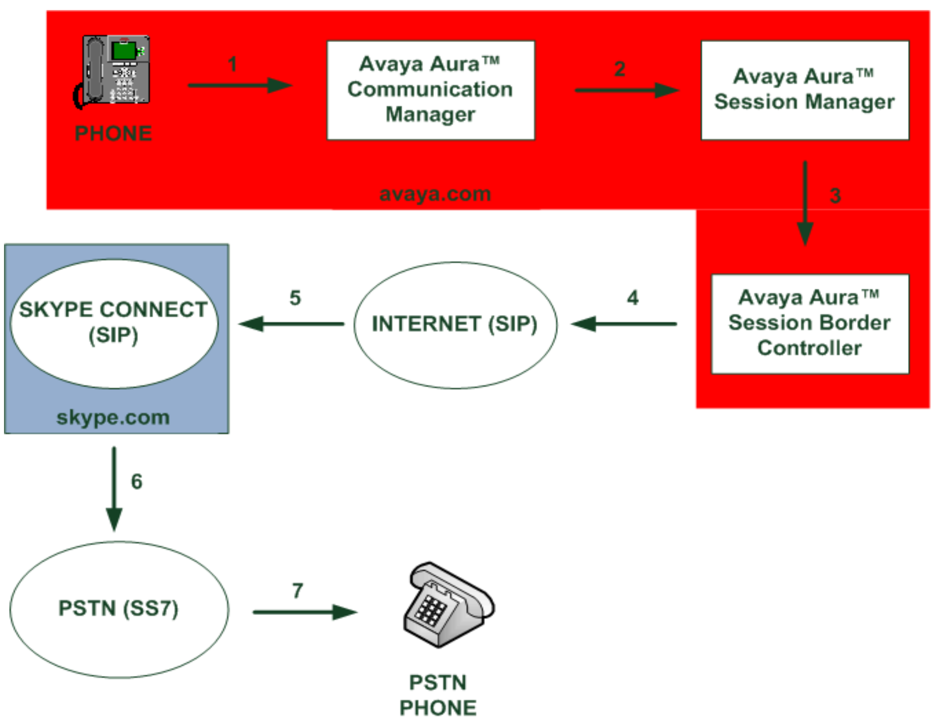


Figure 3: Outbound Skype Connect to PSTN call with CLIP

Note: Avaya Aura™ Session Border Controller caches DNS A records until the TTL value expires.

3.3. Inbound from Skype Client

The third call scenario illustrated in **Figure 4** is an inbound call made from a Skype client to a phone registered to Avaya Aura™ Communication Manager on the enterprise network. **Note:** The call stays within the skype.com service domain and does not break out to the PSTN.

1. Skype user initiates a call by double-clicking the Skype Name of the enterprise user associated with the SIP Profile.
2. The call is routed from the Skype P2P Network to the Skype Connect service network.
3. The Skype Connect service sends the call to the publicly routable IP address of the outside interface of the Avaya Aura™ Session Border Controller.
4. The call is delivered to the Avaya Aura™ Session Border Controller using SIP over UDP.
5. The Avaya Aura™ Session Border Controller performs SIP header modifications, transport protocol conversion (UDP to TCP) and routes the call to Avaya Aura™ Session Manager.
6. Avaya Aura™ Session Manager performs call routing based on the called number and the configured Network Routing Policies. In the sample configuration the Online Number is associated with the Avaya Aura™ Communication Manager SIP Entity.
7. Avaya Aura™ Communication Manager routes the call to a phone.

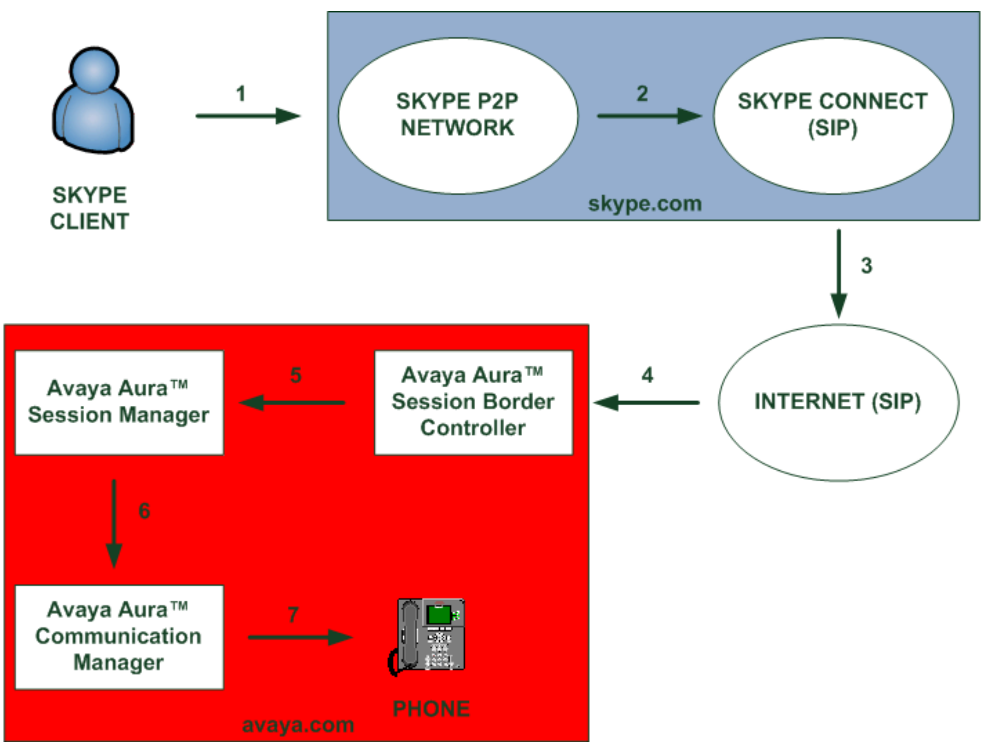


Figure 4: Inbound Skype P2P to Skype Connect call

3.4. Skype Connect to Skype P2P Network

The fourth call scenario illustrated in **Figure 5** is an outbound call made from a phone registered to Avaya Aura™ Communication Manager on the enterprise network to the Online Number of a Skype Client. **Note:** The call stays within the skype.com service domain and does not break out to the PSTN.

1. Avaya phone originates a call to the Online Number of the Skype Client.
2. Avaya Aura™ Communication Manager converts the calling and called numbers to an E164 format before it routes the call to Avaya Aura™ Session Manager using SIP over TCP.
3. Based on the called number Avaya Aura™ Session Manager sends the call to Avaya Aura™ Session Border Controller.
4. Avaya Aura™ Session Border Controller queries the public DNS server for "sip.skype.com". The public DNS server returns the IP address of the geographically closest Skype Connect peer in an A record. The call is sent using SIP over UDP through the Internet.
5. The SIP INVITE arrives to the public interface of the Skype Connect peer.
6. The Skype Connect network recognizes that the called number belongs to a Skype Client (Online Number) and routes the call to the Skype P2P Network.
7. The call is delivered to the Skype Client.

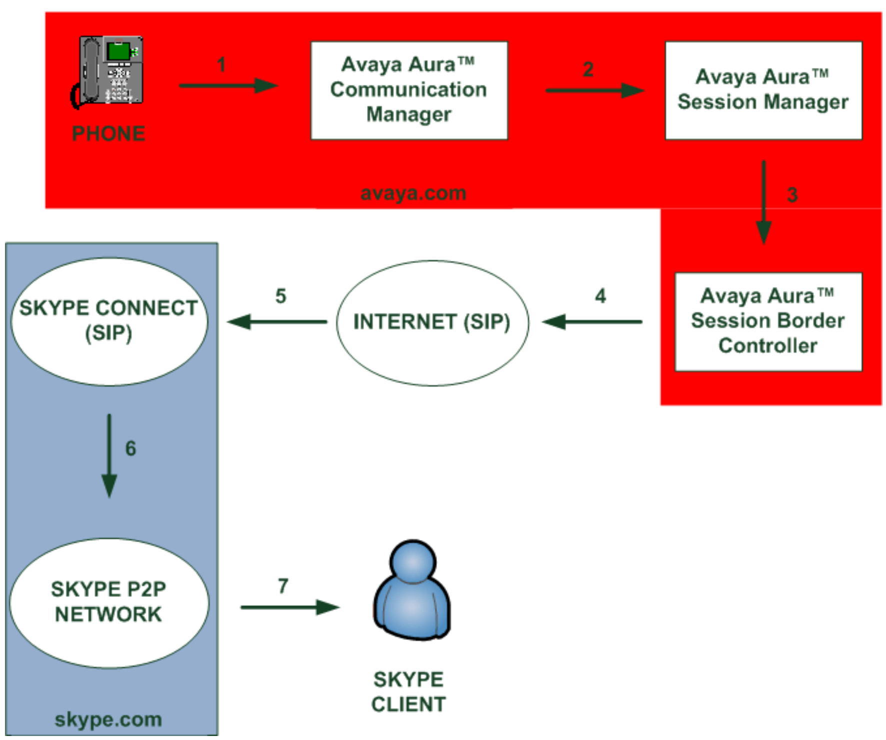


Figure 5: Outbound Skype Connect to Skype P2P Client call

4. Configure Avaya Aura™ Session Manager

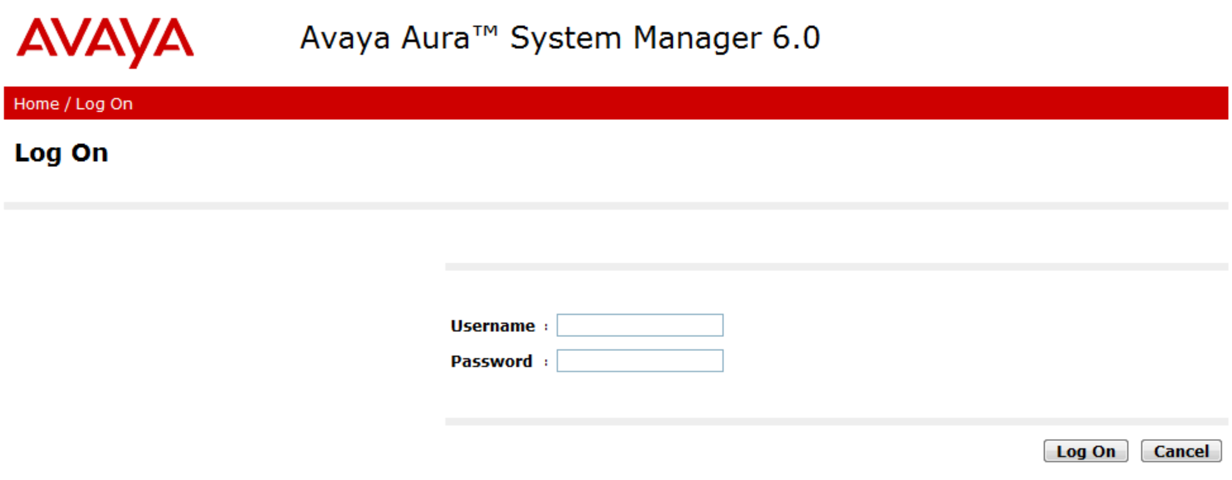
This section provides the procedures for configuring the Session Manager and includes the following items:

- Log in to Avaya Aura™ System Manager using the GUI
- Administer SIP domain
- Define a Location
- Define SIP Entities
- Define Entity Links
- Define Routing Policies
- Define Dial Patterns

The administration of Avaya endpoints and Communication Manager is not covered in these Application Notes. For further information on configuring Session Manager see **Reference [2]**.

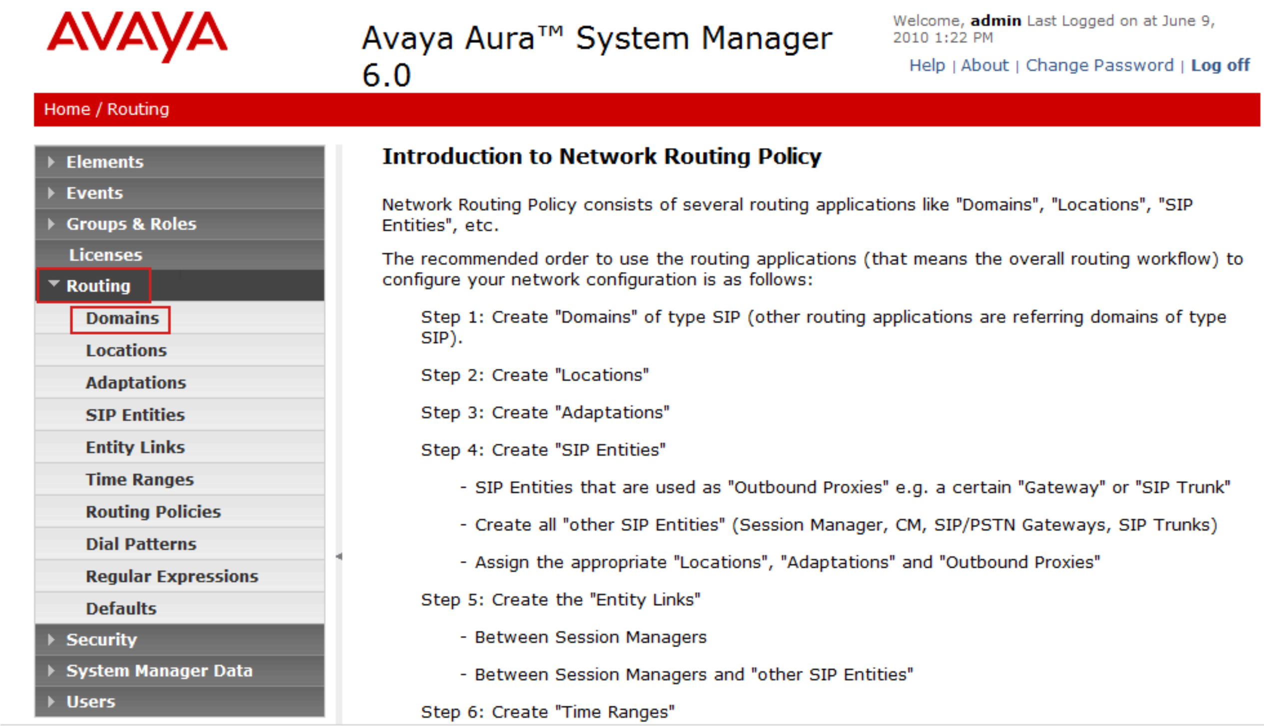
4.1. Log in to Avaya Aura™ System Manager using the GUI

Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL "**https://<ip-address>/SMGR**", where "<ip-address>" is the IP address of System Manager. Log in with the appropriate credentials.

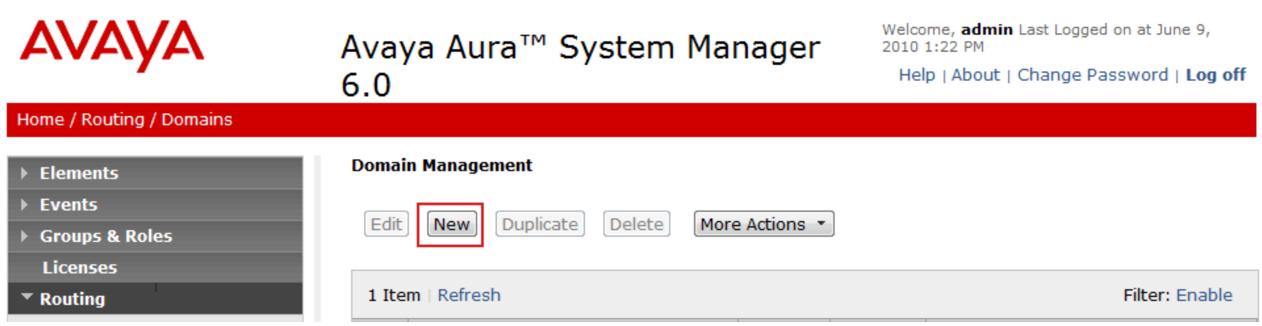


4.2. Administer SIP Domain

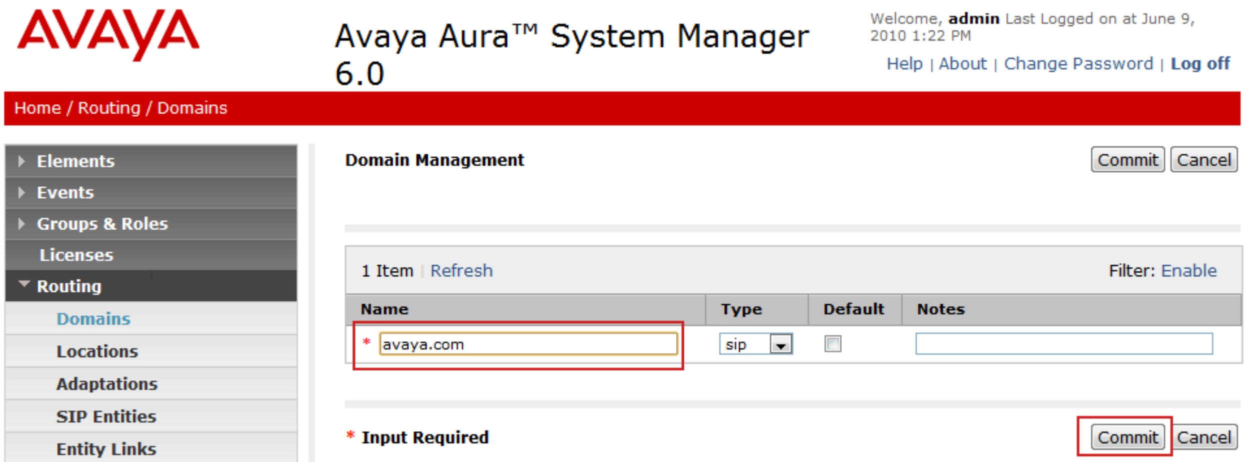
Expand **Routing** and select **Domains**.



Click **New.**

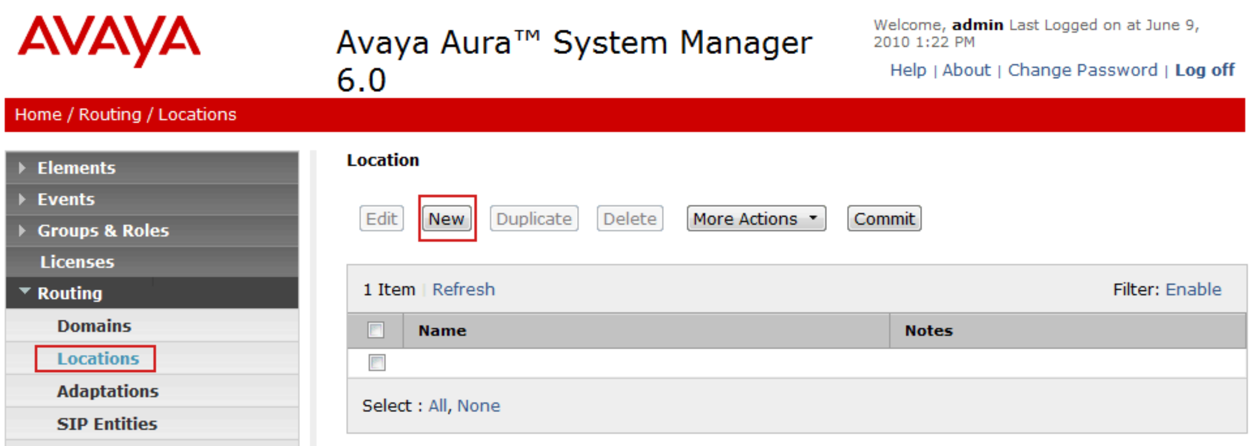


On the **Domain Management** screen under **Name** add a descriptive name. Retain the default values for the remaining fields. Click **Commit** to save.

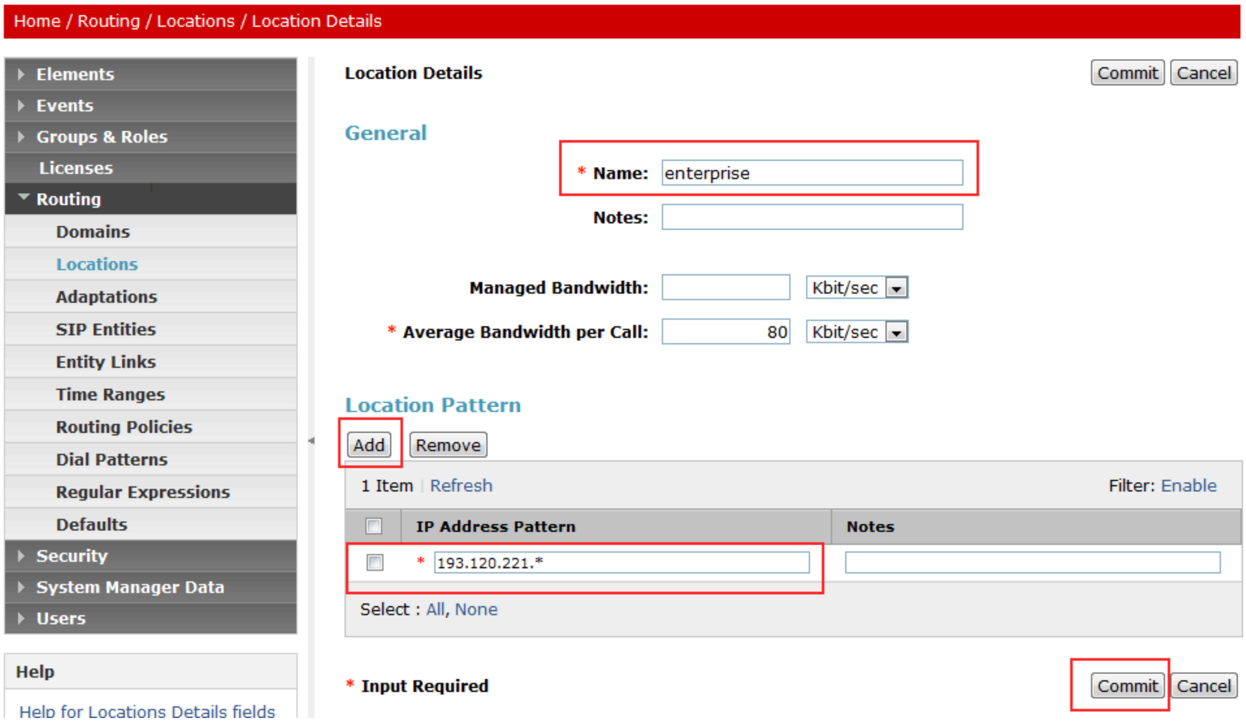


4.3. Define a Location

Expand **Routing** and select **Locations.** Locations are used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management or location-based routing. Click **New.**

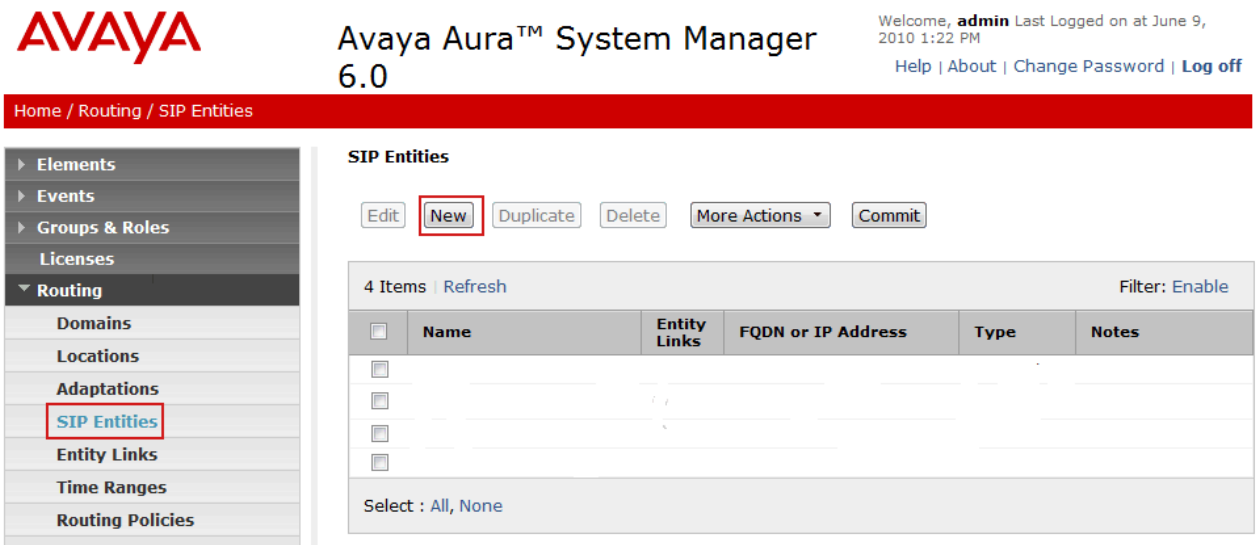


In the **General** Section, under **Name** add a descriptive name. Click on **Add**. In the **Location Pattern** Section under **IP Address Pattern** enter IP address of the subnet on which the SIP Entities (CM and AASBC) reside. Click **Commit** to save.



4.4. Define SIP Entities

Session Manager interconnects three SIP Entities – CM and AASBC - on the enterprise network. Under **Routing** in the left pane click **SIP Entities** then **New** to create a SIP Entity for Communication Manager.

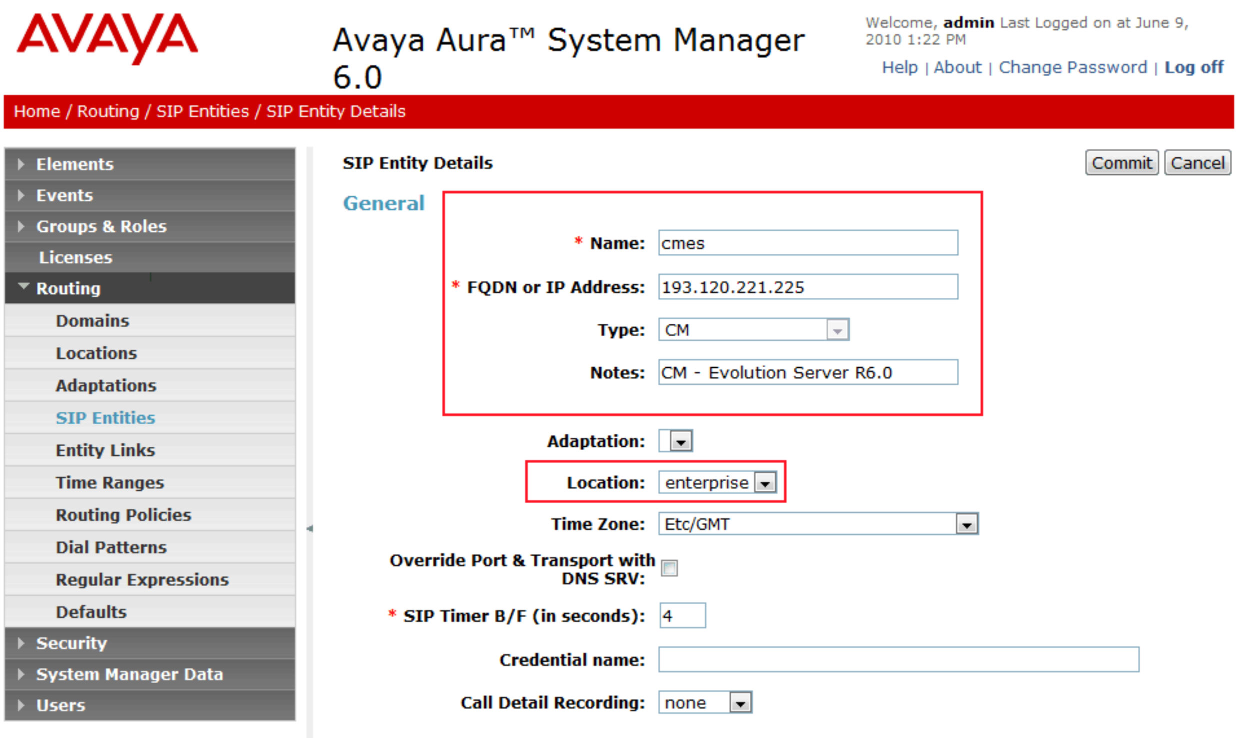


The **SIP Entity Details** screen is displayed.

Under **General**:

- **Name** Type in a descriptive name
- **FQDN or IP Address** Type IP address of the PE interface of Communication Manager
- **Type** Select **CM**
- **Notes** (Optional) Type in description
- **Location** Select the Location created in the previous step

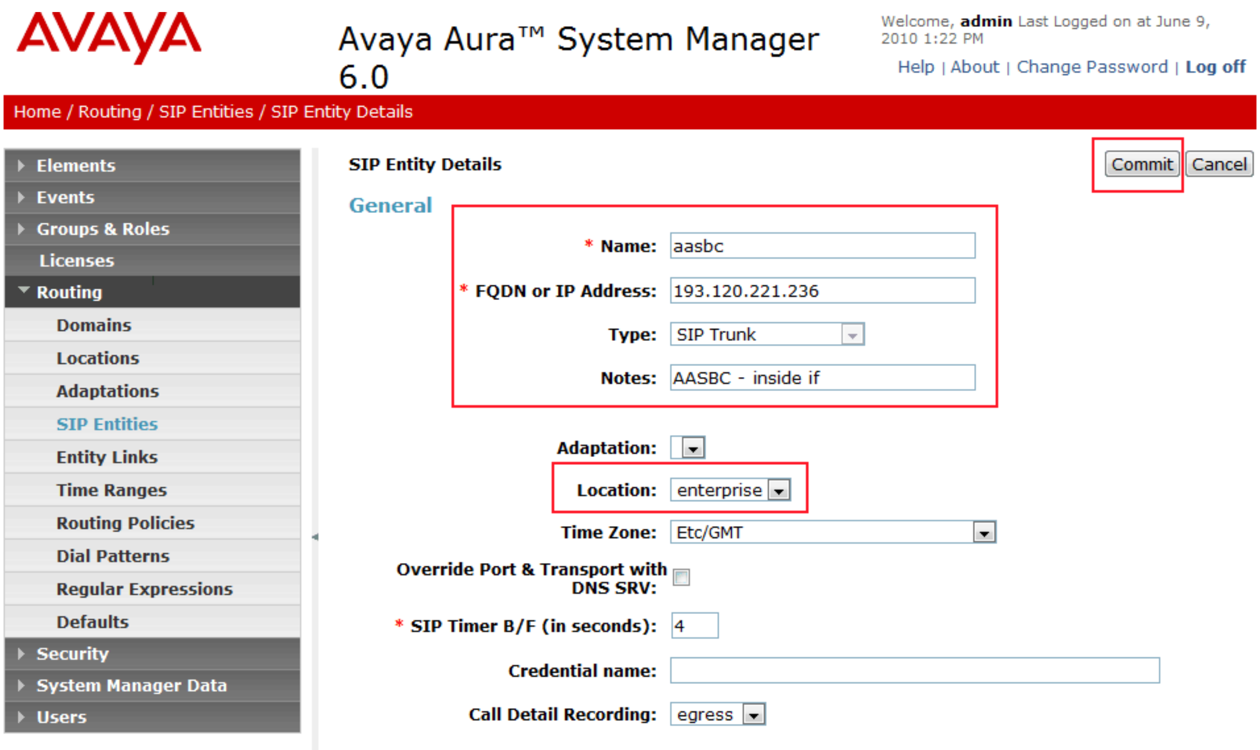
Click **Commit**.



Repeat the steps from the previous section. Click **New** on the **SIP Entities** page. The following screen shows addition of the Avaya Aura™ Session Border Controller as a SIP entity.
Under **General**:

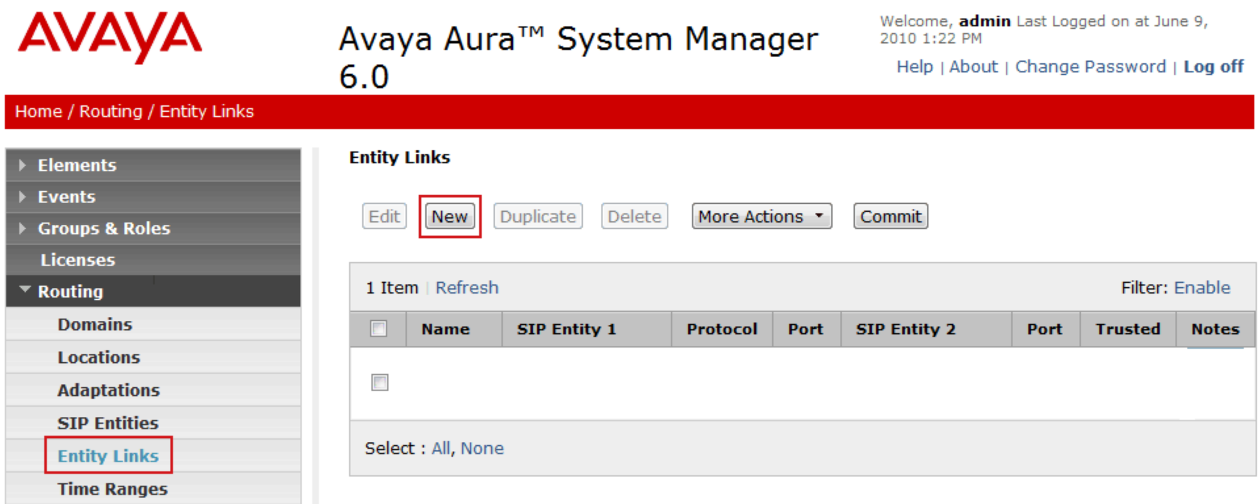
- **Name** Type in a descriptive name
- **FQDN or IP Address** Type IP address of the inside interface of the AASBC
- **Type** Select **SIP Trunk**
- **Notes** (Optional) Type in description
- **Location** Select the Location created in the previous step

Click **Commit**.



4.5. Define Entity Links

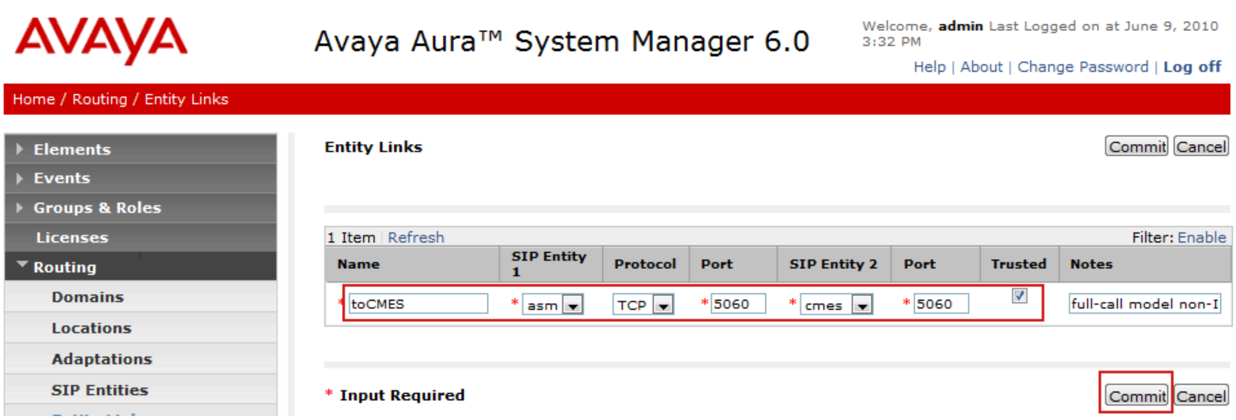
SIP trunks between Session Manager and the three SIP Entities (CM and AASBC) are described by Entity Links. To add an Entity Link for Communication Manager, select **Entity Links** on the left pane then click **New**.



The **Entity Links** screen is displayed. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name
- **SIP Entity 1:** Select the **Session Manager**.
- **Protocol:** Select **TCP**.
- **Port:** Type **5060**. The Session Manager listens for SIP requests on TCP port 5060.
- **SIP Entity 2:** Select the Communication Manager SIP Entity
- **Protocol:** Select **TCP**
- **Port:** Type **5060**. Communication Manager listens for SIP requests on TCP port 5060.
- **Trusted:** Check this box, otherwise SIP calls will be denied to/from Communication Manager

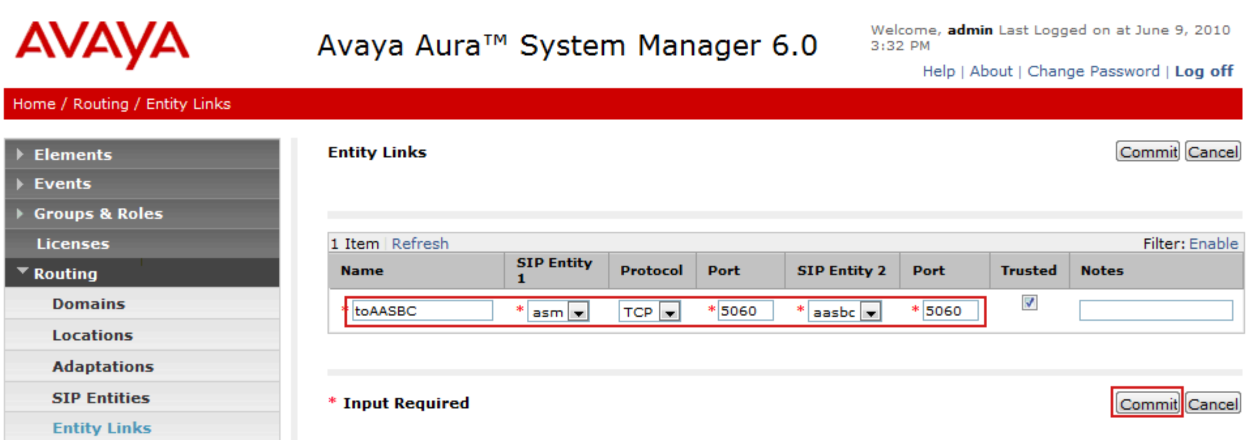
Click **Commit**.



Repeat the steps from the previous section. Click **New** on the **Entity Links** page to add a link to the Avaya Aura™ Session Border Controller. The **Entity Links** screen is displayed. Fill in the following fields in the new row that is displayed:

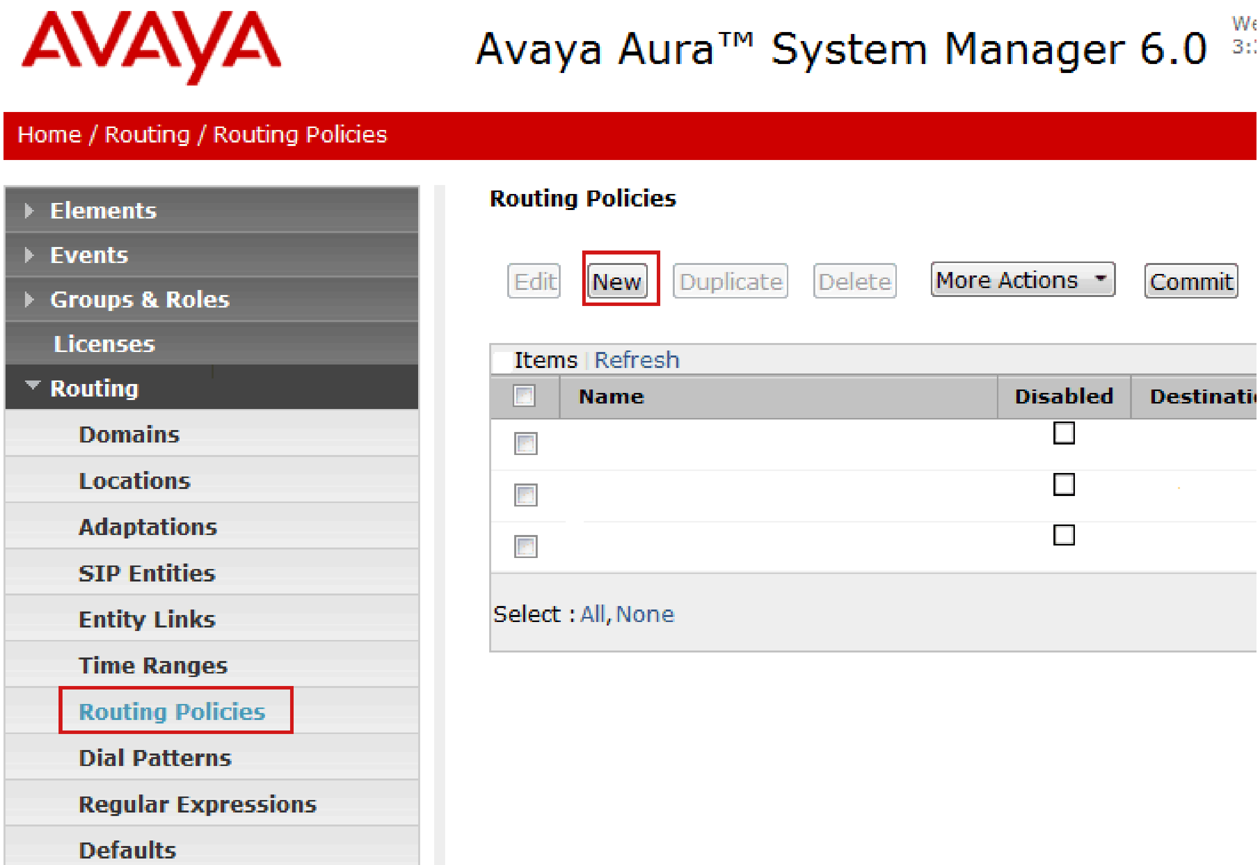
- **Name:** A descriptive name
- **SIP Entity 1:** Select the **Session Manager**.
- **Protocol:** Select **TCP**.
- **Port:** Type **5060**. The Session Manager listens for SIP requests on TCP port 5060.
- **SIP Entity 2:** Select the Avaya Aura™ Session Border Controller SIP Entity
- **Protocol:** Select **TCP**
- **Port:** Type **5060**. Avaya Aura™ Session Border Controller listens for SIP requests on TCP port 5060.
- **Trusted:** Check this box, otherwise SIP calls will be denied to/from Communication Manager

Click **Commit**.



4.6. Define Routing Policies

A routing policy describes the conditions under which calls will be routed to a particular SIP Entity. To add a routing policy for Communication Manager, select **Routing Policies** on the left pane then click **New**.



The **Routing Policy Details** screen is displayed. Fill in the following under **General**:

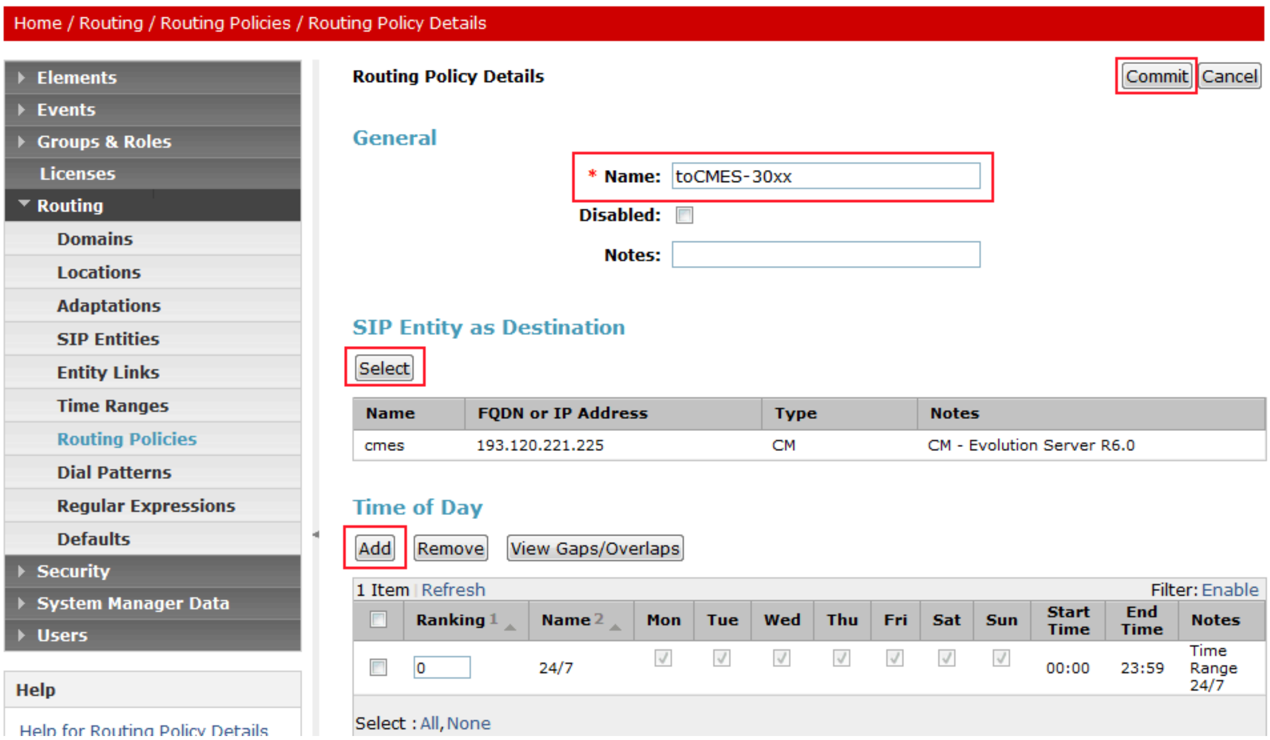
- **Name** Descriptive name.

Under **SIP Entity as Destination**:

- Click **Select**, and then select the Communication Manager SIP Entity to which this routing policy applies.

Under **Time of Day**:

- Click **Add**, and select the default **24/7** time range. Defaults can be used for the remaining fields. Click **Commit.**



Repeat the steps from the previous section. The **Routing Policy Details** screen is displayed. Fill in the following under **General**:

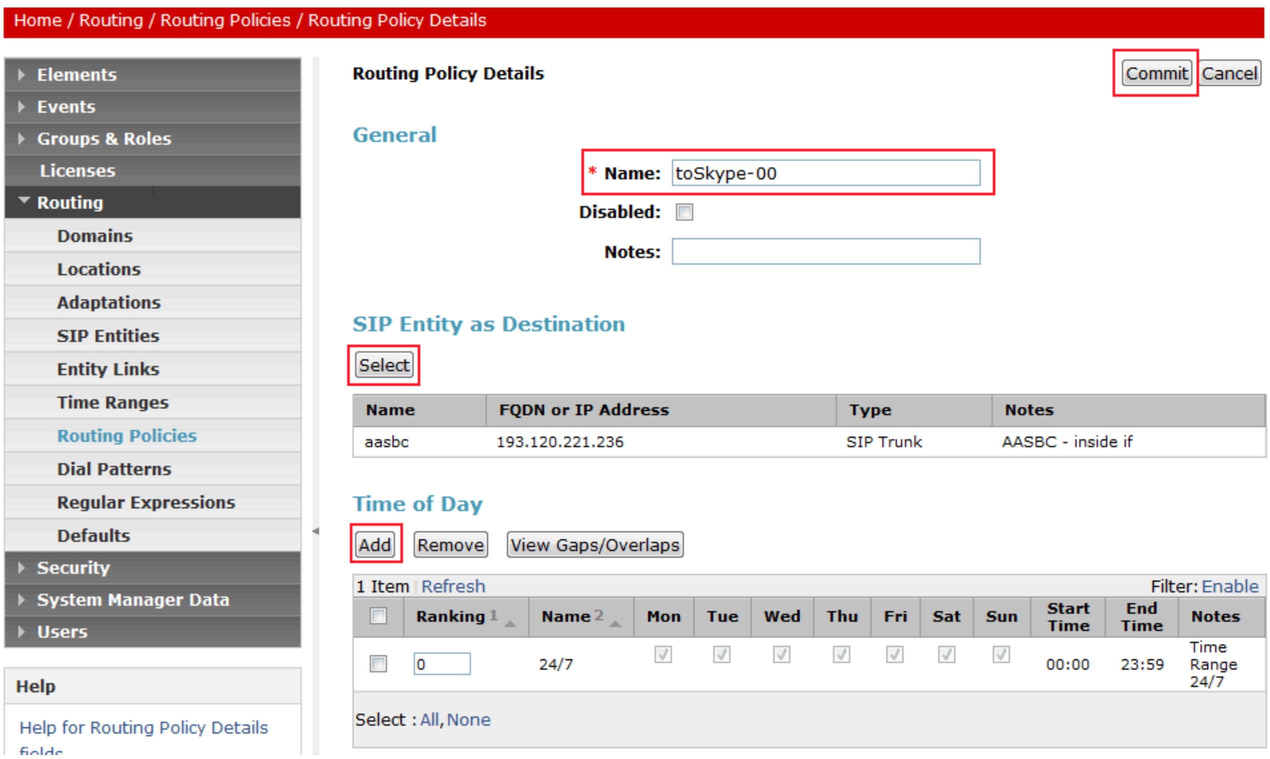
- **Name** Descriptive name.

Under **SIP Entity as Destination**:

- Click **Select**, and then select the Avaya Aura^{TM} Session Border Controller SIP Entity to which this routing policy applies.

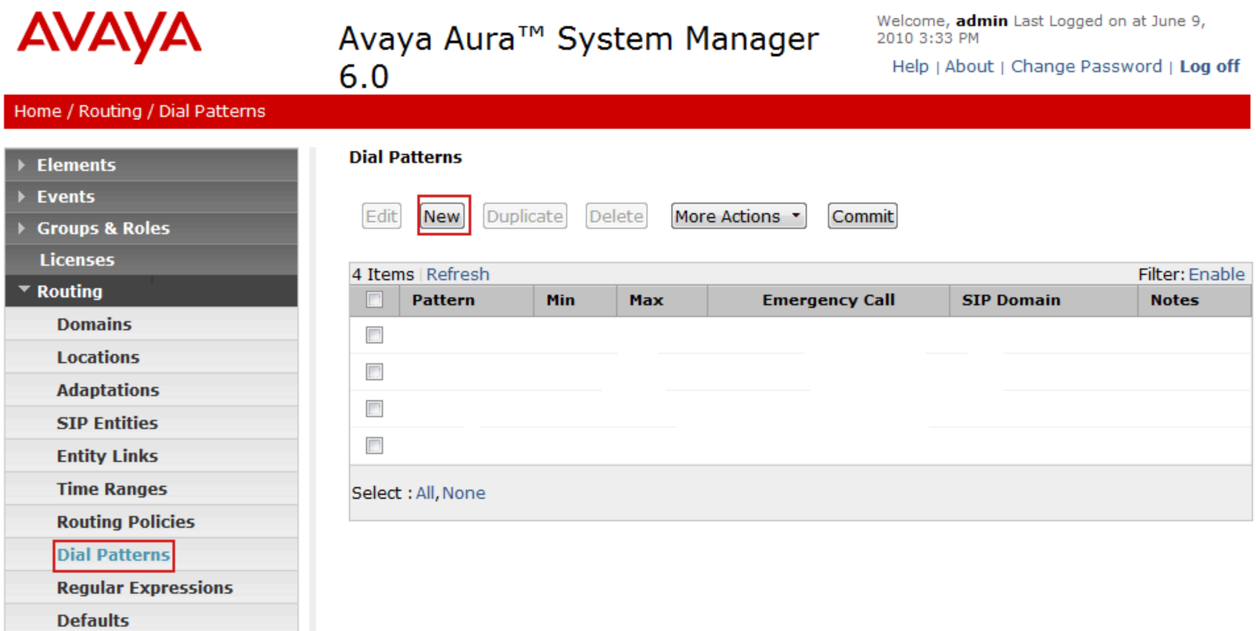
Under **Time of Day**:

- Click **Add**, and select the default **24/7** time range. Defaults can be used for the remaining fields. Click **Commit.**



4.7. Define Dial Patterns

Define dial patterns to direct calls to the appropriate SIP Entity. Calls to 4-digit extensions beginning with **30** should be routed to Communication Manager. To add a dial pattern, select **Dial Patterns** on the left pane then click **New**.



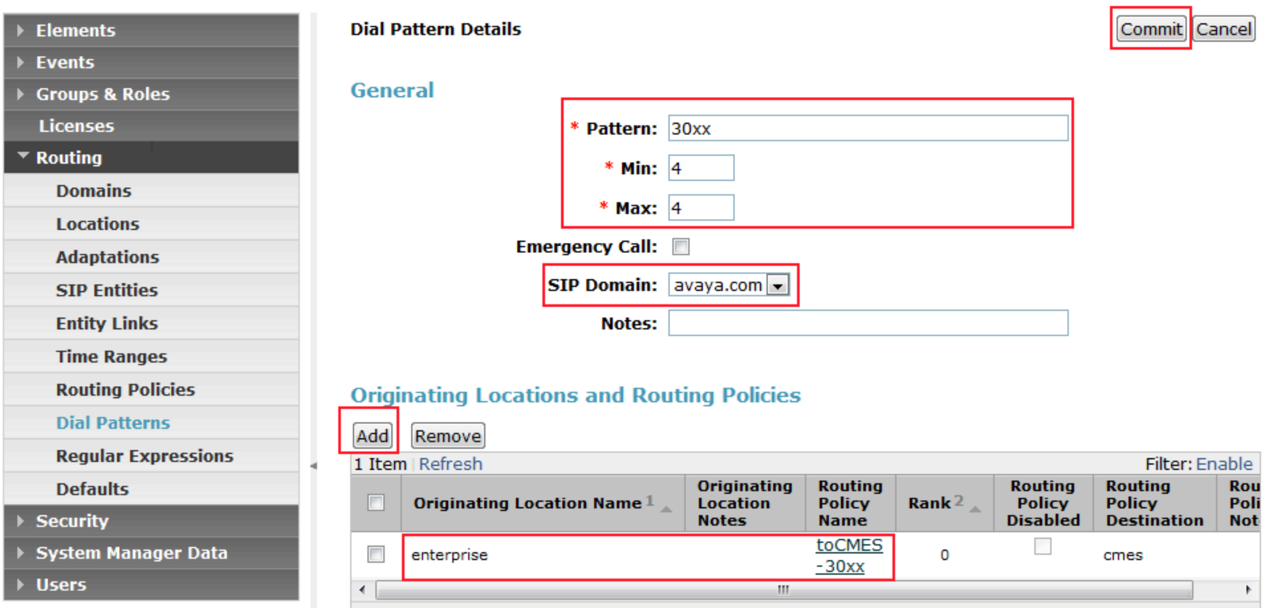
The **Dial Pattern Details** screen is displayed. Under **General** fill in the following fields:

- **Pattern:** Type **30xx** as four digit extensions are used in the sample configuration
- **Min:** Minimum length of dialled number. Type **4**
- **Max:** Maximum length of dialled number. Type **4**
- **SIP Domain:** Select the SIP domain specified in **Section 4.2**

Under **Originating Locations and Routing Policies**, click **Add**. Select the following entries:

- **Originating Location Name** Select the Location created in **Section 4.3**
- **Routing Policy Name** Select **toCMES-30xx**, the Routing Policy created in **Section 4.6**

Default values can be used for the remaining fields. Click **Commit** to save the dial pattern.



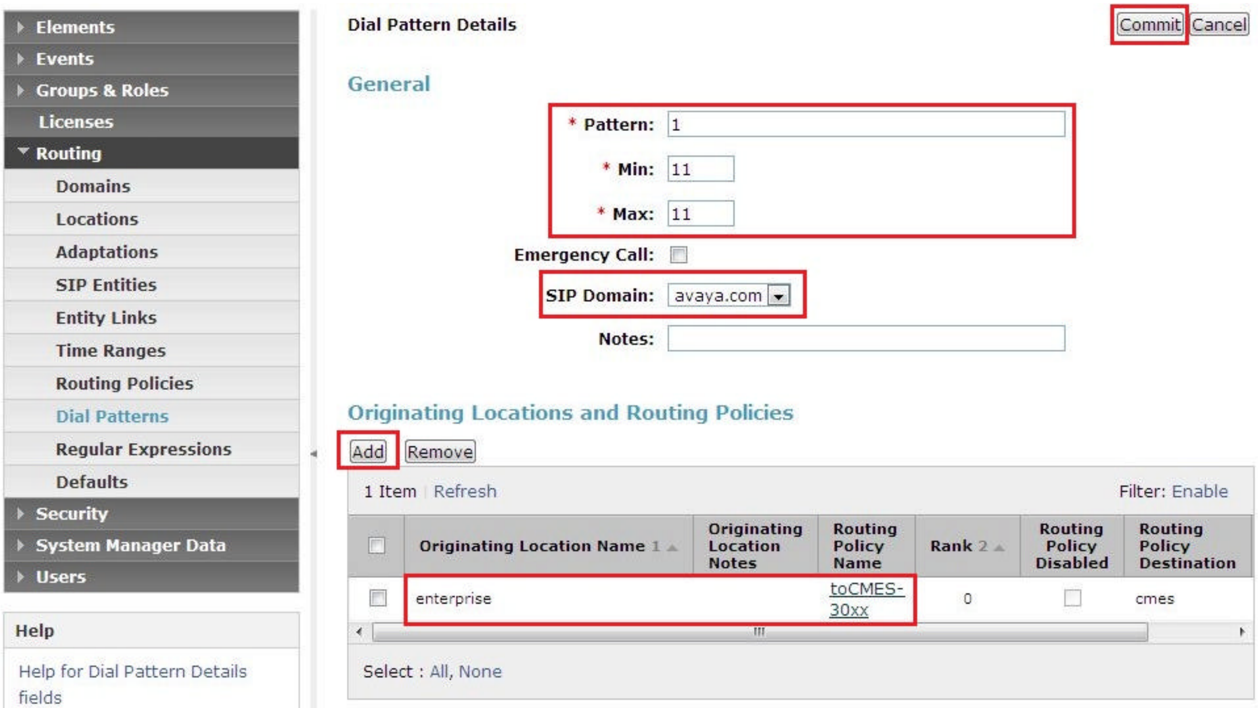
Repeat the steps from the previous section to add an inbound dial pattern from Skype Connect until the **Dial Pattern Details** screen is displayed.

- **Pattern:** Type **1** as the Online Number of the SIP Profile in **Section 5.12** is set to 12024703313.
- **Min:** Minimum length of dialled number. Type **11**
- **Max:** Maximum length of dialled number. Type **11**
- **SIP Domain:** Select the SIP domain specified in **Section 4.2**

Under **Originating Locations and Routing Policies,** click **Add**. Select the following entries:

- **Originating Location Name** Select the Location created in **Section 4.3**
- **Routing Policy Name** Select **toCMES-30xx**, the Routing Policy created in **Section 4.6**

Default values can be used for the remaining fields. Click **Commit** to save the dial pattern.



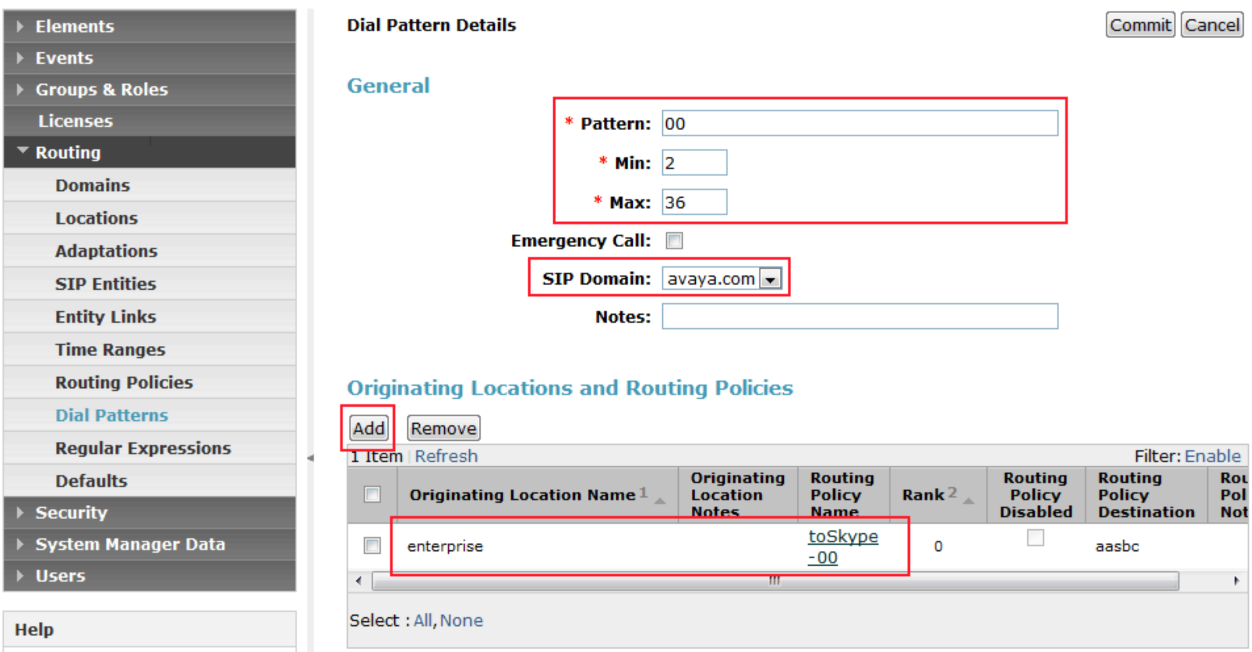
Repeat the steps from the previous section to add an outbound dial pattern to Skype Connect until the **Dial Pattern Details** screen is displayed.

- **Pattern:** Type **00** as the International numbering format is used in the sample configuration
- **Min:** Minimum length of dialled number. Type **2**
- **Max:** Maximum length of dialled number. Type **36**
- **SIP Domain:** Select the SIP domain specified in **Section 4.2**

Under **Originating Locations and Routing Policies,** click **Add**. Select the following entries:

- **Originating Location Name** Select the Location created in **Section 4.3**
- **Routing Policy Name** Select **toSkype-00** the Routing Policy created in **Section 4.6**

Default values can be used for the remaining fields. Click **Commit** to save the dial pattern.



5. Avaya Aura™ Communication Manager

This section provides the procedures for configuring Communication Manager and includes the following items:

- Administer System Parameters
- Administer Dial Plan
- Administer IP Network Parameters
- Administer Signaling Group
- Administer Trunk Group
- Administer Automatic Route Selection
- Administer Route Pattern
- Administer Public Unknown Numbering
- Administer Incoming Call Handling Treatment

The steps are performed from the Communication Manager System Access Terminal (SAT) interface. These Application Notes assume that basic Communication Manager administration, including stations, Processor Ethernet, etc, has already been performed.

5.1. Administer System Parameters

This section reviews the Communication Manager licenses and features that are required for the reference configuration described in these Application Notes. For required licenses that are not enabled in the steps that follow, contact an authorized Avaya account representative to obtain the licenses. Enter the **display system-parameters customer-options** command. On Page 2 of the **system-parameters customer-options** form, verify that the **Maximum Administered SIP Trunks** number is sufficient for the number of expected SIP trunks.

```
Display system-parameters customer-options                    Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 8000  0
           Maximum Concurrently Registered IP Stations: 18000 4
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 5000  250
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 10    1
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   2
   Maximum Number of Expanded Meet-me Conference Ports: 0     0

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

On Page 4 of the **system-parameters customer-options** form, verify that the **IP Trunks** field in the following screenshot is set to **y**.

```
display system-parameters customer-options                    Page   4 of  11
                                OPTIONAL FEATURES

   Emergency Access to Attendant? Y                               IP Stations? Y
           Enable 'dadmin' Login? Y
           Enhanced Conferencing? Y                         ISDN Feature Plus? Y
                  Enhanced EC500? Y     ISDN/SIP Network Call Redirection? N
    Enterprise Survivable Server? N                           ISDN-BRI Trunks? Y
       Enterprise Wide Licensing? N                                  ISDN-PRI? Y
              ESS Administration? N              Local Survivable Processor? N
          Extended Cvg/Fwd Admin? Y                      Malicious Call Trace? N
     External Device Alarm Admin? N                  Media Encryption Over IP? N
  Five Port Networks Max Per MCC? N   Mode Code for Centralized Voice Mail? N
                Flexible Billing? N
   Forced Entry of Account Codes? N                  Multifrequency Signaling? Y
      Global Call Classification? N        Multimedia Call Handling (Basic)? Y
             Hospitality (Basic)? Y     Multimedia Call Handling (Enhanced)? Y
 Hospitality (G3V3 Enhancements)? N              Multimedia IP SIP Trunking? N
                       IP Trunks? Y

           IP Attendant Consoles? N
```

5.2. Administer Dial Plan

Enter the **change dialplan analysis** command to provision the dial plan. Note the following dialed strings administered below:

- 3-digit dial access codes (indicated with a **Call Type** of **dac**) beginning with * – Trunk Access Codes (TACs) defined for trunk groups in this reference configuration conform to this format.
- 4-digit extensions with a **Call Type** of **ext** beginning with the digits **30** – local extensions for Communication Manager stations in this reference configuration conform to this format.
- 1-digit facilities access code (indicated with a **Call Type** of **fac**) beginning with the digit **9** – access code for outbound ARS dialing.

```
Change dialplan analysis                                      Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location:  all         Percent Full:    2
      Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
      String   Length Type    String   Length Type     String   Length Type
    *            3    dac
    30           4    ext
    9            1    fac
```

5.3. Administer IP Network Parameters

These Application Notes assume that the appropriate IP network regions and IP codec sets have already been administered to support internal calls. For simplicity in this reference configuration, all Communication Manager elements – stations, PE interface, G450 Media Gateway – within the Avaya site are assigned to a single IP network region (region 1) and all internal calls use a single IP codec set. This section describes the steps for administering an additional IP network region to represent the Skype Connect service, and another IP codec set for external calls.

Enter the **change ip-codec-set x** command, where **x** is the number of an unused IP codec set (e.g. **2**). This IP codec set will be used for off-net calls to Skype Connect. On Page 1 of the **ip-codec-set** form, provision the codecs in the order shown below.

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

                          IP Codec Set
    Codec Set: 2

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

On Page 2 of the **ip-codec-set** form, set **FAX Mode** to **off**.

```
Change ip-codec-set 2                                          Page   2 of   2
                          IP Codec Set
                              Allow Direct-IP Multimedia? N
                Mode                Redundancy
    FAX         off                  0
    Modem       off                  0
    TDD/TTY     off                  0
    Clear-channel   n                0
```

Enter the **change ip-network-region 1** command. This IP network region is used for on-net calls. Enter **avaya.com** for the **Authoritative Domain** parameter.

```
Change ip-network-region 1                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location:                Authoritative Domain:avaya.com
    Name: Skype
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
```

Enter the **change ip-network-region 2** command. This IP network region will be used to represent the SIP Trunk to Skype Connect. Enter **2** for the **Codec Set** parameter.

```
Change ip-network-region 2                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 2
Location:          Authoritative Domain:
    Name: Skype
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              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
```

On Page 4 of the form enter **2** for the **Codec Set** parameter.

```
change ip-network-region 2                                     Page   4 of  19

 Source Region: 2    Inter Network Region Connection Management     I       M
                                                                     G  A    e
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    a
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    s
 1    2     y    NoLimit                                             n
```

5.4. Administer Signaling Group

This section describes the steps for administering a single signaling group, shared by both outbound and inbound calls. Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group (e.g. **100**), and provision the following:

- **Group Type** Set to **sip**
- **Transport Method** Set to **tcp**
- **Near-end Node Name** Set to the node name of the Processor Ethernet interface
- **Far-end Node Name** Set to the node name of Session Manager
- **Near-end Listen Port** Set to **5060**
- **Far-end Listen Port** Set to **5060**
- **Far-end Network Region** Set to IP network region **2**

```
Add signaling-group 100
                                SIGNALING GROUP

 Group Number: 100              Group Type: sip
                        Transport Method: tcp
  IMS Enabled? N

   Near-end Node Name: procr          Far-end Node Name: sm100
 Near-end Listen Port: 5060               Far-end Listen Port: 5060
                                        Far-end Network Region:2
Far-end Domain:
                                                 Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                      RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload              Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                         IP Audio Hairpinning? n
         Enable Layer 3 Test? Y                         Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 6
```

5.5. Administer Trunk Group

Enter the **add trunk-group x** command, where **x** is the number of an unused trunk group (e.g. **100**). On Page 1 of the **trunk-group** form, provision the following:

- **Group Type** Set to **sip**
- **Group Name** Enter a descriptive name
- **TAC** Enter a trunk access code that is consistent with the dial plan
- **Service Type** Set to **public-ntwrk**
- **Signaling Group** Set to the number of the signaling group administered in the previous step
- **Number of Members** Enter the maximum number of simultaneous calls permitted on this trunk group

```
Add trunk-group 100                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 100                   Group Type: sip          CDR Reports: y
  Group Name: SkypeConnect             COR: 1        TN: 1        TAC: *22
   Direction: both       Outgoing Display? n
 Dial Access? n                                  Night Service:
                                    Auth Code? n
Service Type: public-ntwrk
                                                       Signaling Group: 52
                                                     Number of Members: 20
```

On Page 3 of the **Trunk Group** form:

- Set **Numbering Format:** to **public**

```
add trunk-group 100                                         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
```

5.6. Administer Automatic Route Selection

Use the **change feature-access-codes** command to specify **9** as the access code for external dialing.

```
Change feature-access-codes                                   Page   1 of   9
                               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:

      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:
                         Call Park Access Code:
                       Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
            Conditional Call Extend Activation:        Deactivation:
                   Contact Closure   Open Code:          Close Code:
```

Use the **change ars analysis** command to specify the called number patterns which are dialed following the ARS access code. In the reference configuration, outbound calls are placed to international numbers.

```
Display ars analysis 0                                    Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    0
          Dialed           Total     Route     Call   Node   ANI
          String          Min  Max  Pattern    Type   Num    Reqd
    011                   10   18    100        intl          n
```

5.7. Administer Route Pattern

Use the **change route-pattern** command to define the trunk group administered in **Section 5.5** included in the route pattern that ARS selects.

```
change route-pattern 100                                          Page   1 of   3
                    Pattern Number: 68  Pattern Name:
                             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: 100   0                                                               n   user
```

5.8. Administer Public Unknown Numbering

For Calling Line Identification Presentation (CLIP) to work on outbound calls, Skype Connect expects to receive one of the online or landline numbers associated with the SIP Profile. The calling number is converted to an E.164 format by the public-unknown-numbering table and is inserted into the From and PAI headers of the outgoing INVITE request. Enter the **change public-unknown-numbering 0** command to specify the calling party numbers that are to be sent to the PSTN through Skype Connect. In the **public-unknown-numbering** form, for each local extension assigned to Avaya Aura™ Communication Manager provision an entry as follows:

- **Ext Len** Enter the total number of digits in the local extension range.
- **Ext Code** Enter enough leading digits to identify the local extension or extension range.
- **Trk Grp(s)** Enter the number of the outbound trunk group to Skype Connect.
- **CPN Prefix** Enter the online or Skype Connect verified landline numbers that are associated with the SIP Profile in Skype Manager. In **Section 6.6** the Caller ID is set to +12024703313 in Skype Manager.
- **CPN Len** Enter the total number of digits to be sent to Session Manager.

```
change public-unknown-numbering 0                             Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                          Total
Ext Ext              Trk       CPN            CPN
Len Code             Grp(s)    Prefix         Len
                                                   Total Administered: 2
                                                   Maximum Entries: 9999
 5  3001            100        12024703313     11
```

5.9. Administer Incoming Call Handling Treatment

In the reference configuration Skype Connect delivers 11 digit Online Numbers to the enterprise network. Avaya Aura™ Communication Manager converts the incoming 11 digit Online Number to a 4 digit extension using the Incoming Call Handling Treatment table. In **Section 5.12** the Online Number is set to +12024703313 in Skype Manager.

```
change inc-call-handling-trmt trunk-group 100                  Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT

 Service/      Number   Number      Del Insert
 Feature        Len      Digits

 public-ntwrk    11 12024703313      11  3001
```

6. Configure Skype Manager

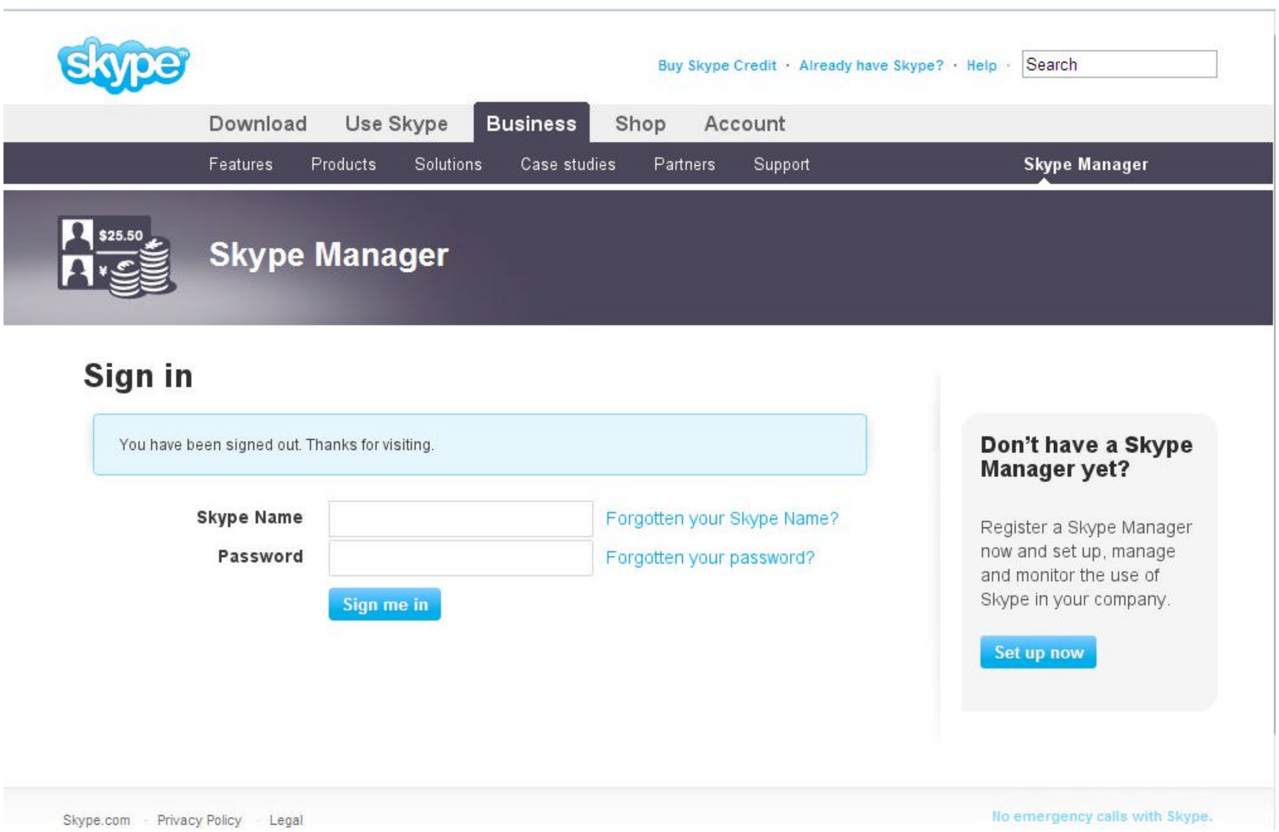
This section provides the procedures for configuring Skype Manager and includes the following items:

- Log in to Skype Manager
- Create SIP Profile
- Administer Authentication Method
- Administer Maximum Simultaneous Calls
- Administer Online Numbers
- Administer Caller ID
- Create a Business User
- Add Business User to SIP Profile

These Application Notes assume that the Skype Manager account has enough credit allocated to create a new SIP Profile and associate Online Numbers with it.

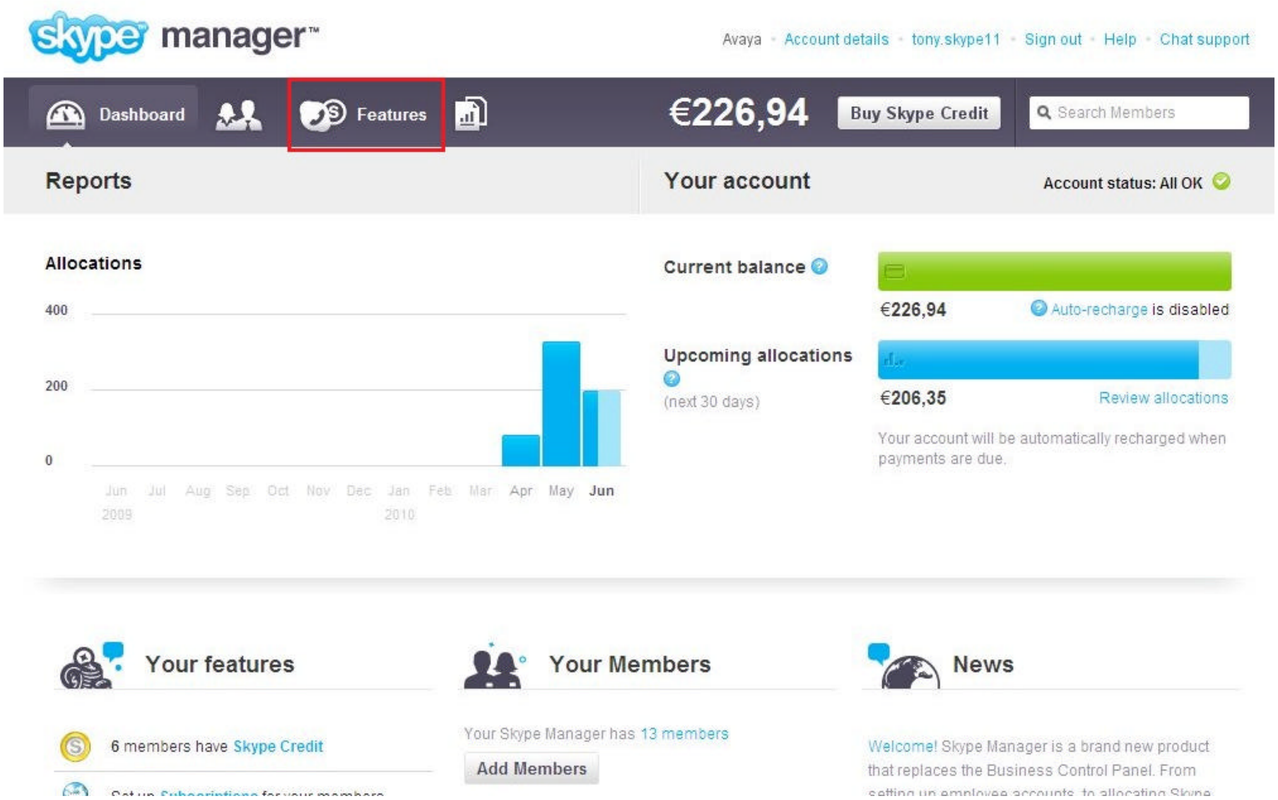
6.1. Log in to Skype Manager

Configuration of a new SIP Profile and Business Users is accomplished by accessing the browser-based GUI of Skype Manager at **http://manager.skype.com**. Log in with the appropriate credentials.

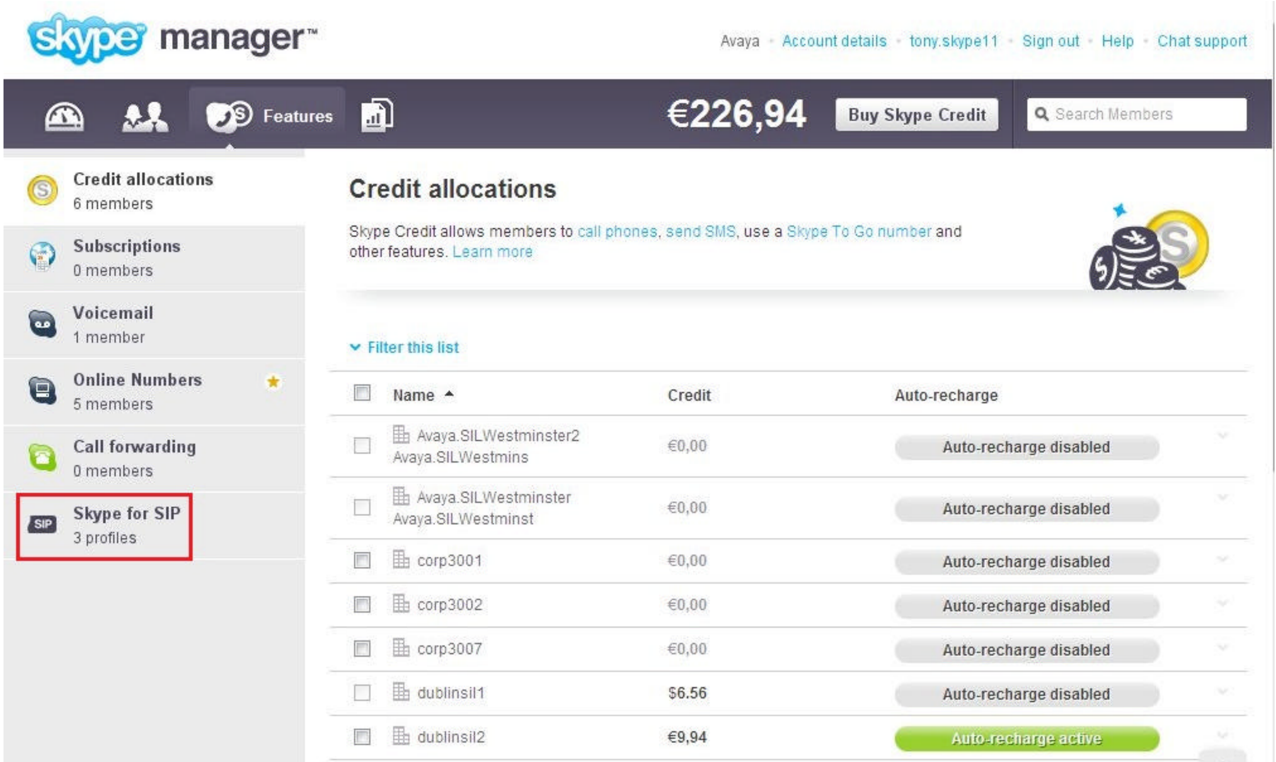


6.2. Create SIP Profile

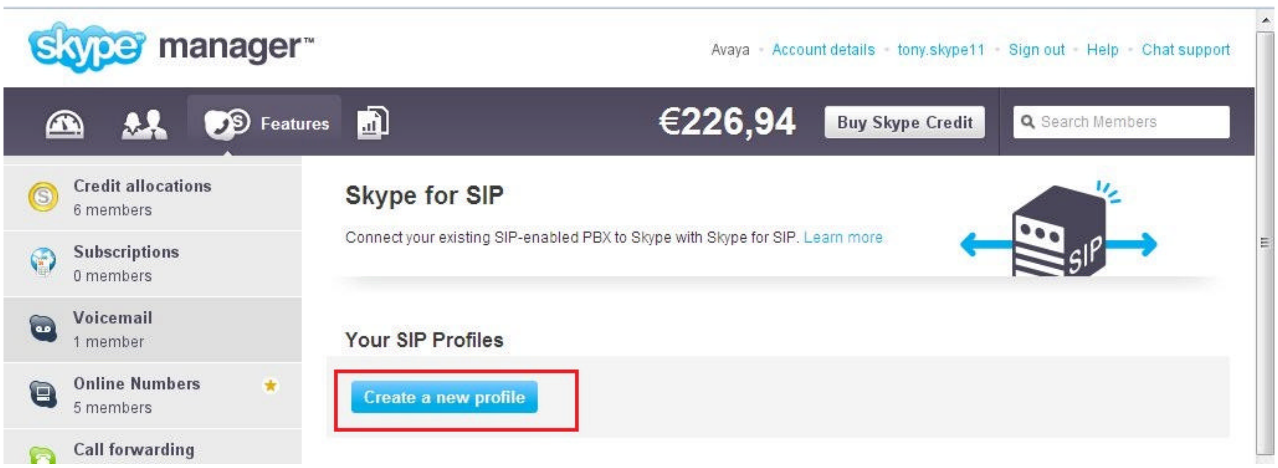
The **Dashboard** page is displayed. Verify that enough credit is available in your company's account to create a new SIP Profile. The available total credit is shown on the toolbar below. Click **Features**.



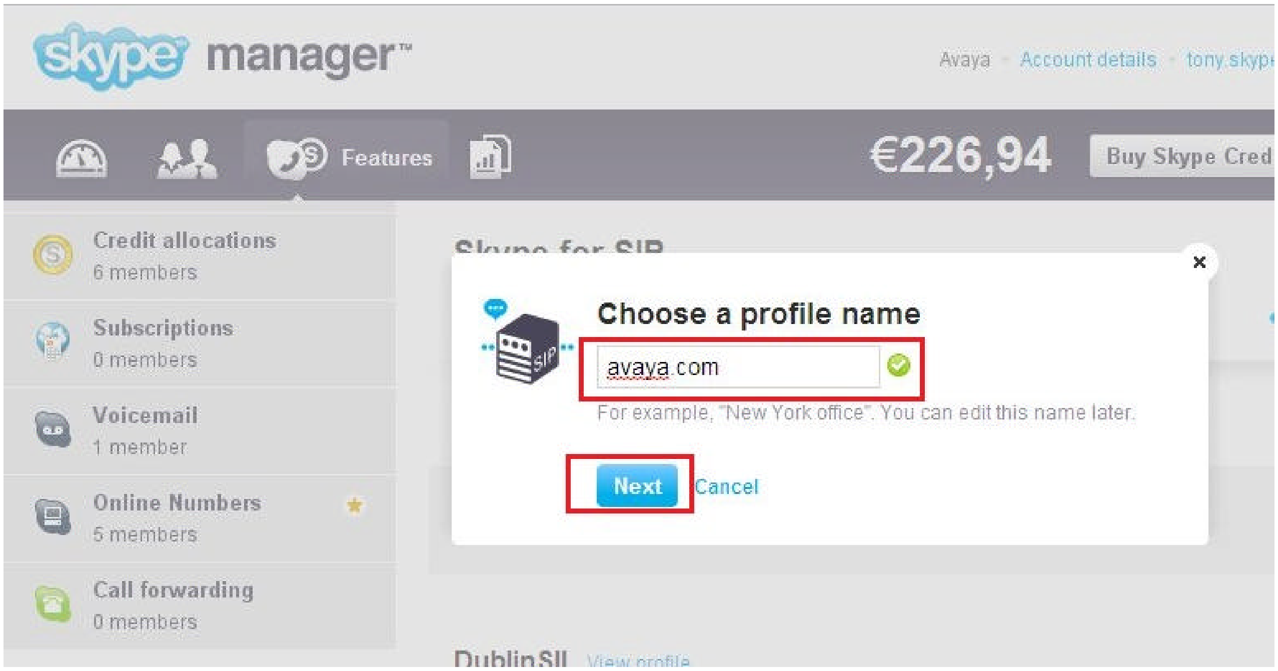
The **Features** page is displayed. Click **Skype for SIP** on the left pane.



The **Skype for SIP** page is displayed. Click **Create a new profile**.

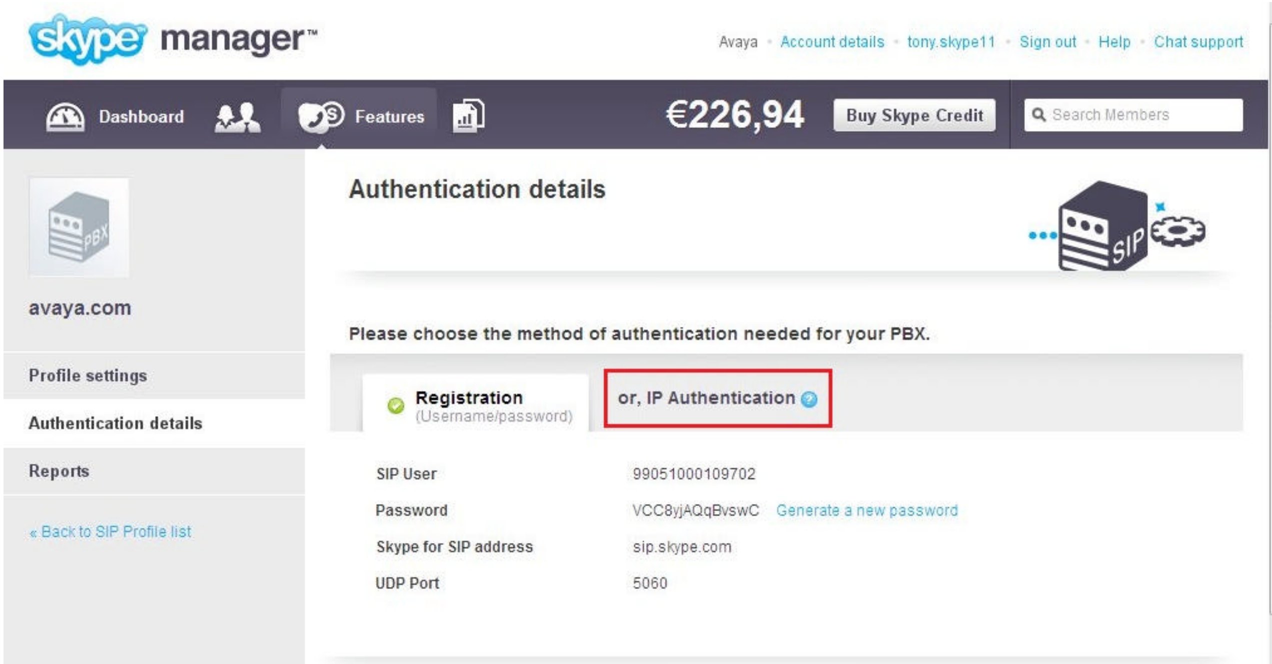


The **Choose a profile name** pop-up window is displayed. Type the name of your SIP Profile and click **Next**.

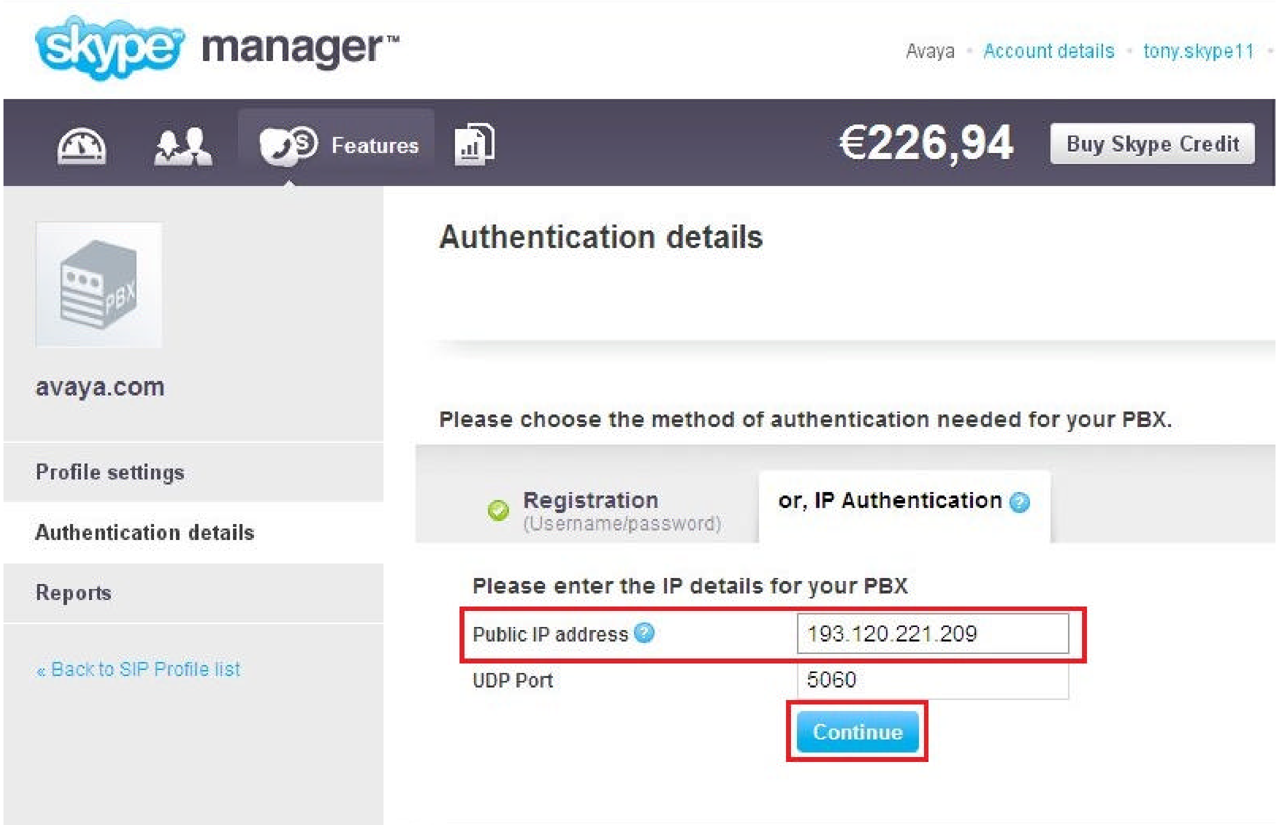


6.3. Administer Authentication Method

The **Authentication details** page is displayed. Click the **IP Authentication** tab.



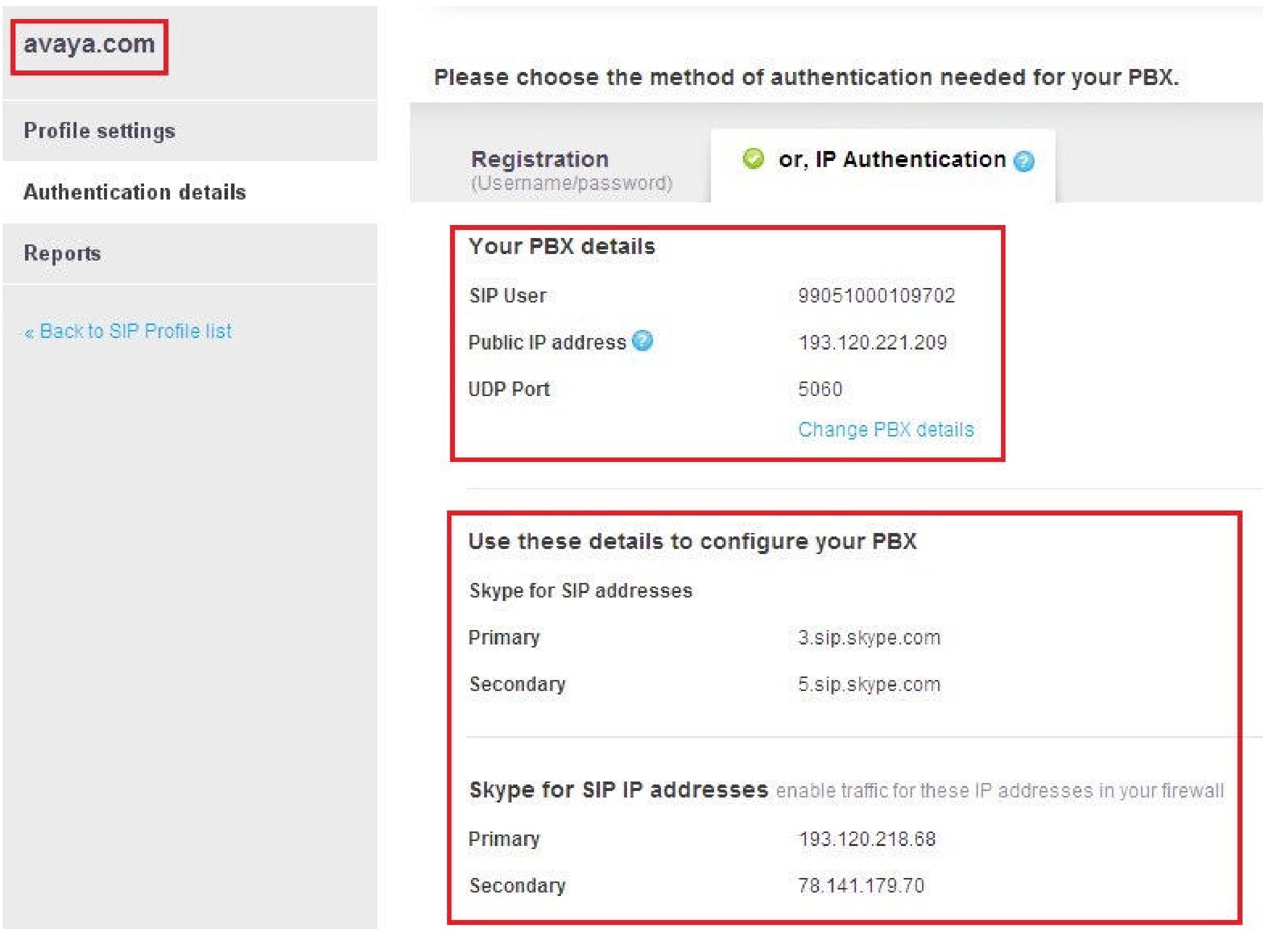
The IP Authentication page is displayed. In the **Public IP address** field, type the IP address of the Avaya AuraTM Session Border Controller's outside interface (eth2), then click **Continue**.



Skype Manager automatically generates a unique SIP User and associates it with the newly created **avaya.com** SIP Profile. Calls originating from the Avaya Aura™ Session Border Controller must include the SIP User and the sip.skype.com domain in the From header. Make a note of the SIP User name before proceeding to the next step. In the sample configuration the From header of an outgoing INVITE request from the Avaya Aura™ Session Border Controller is displayed as follows:

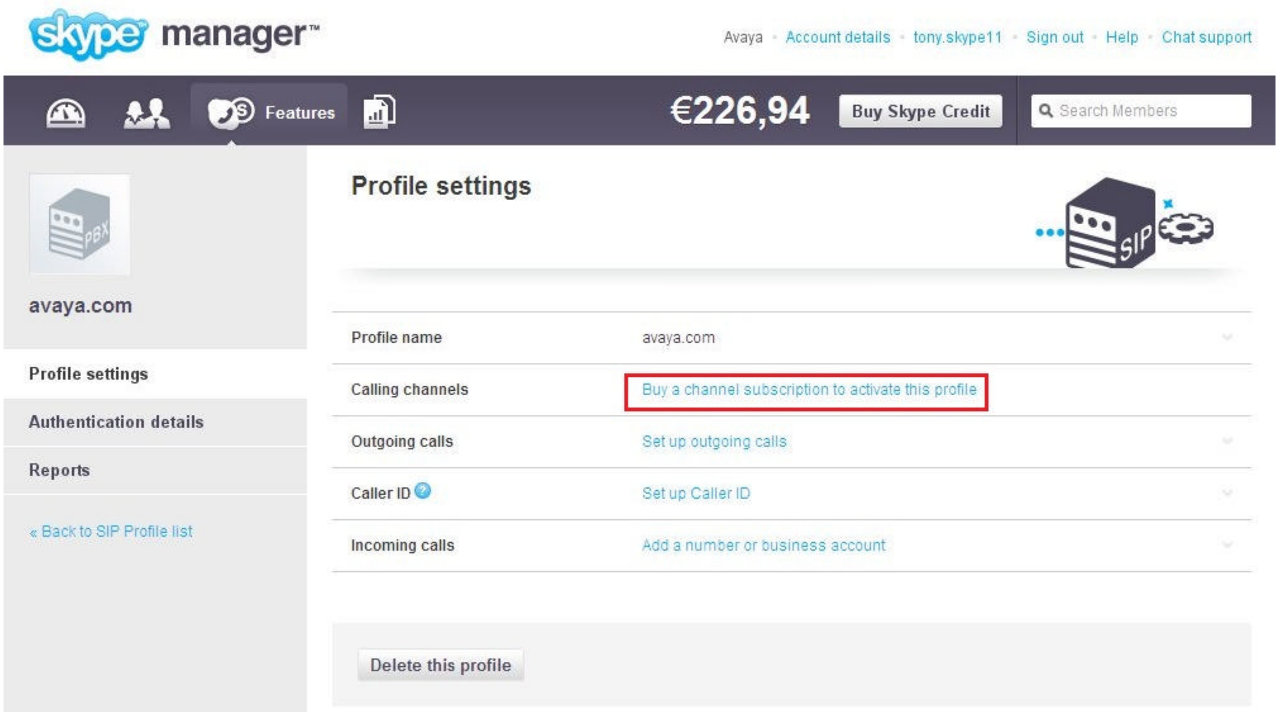
From: "H323-1608-3002" **<sip:99051000109702@sip.skype.com>**;tag=ecdd78c1-13c4-4c04c659-22e15675-7a5bc12c

Skype Manager allocates a Primary and Secondary **Skype for SIP address**. Click **Profile settings**.

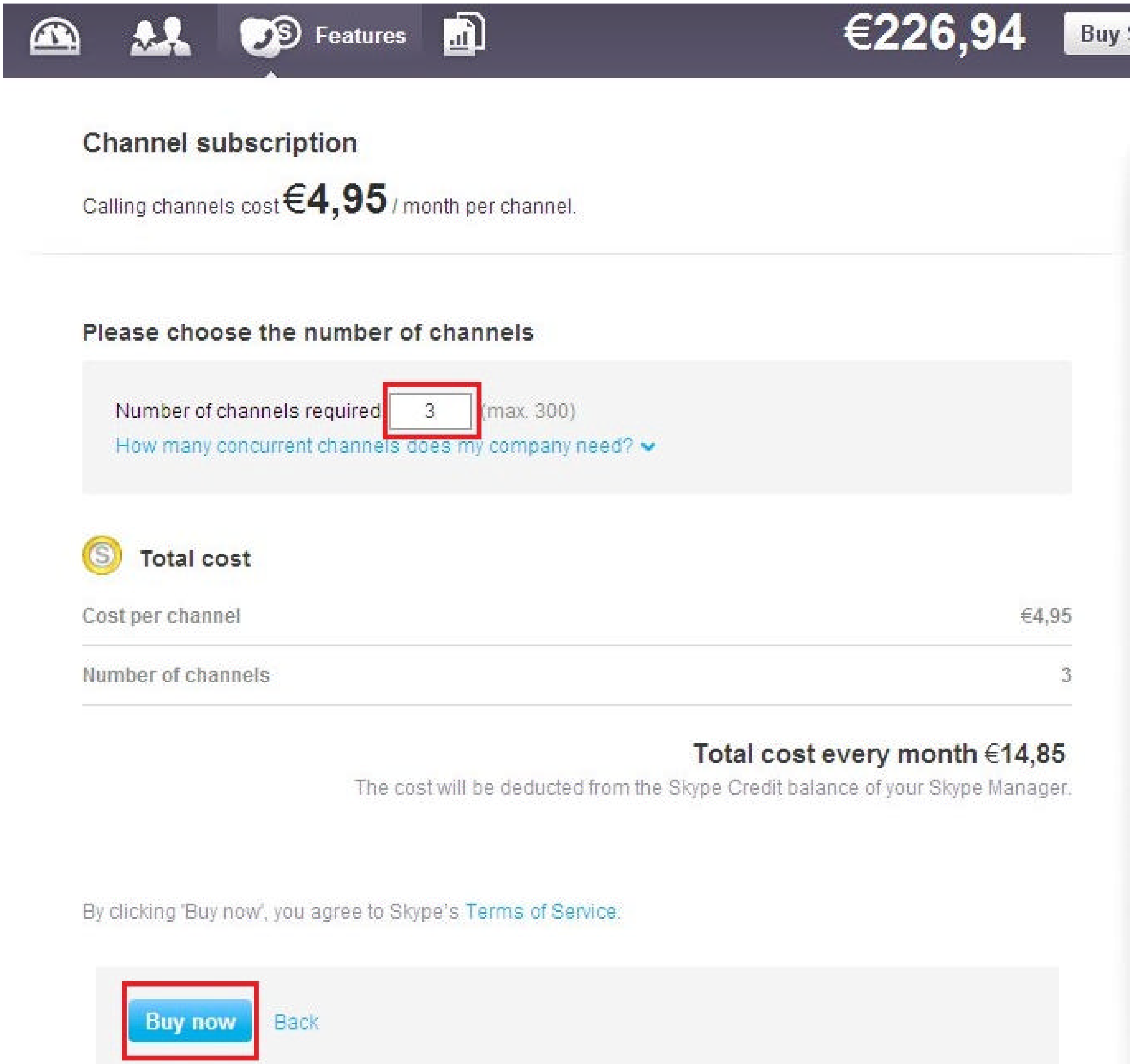


6.4. Administer Maximum Simultaneous Calls

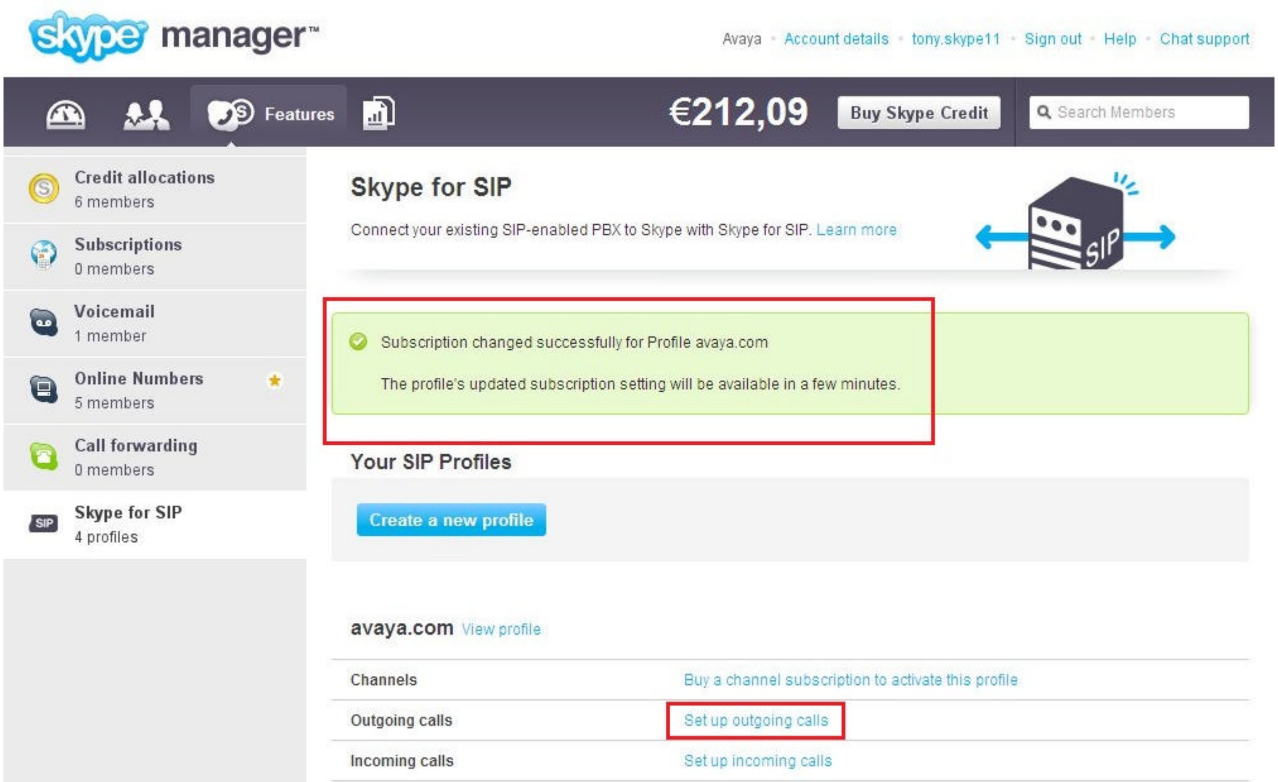
The **Profile settings** page is displayed. Click **Buy a channel subscription to activate this profile**.



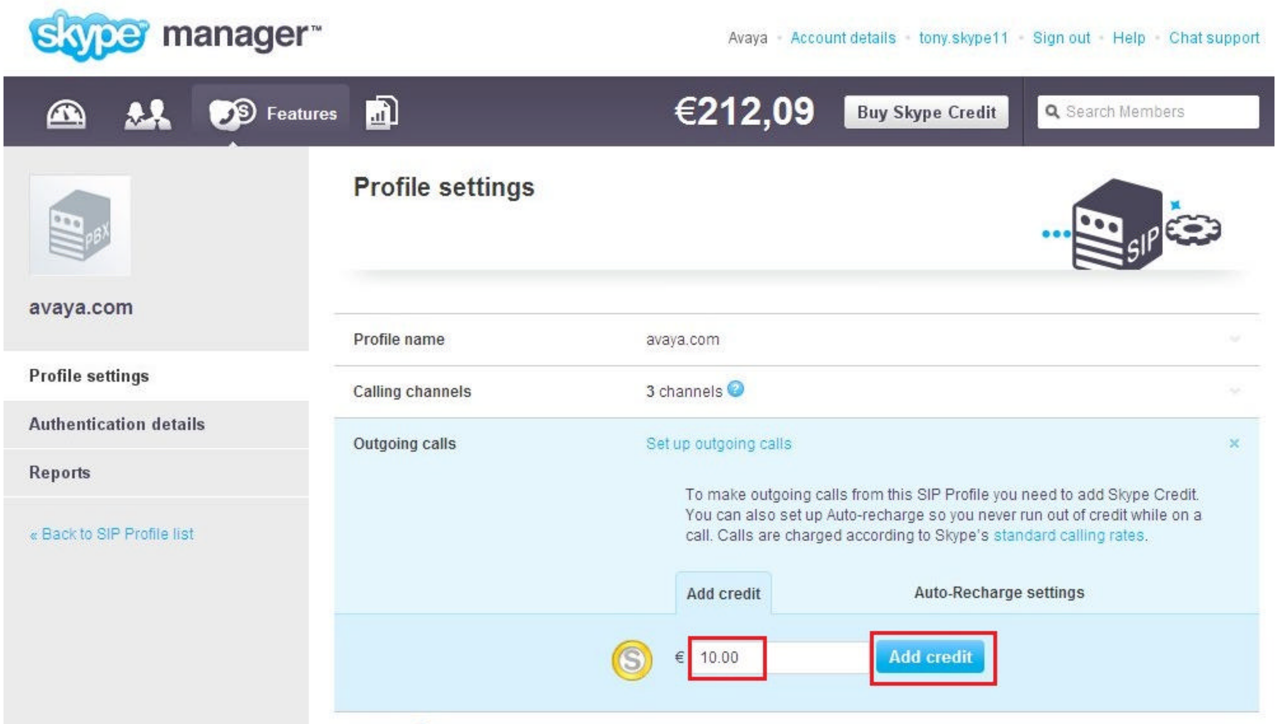
The **Channel subscription** page is displayed. Type the maximum number of simultaneous calls that your business users are expected to make. Click **Buy now**.



The below status message is displayed to confirm that the channel subscription was successful. Click **Set up outgoing calls**.

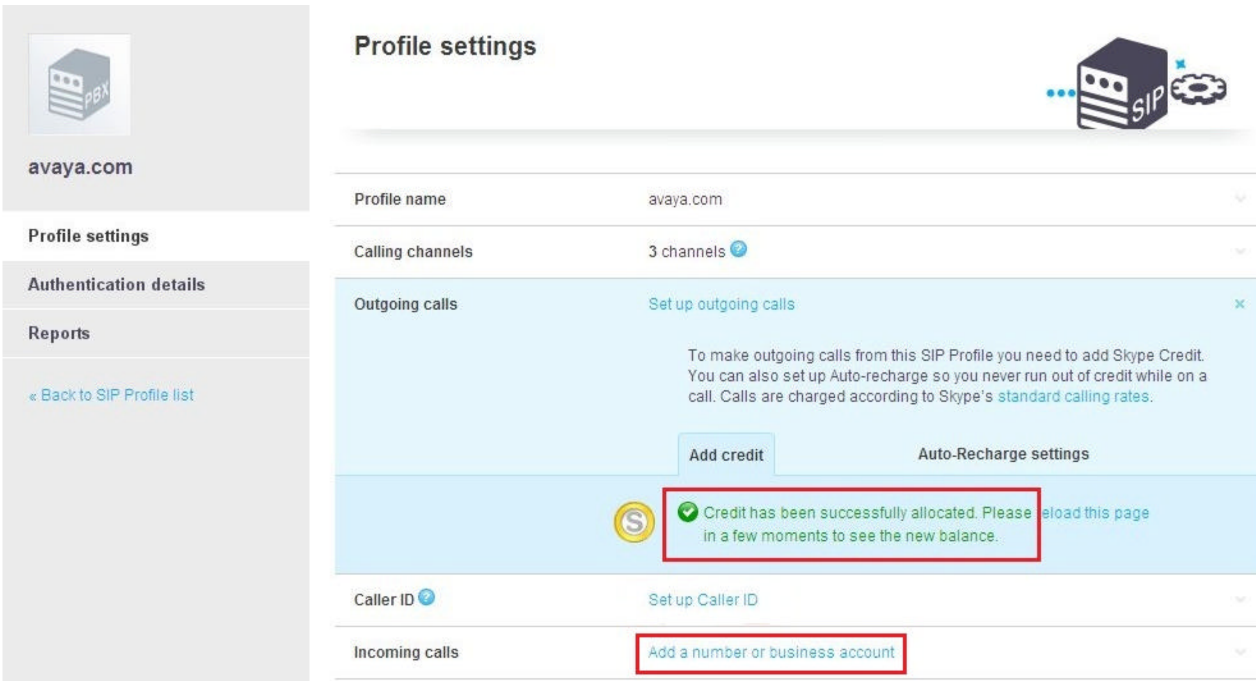


The **Profile settings** page is displayed. On the **Add credit** tab type the amount of credit you'd like to allocate for outbound calls. Note that calls to other SIP Profiles or Online Numbers of Skype Clients on the P2P network are free of charge.
Click **Add credit**.

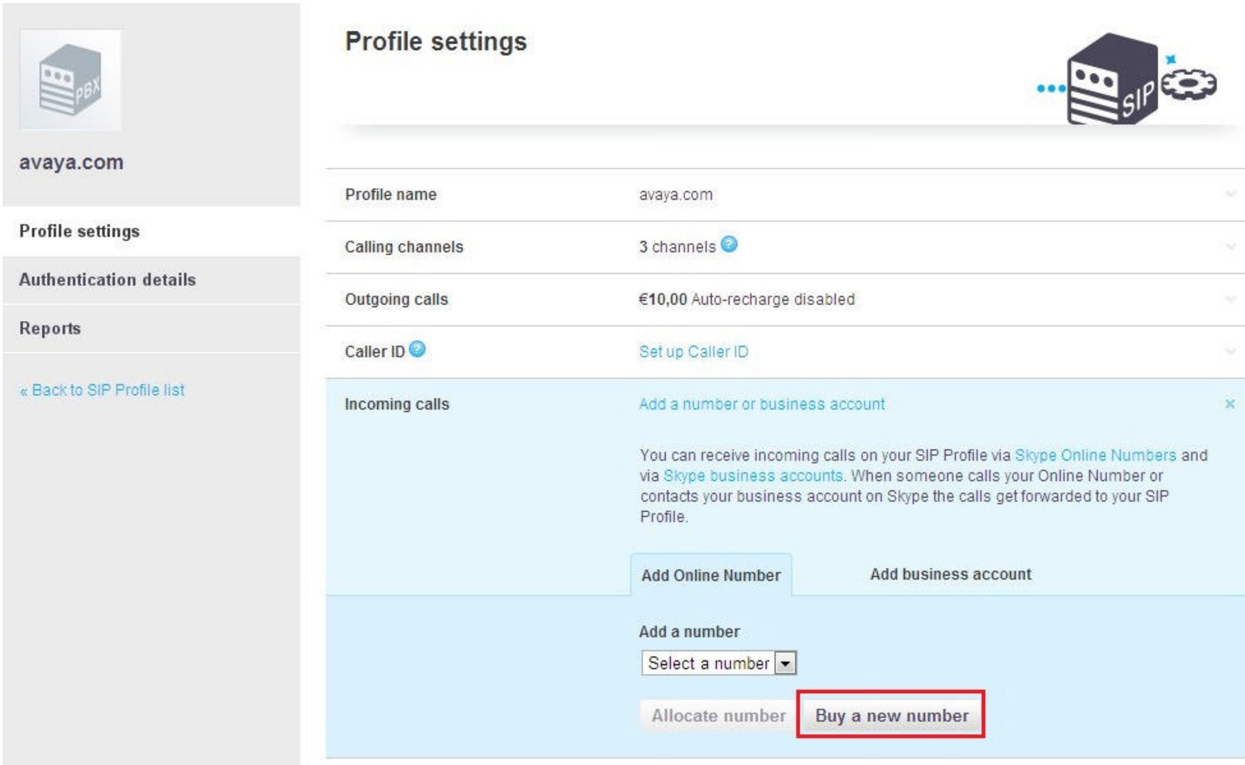


6.5. Administer Online Numbers

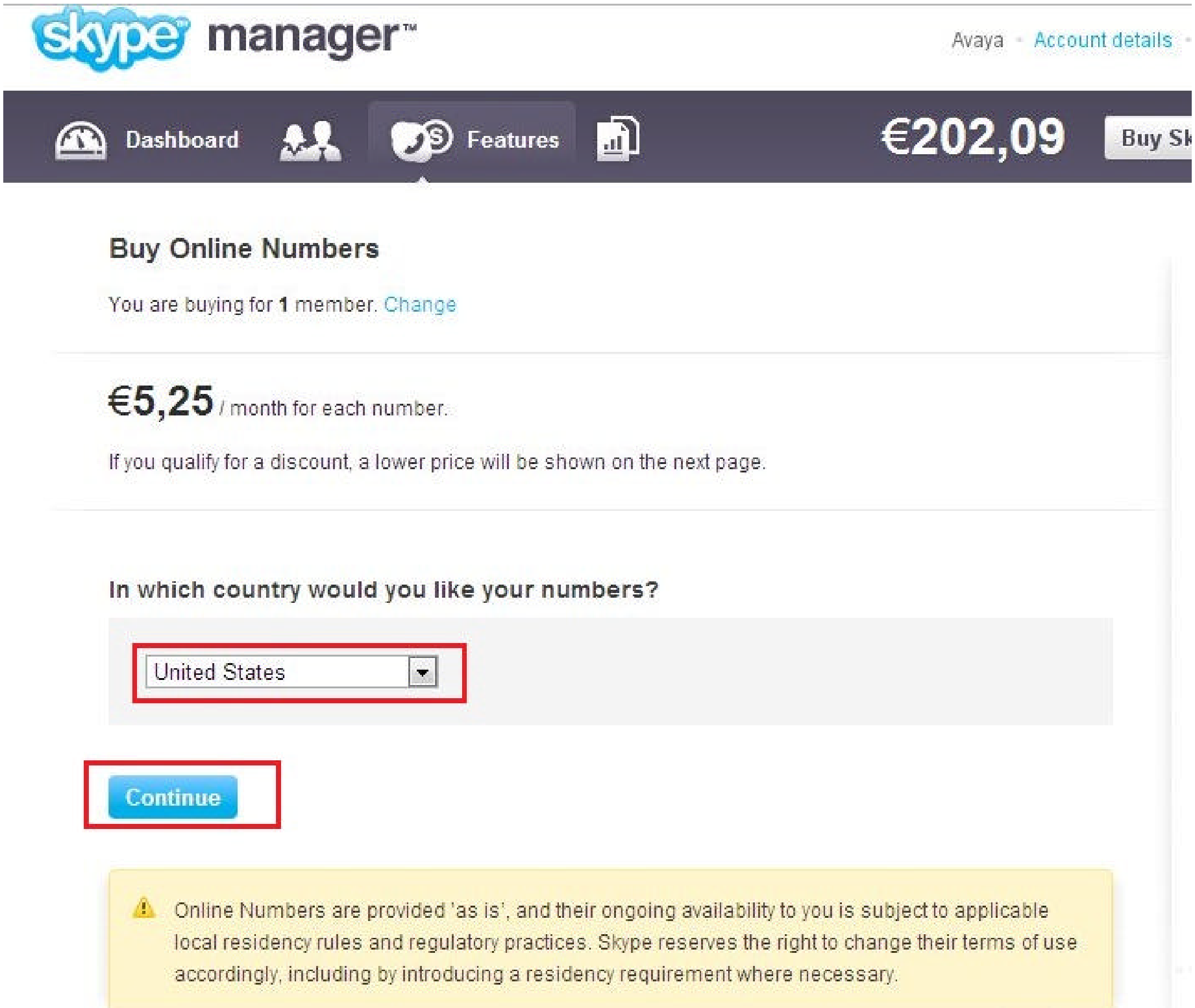
Online Numbers provide the same functionality as DID/DDIs in a traditional telephony environment. The below confirmation message is displayed once the credit allocation for outbound calls is complete. Click **Add a number or business account**.



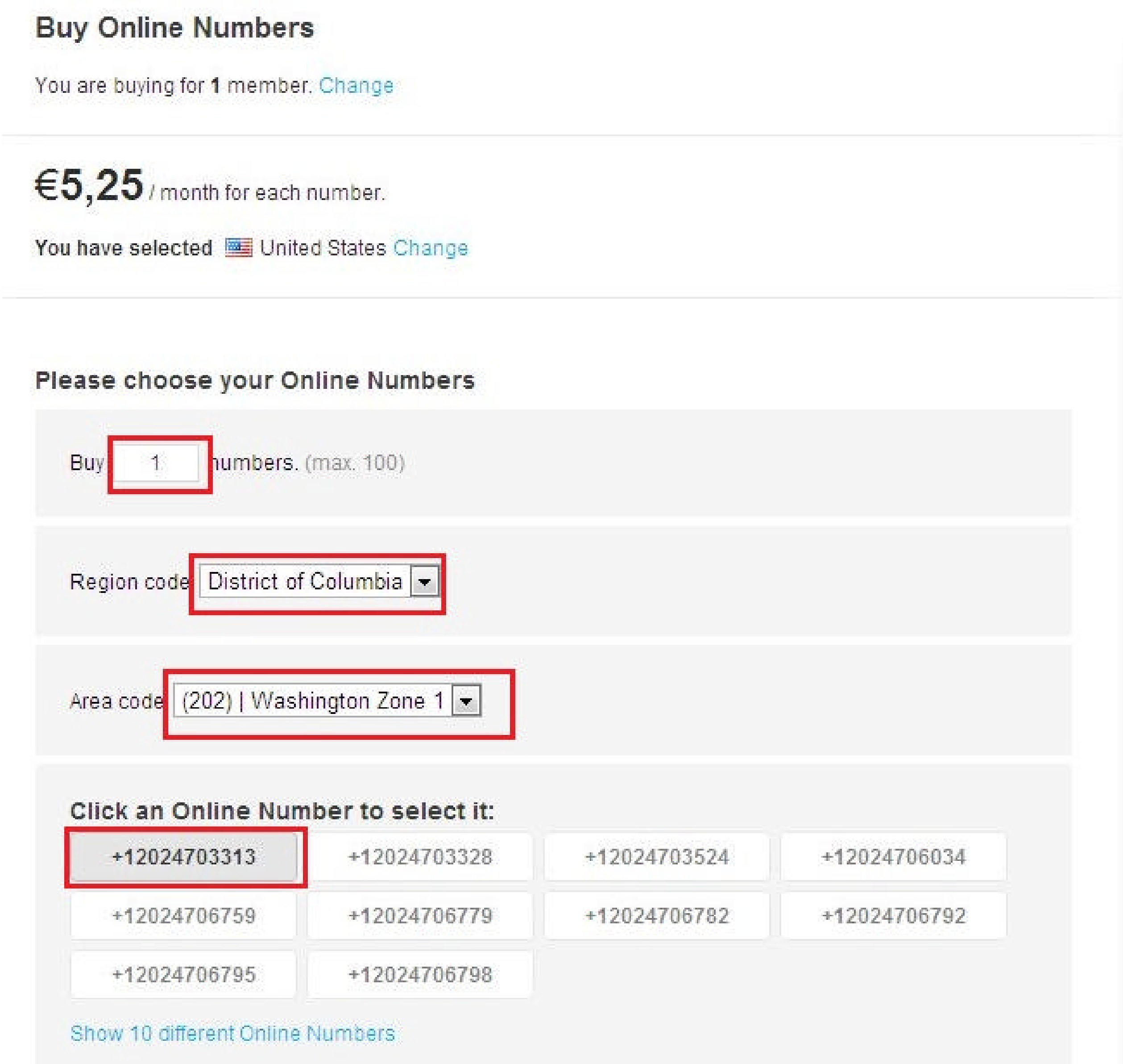
The **Incoming calls** tab is expanded. Click **Buy a new number**.



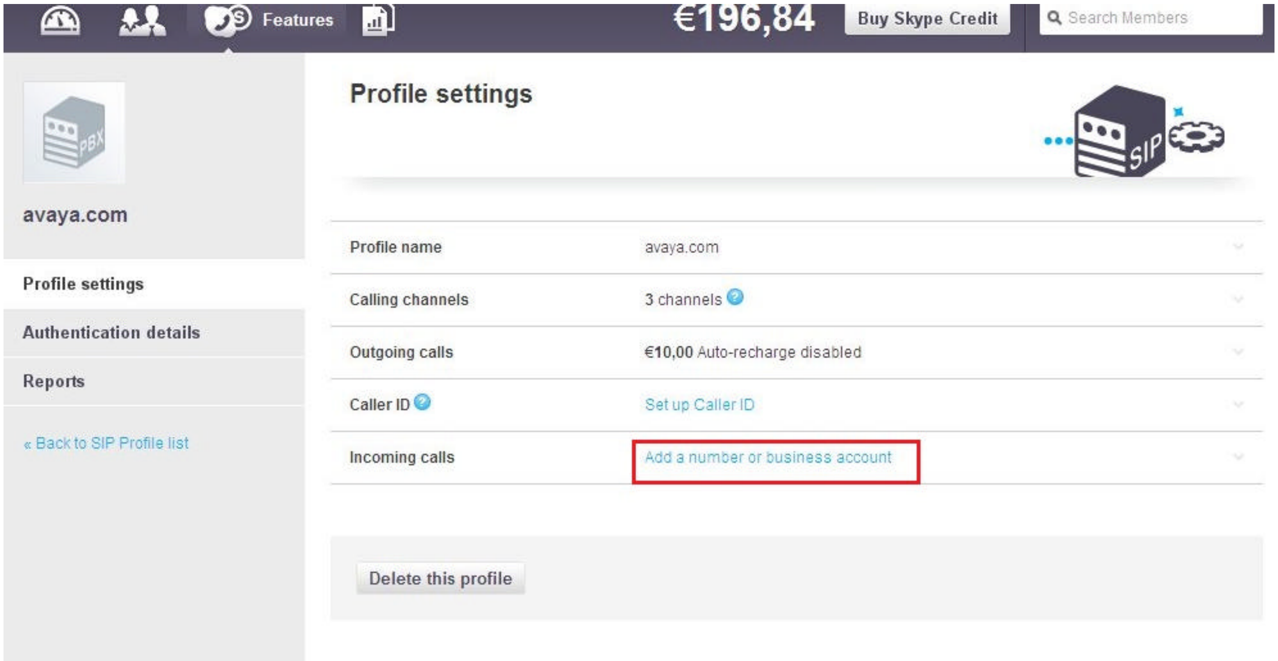
The **Buy Online Numbers** page is displayed. Select **a country** from the drop-down list box and click **Continue**.



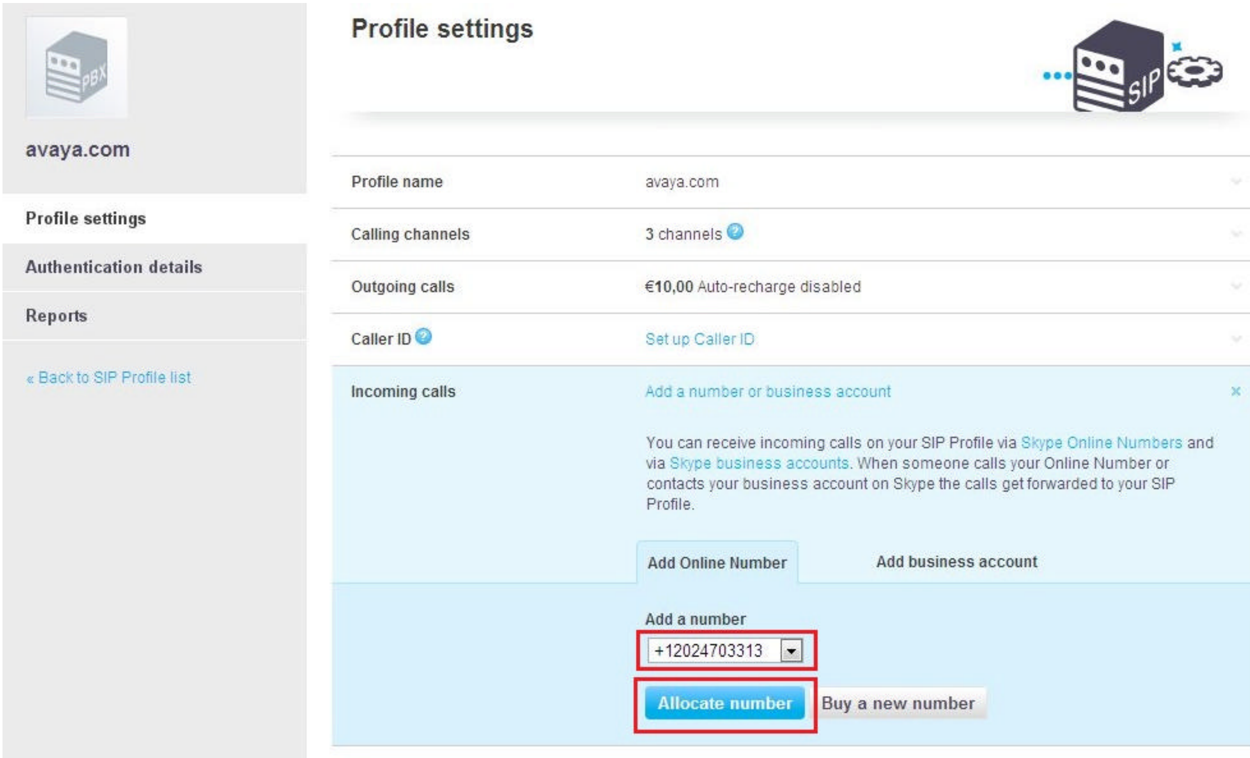
The **Buy Online Numbers** page is displayed. Under **Please choose your Online Numbers** type the number of required Online Numbers which are used as DIDs in conjunction with the SIP Profile. Select **a state or county** from the **Region Code** drop-down list box. Select an **area code** from the **Area code** drop-down list box. Under **Click an Online Number to select it:** Select a number. Scroll down and click **Buy now** (not shown).



The **Profile settings** page is displayed. Click **Add a number or business account**.

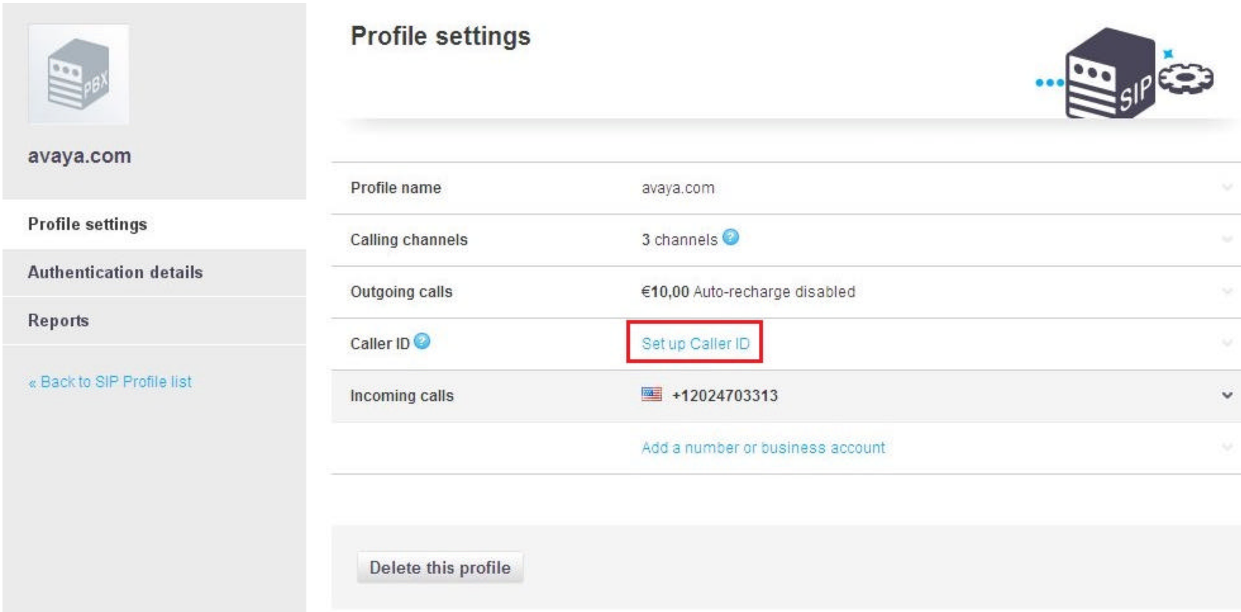


Select the newly purchased Online Number from the **Add a number** drop-down list box then click **Allocate number**.

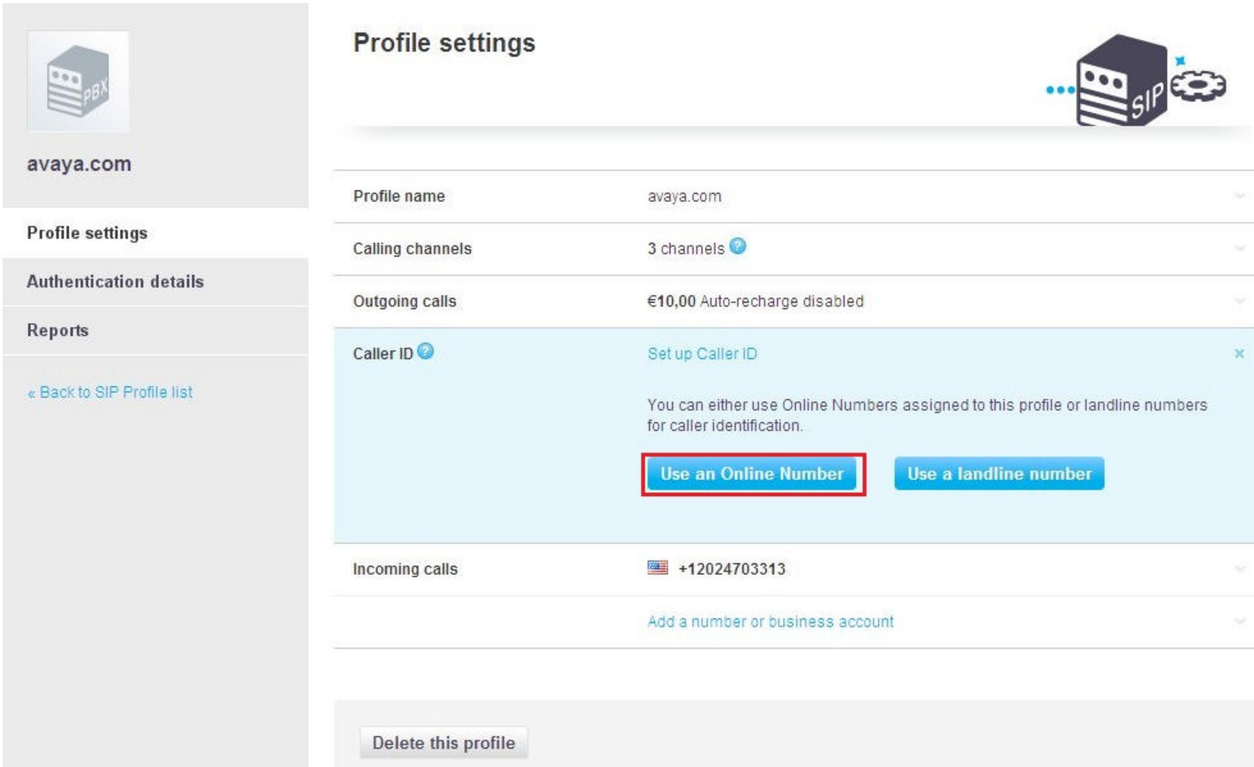


6.6. Administer Caller ID

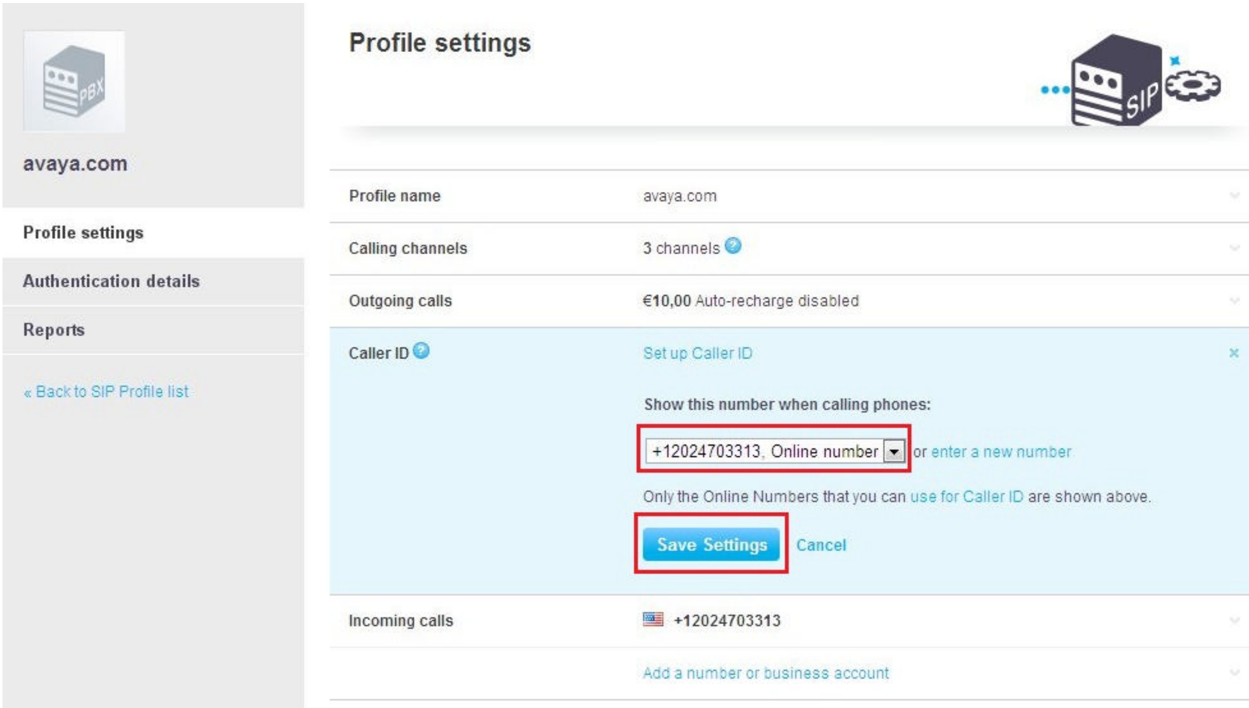
The **Profile settings** page is displayed. Click **Set up Caller ID**.



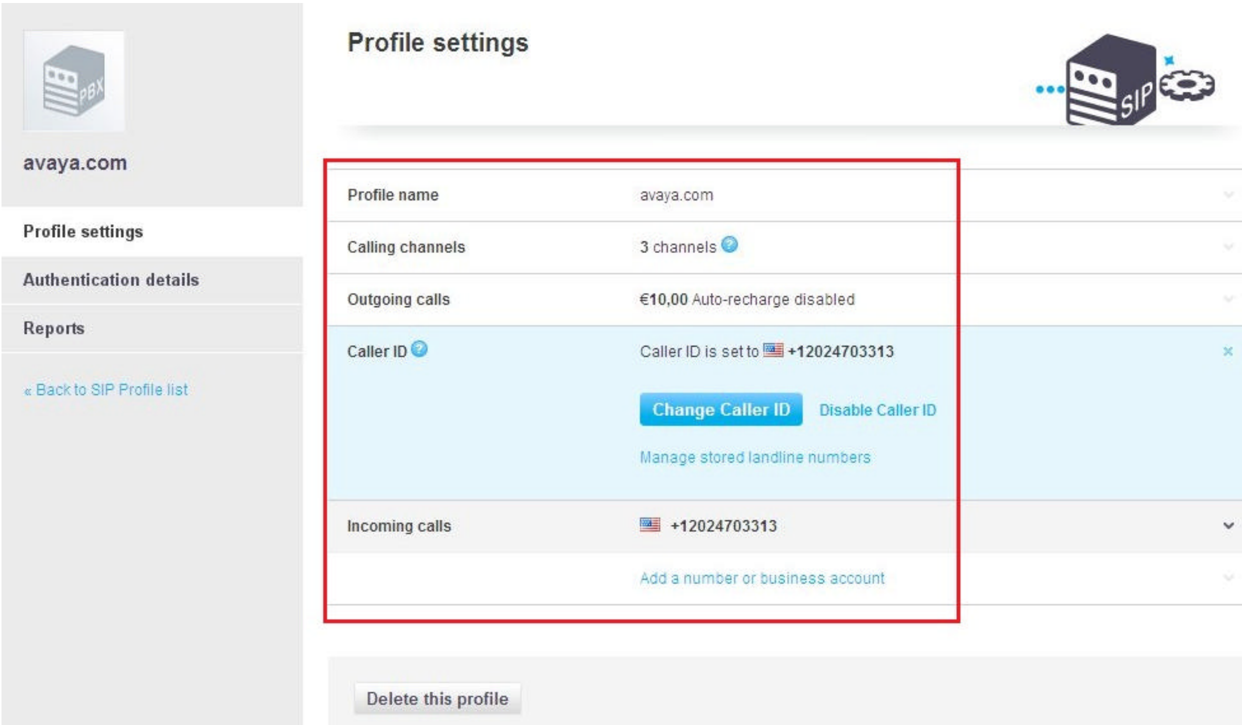
Click **Use and Online Number**.



Select an Online Number from the **Show this number when calling phones:** drop-down list box than click **Save Settings**.

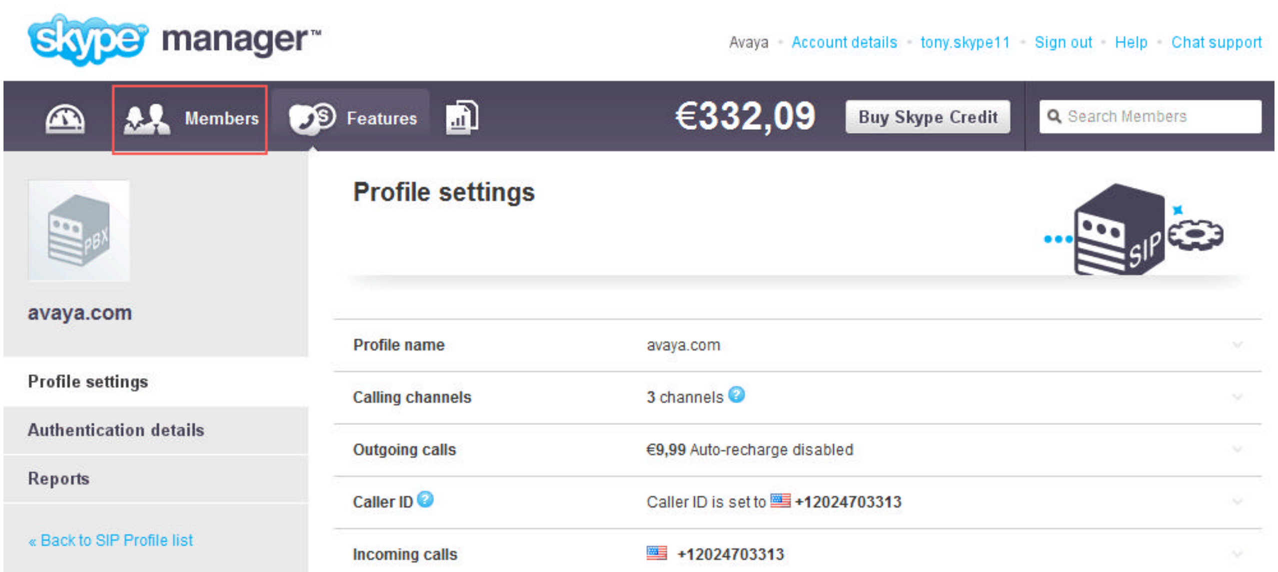


The following screenshot displays a sample Profile with both the Caller ID and the Incoming calls (DID) set to the same Online Number.

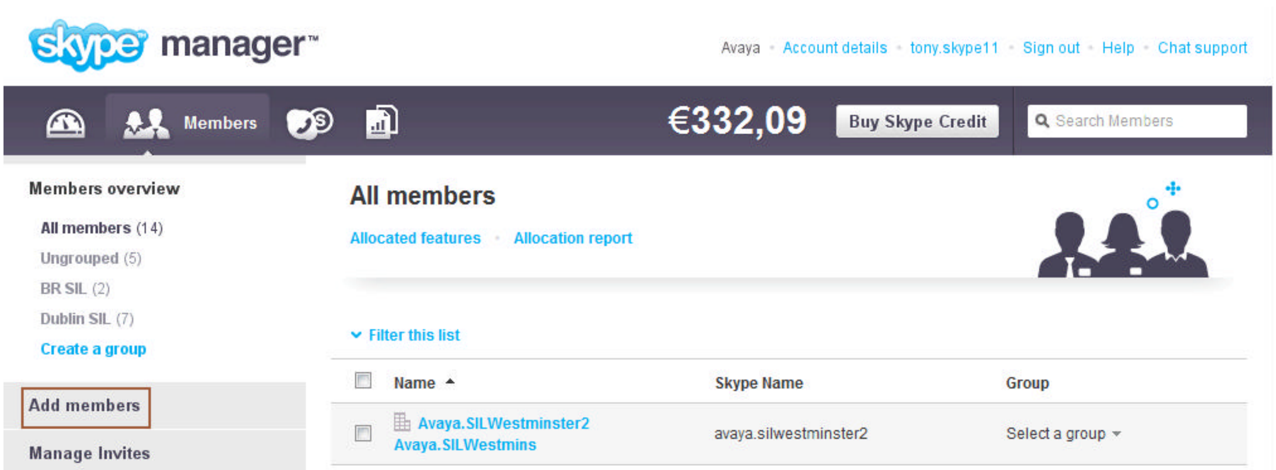


6.7. Create a Business User

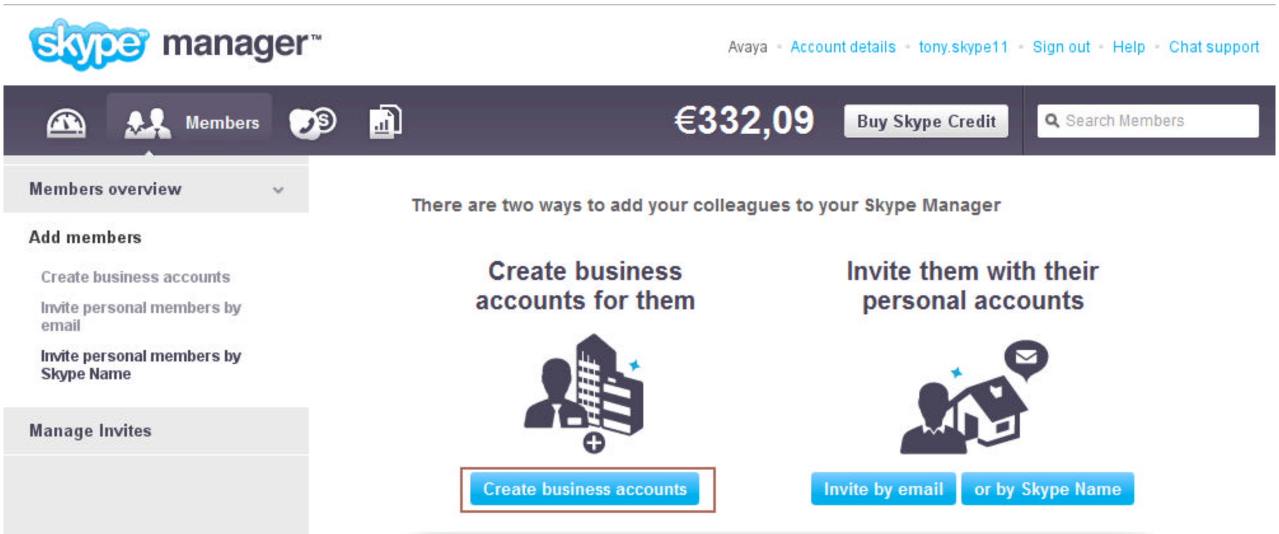
A Business User with a unique Skype Name needs to be created and associated with the extension of the enterprise user (3001) for click-to-call functionality to work from the Skype P2P Network. Click **Members** on the toolbar.



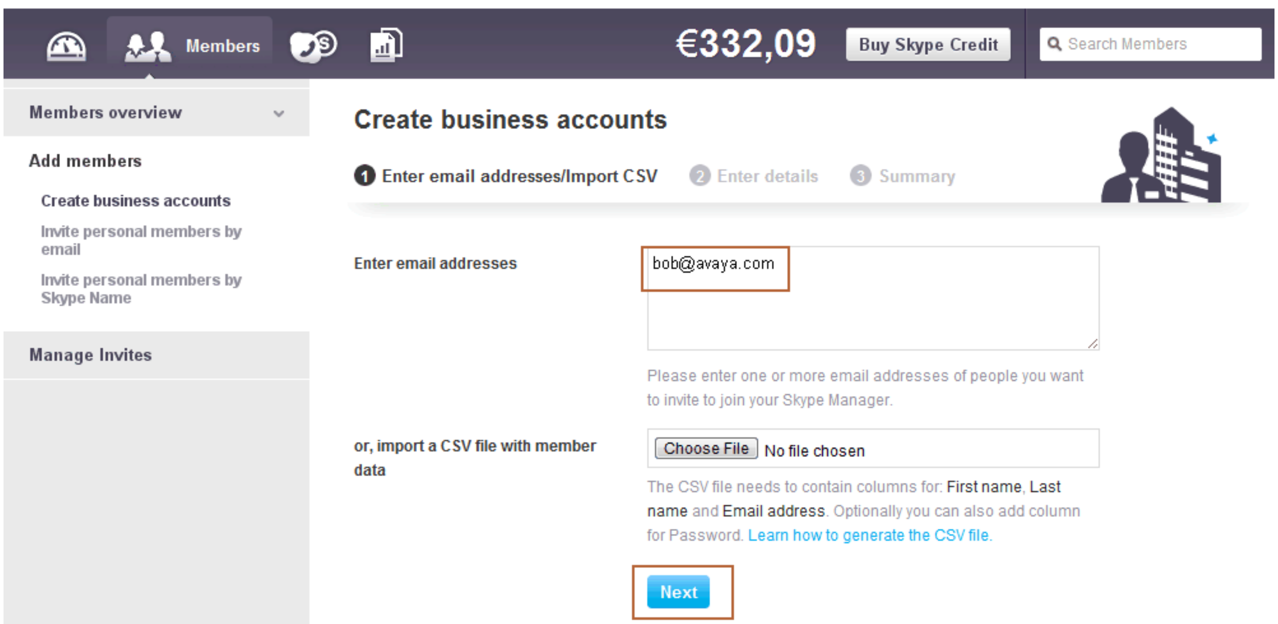
The **All Members** page is displayed. Click **Add members**.



The **Add members** page is displayed. Click **Create business accounts**.



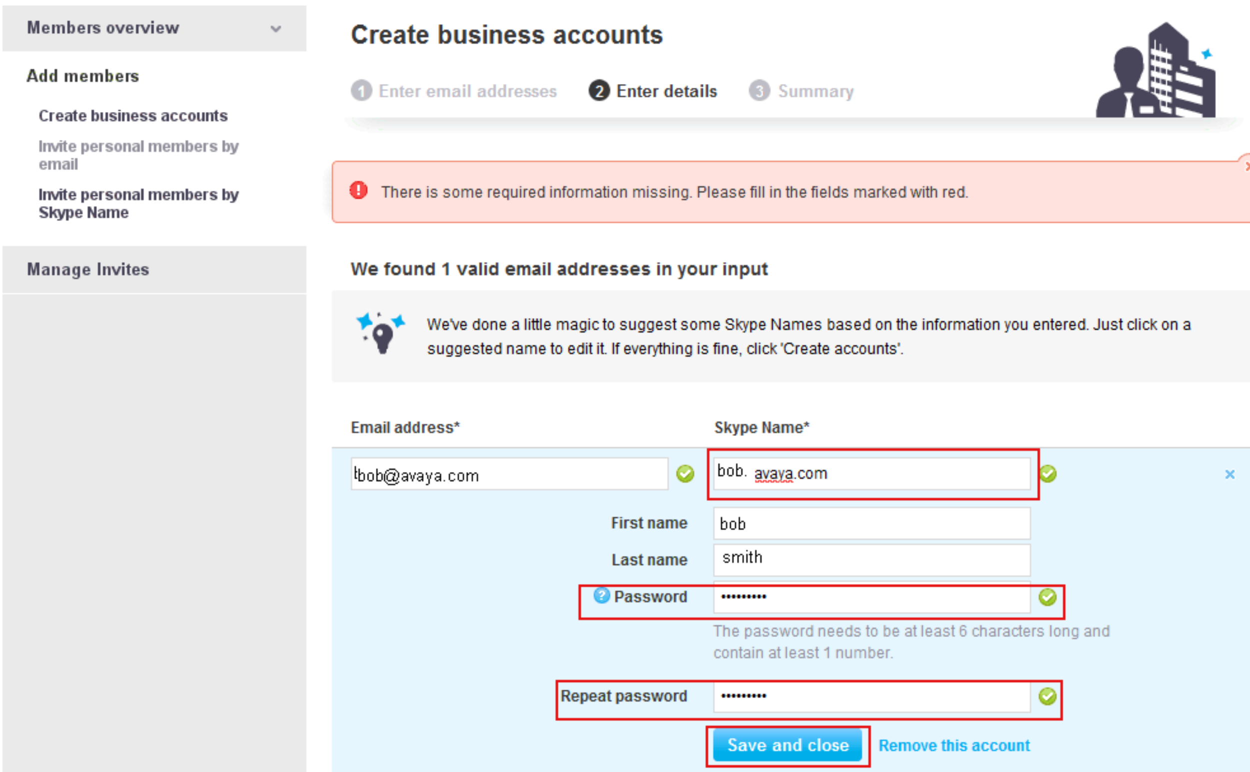
The **Create business accounts** page is displayed. Enter an email address in the text field as Skype Manager sends an automated email once the new Business account is created. Click **Next**.



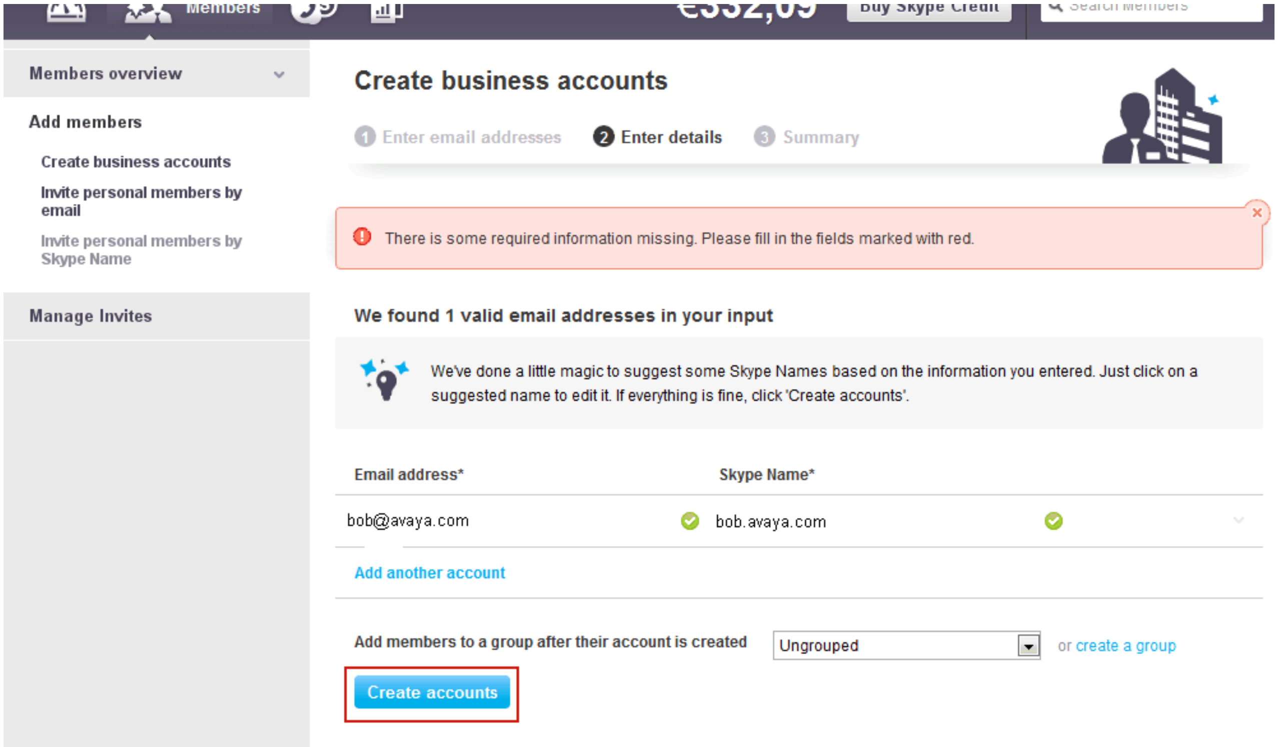
The **Create business accounts** page is displayed. Under **Enter details** fill in the following fields:

- **Skype Name :** Type a unique name. Skype Client users will click on this Skype Name to initiate a call to the enterprise user with extension 3001.
- **Password:** Type a password.
- **Repeat Password:** Repeat the password.

Default values can be used for the remaining fields. Click **Save and Close.**

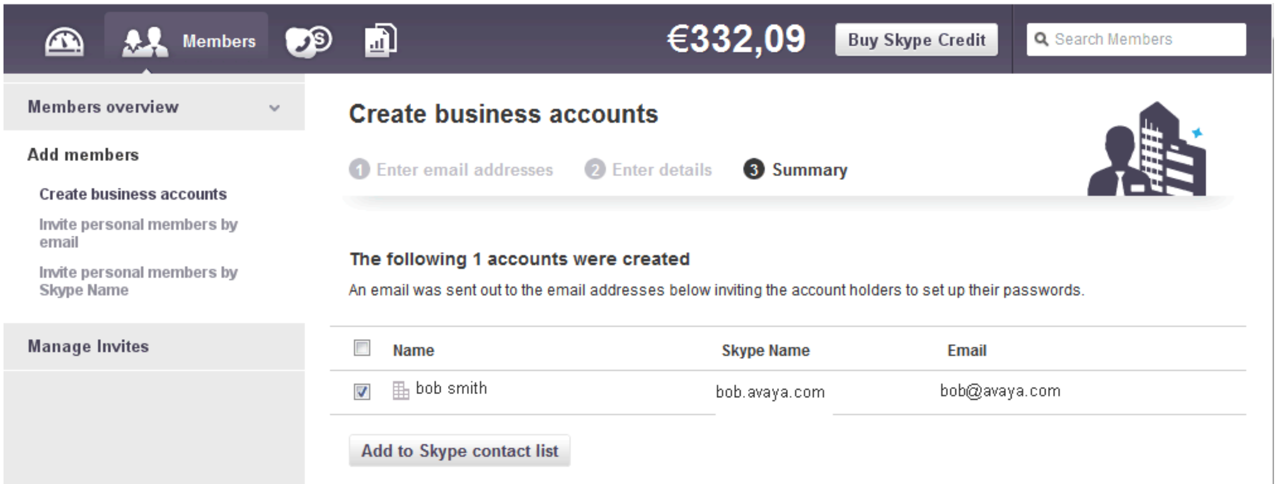


The **Create business accounts – Enter details** page is displayed. Click **Create accounts**.

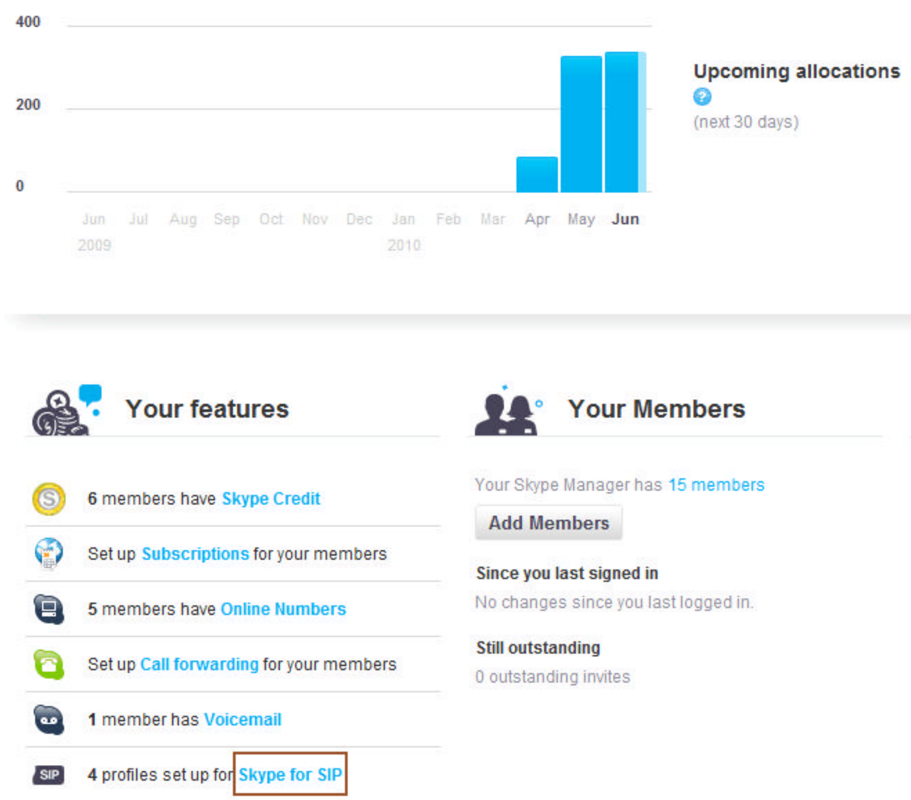


6.8. Add Business User to SIP Profile

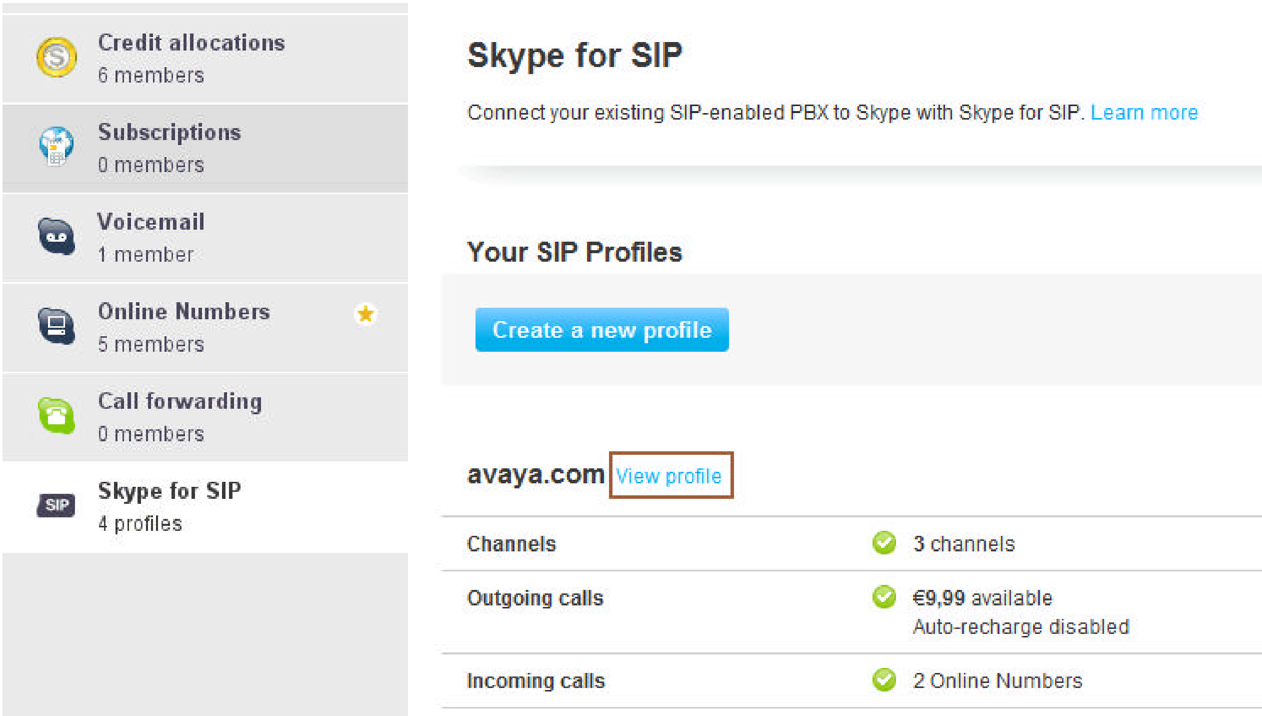
The **Create business accounts - Summary** page is displayed. Click **Dashboard**.



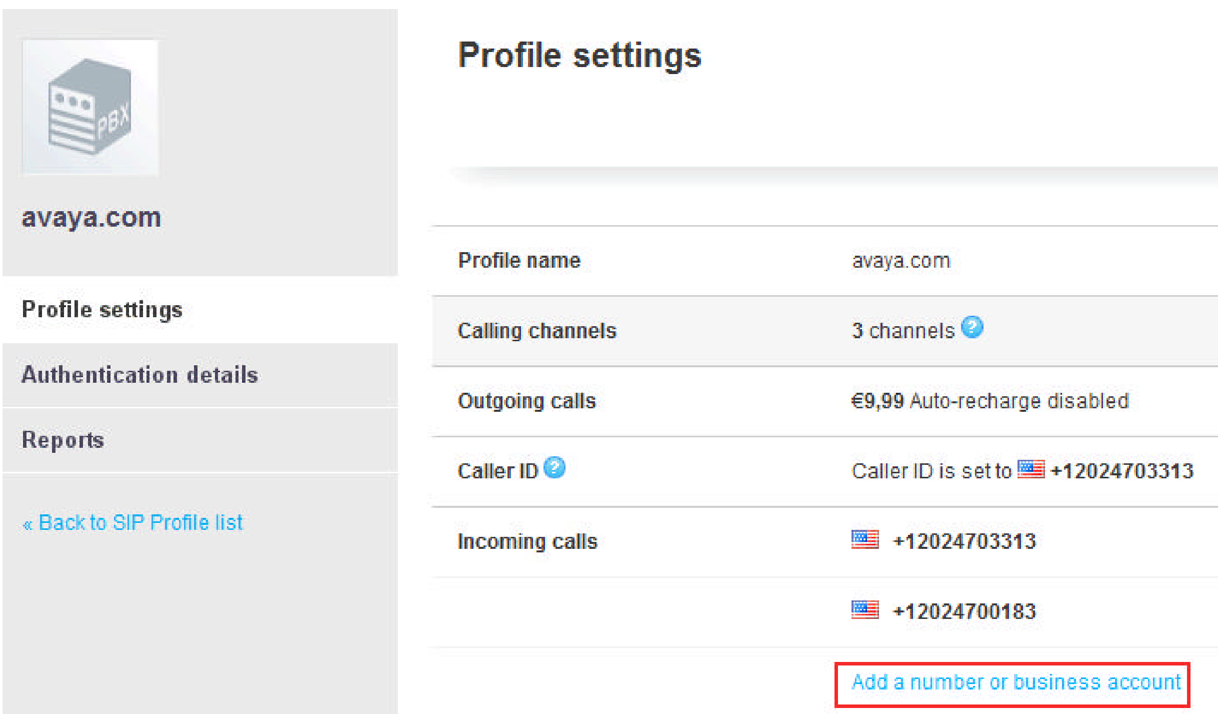
The **Dashboard** screen is displayed. Scroll down and click **Skype for SIP**.



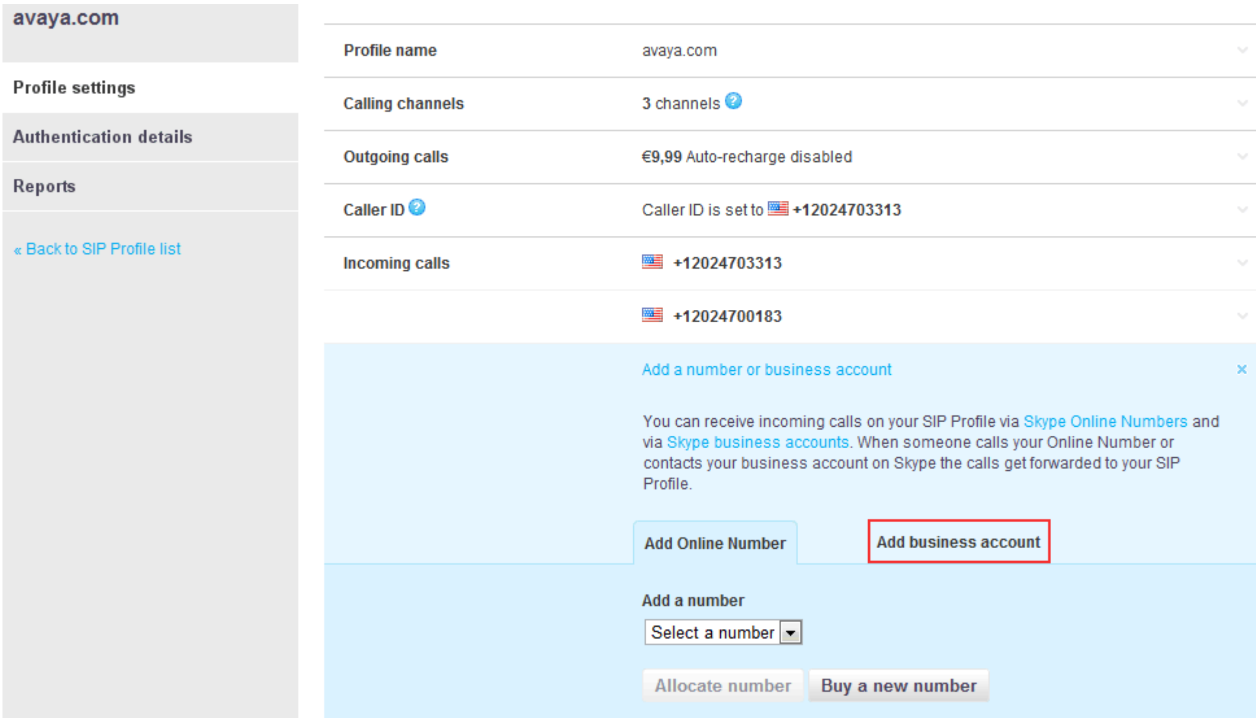
The **Skype for SIP** page is displayed. Click **View profile**.



The **Profile settings** page is displayed. Click **Add a number or business account**.



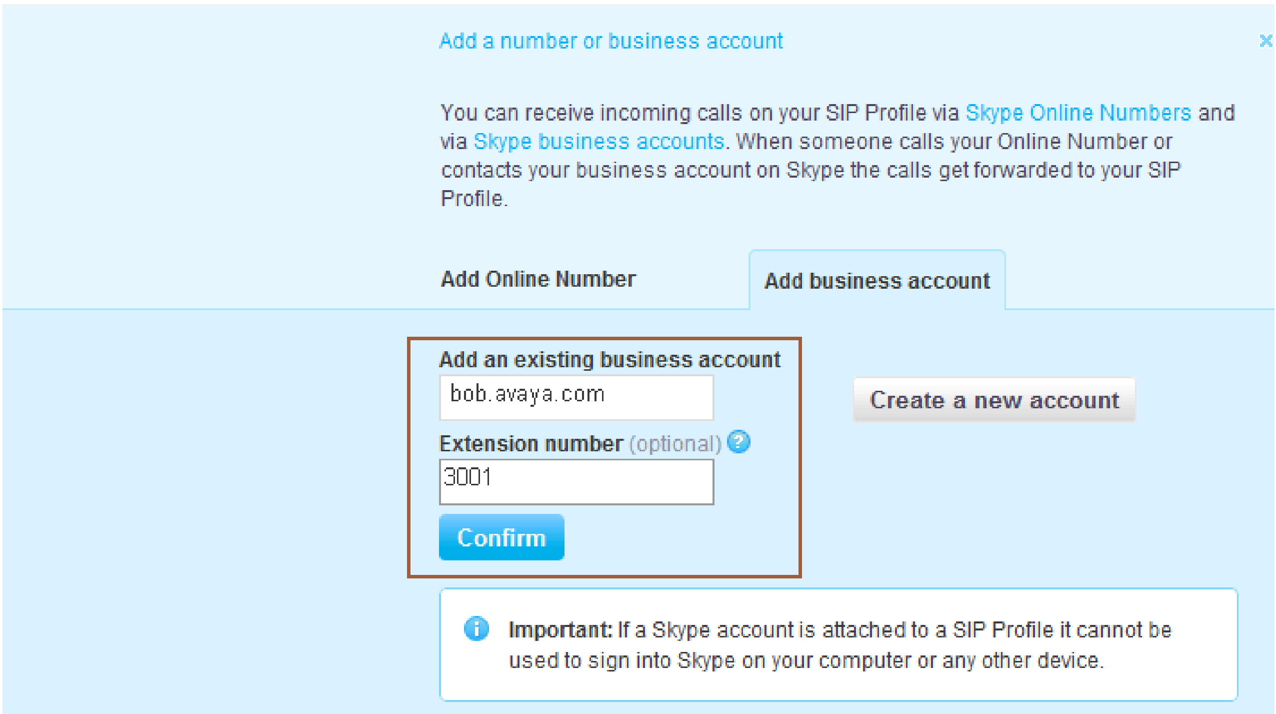
The **Add Online Number** tab is displayed. Click **Add business account**.



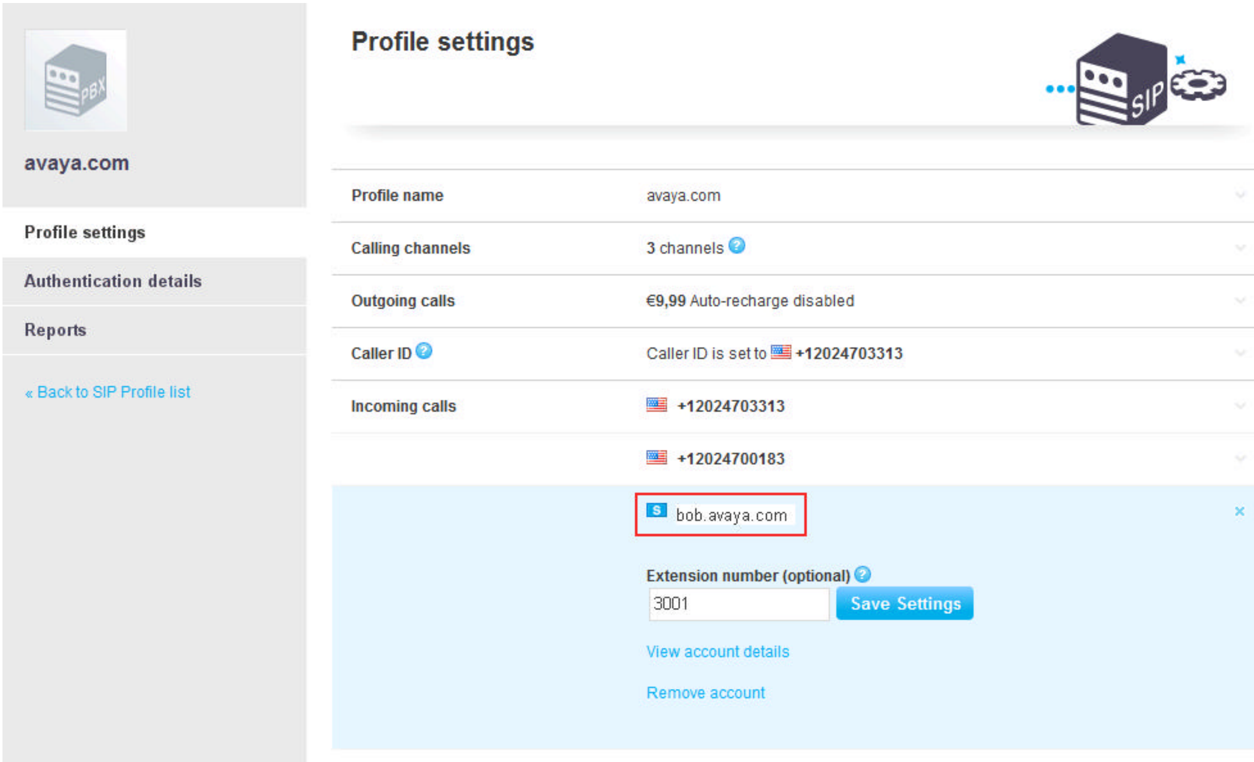
Under the **Add business account** tab fill in the following fields:

- **Add an existing business account:** Type a unique name. Skype Client users will click on the Skype Name to initiate a call to the enterprise user with extension 3001.
- **Extension number:** Type an extension.

Click **Confirm.**



The **Profile settings** page is displayed. Verify that the newly created Skype Name is displayed under the **Incoming calls** section.



7. Configure Avaya Aura™ Session Border Controller

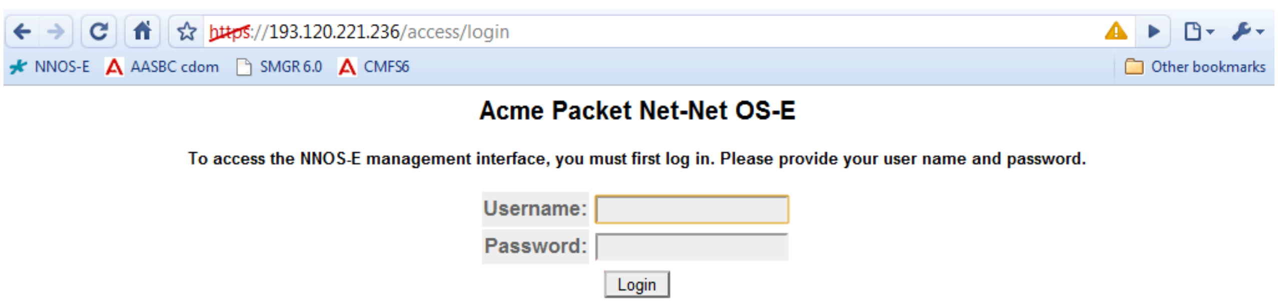
This section provides the procedures for configuring Avaya Aura™ Session Border Controller and includes the following items:

- Log in to Avaya Aura™ Session Border Controller using the GUI
- Administer SIP Domains
- Administer Outbound SIP Header Manipulation Rules
- Administer Inbound SIP Header Manipulation Rules
- Administer SIP Header Rules for Topology Hiding
- Save the Configuration

These Application Notes assume that the Avaya Aura™ Session Border Controller was installed with the AT&T Template.

7.1. Log in to Avaya Aura™ Session Border Controller using the GUI

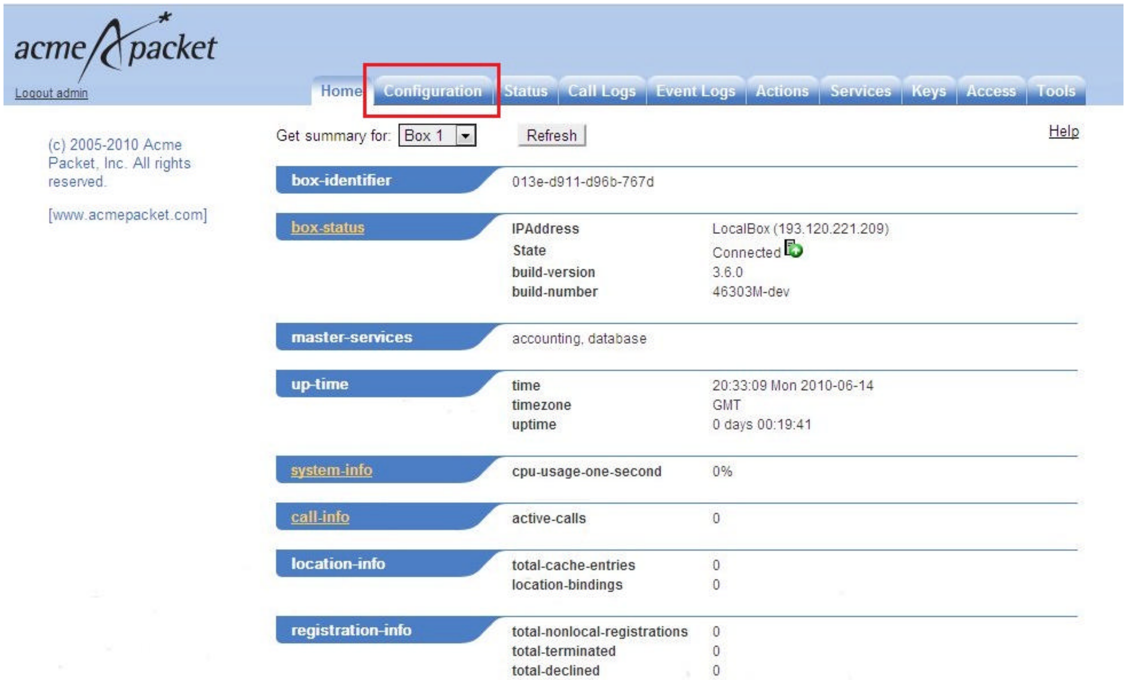
Configuration is accomplished by accessing the browser-based GUI of Avaya Aura™ Session Border Controller, using the URL "**https://<ip-address>**", where "<ip-address>" is the IP address of the inside interface of the Avaya Aura™ Session Border Controller. Log in with the appropriate credentials.



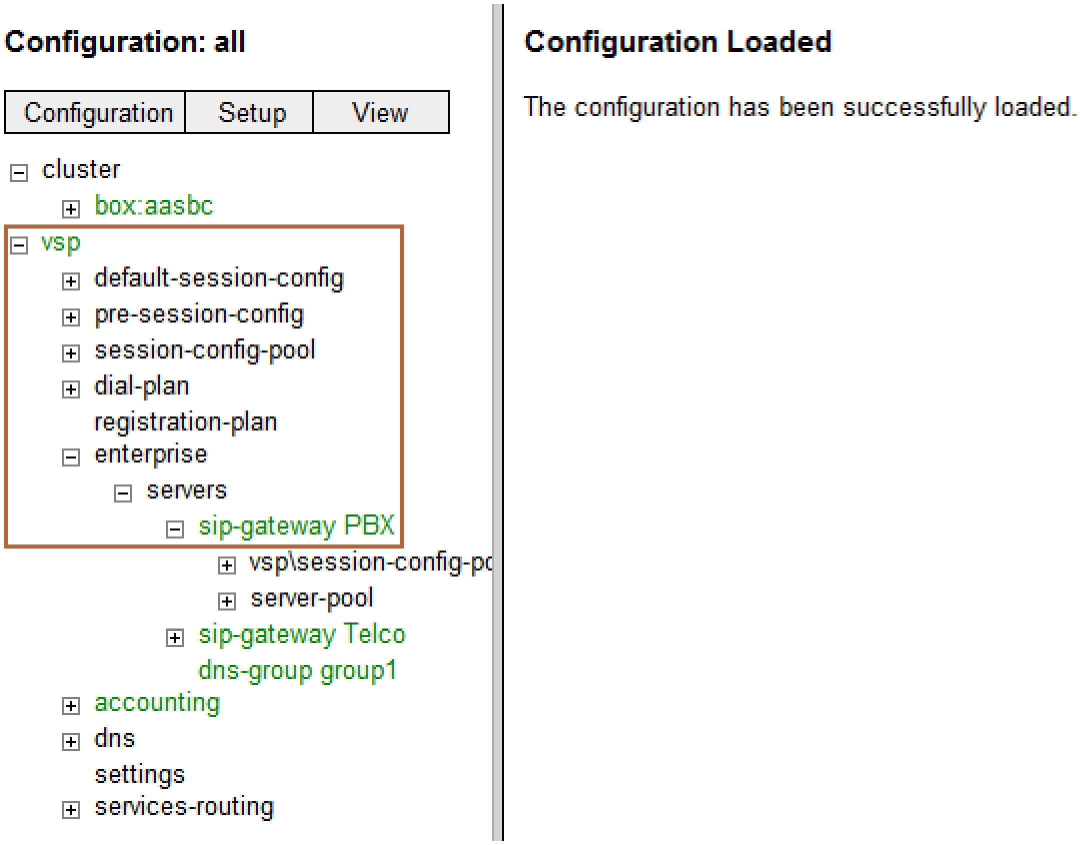
7.2. Administer SIP Domains

The Avaya Aura™ Session Border Controller performs topology hiding by translating the private domain avaya.com to the public domain sip.skype.com for outbound calls and vice-versa for inbound calls. The following steps assign the domain names to the corresponding SIP Entities.

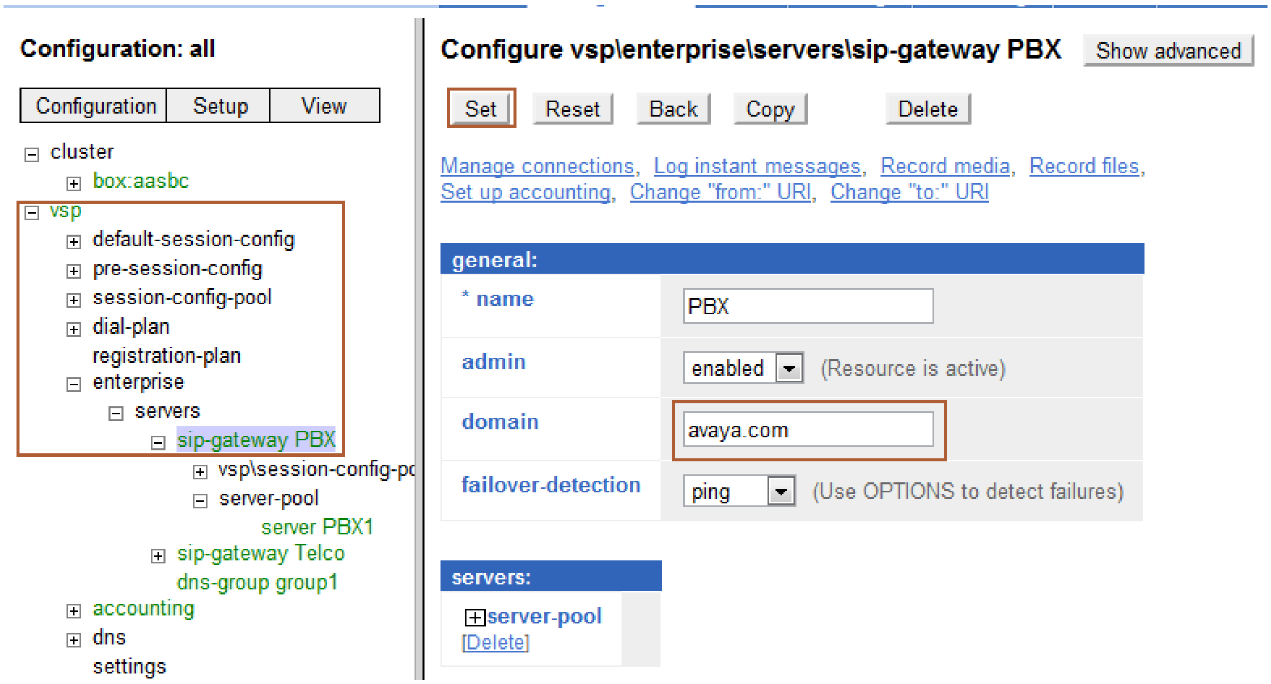
The **Home** page is displayed. Select the **Configuration** tab on the toolbar.



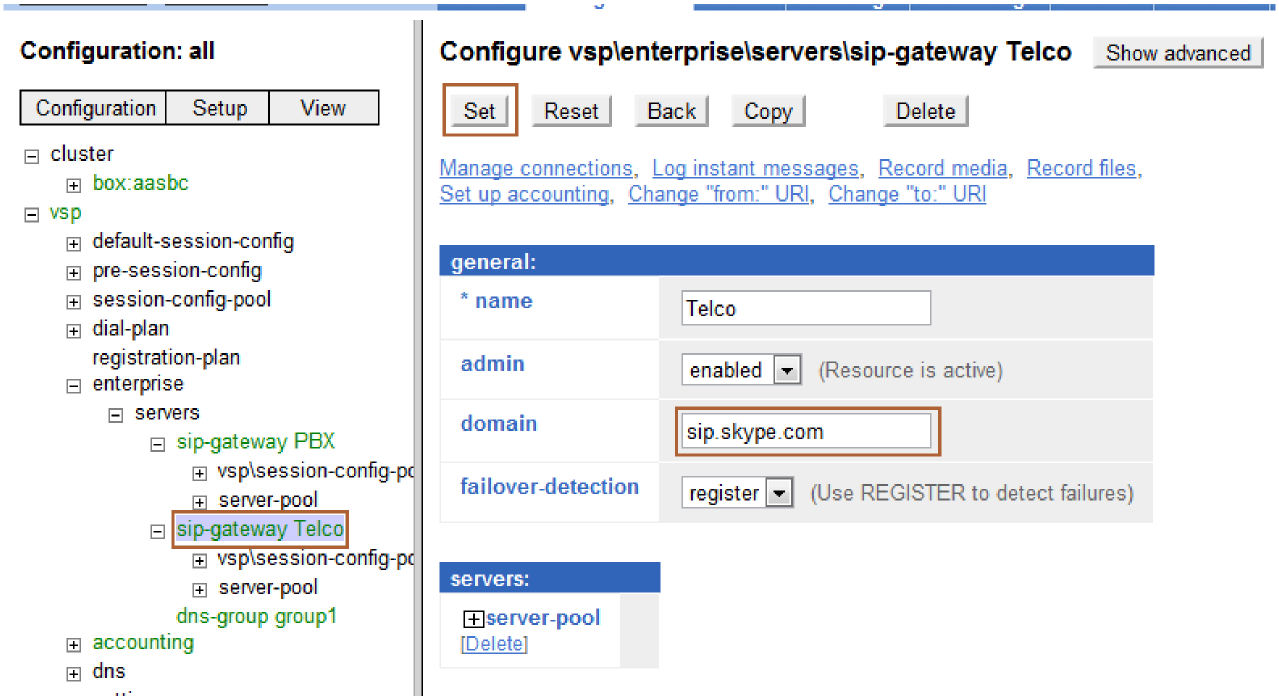
The **Configuration Loaded** page is displayed. Expand **vsp -> enterprise -> servers** and click on **sip-gateway PBX.**



The **Configure vsp\enterprise\servers\sip-gateway PBX** page is displayed. Type **avaya.com** in the domain field then click **Set**.

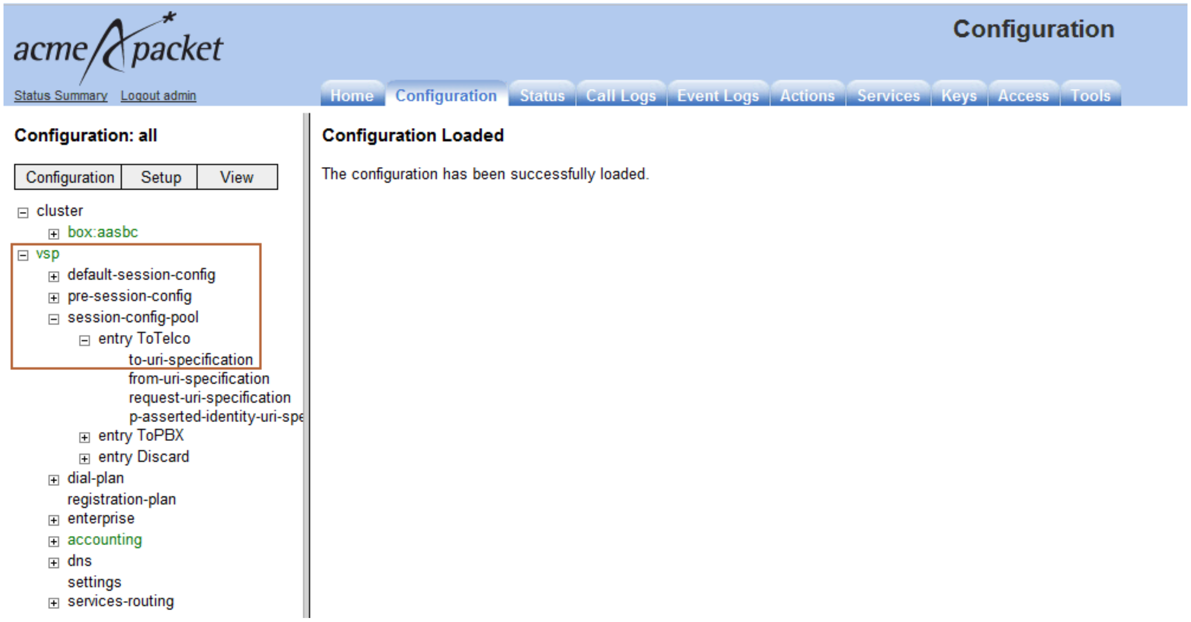


Select **sip-gateway Telco** on the left pane. The **Configure vsp\enterprise\servers\sip-gateway Telco** page is displayed. Type **sip.skype.com** in the **domain** field then click **Set**.

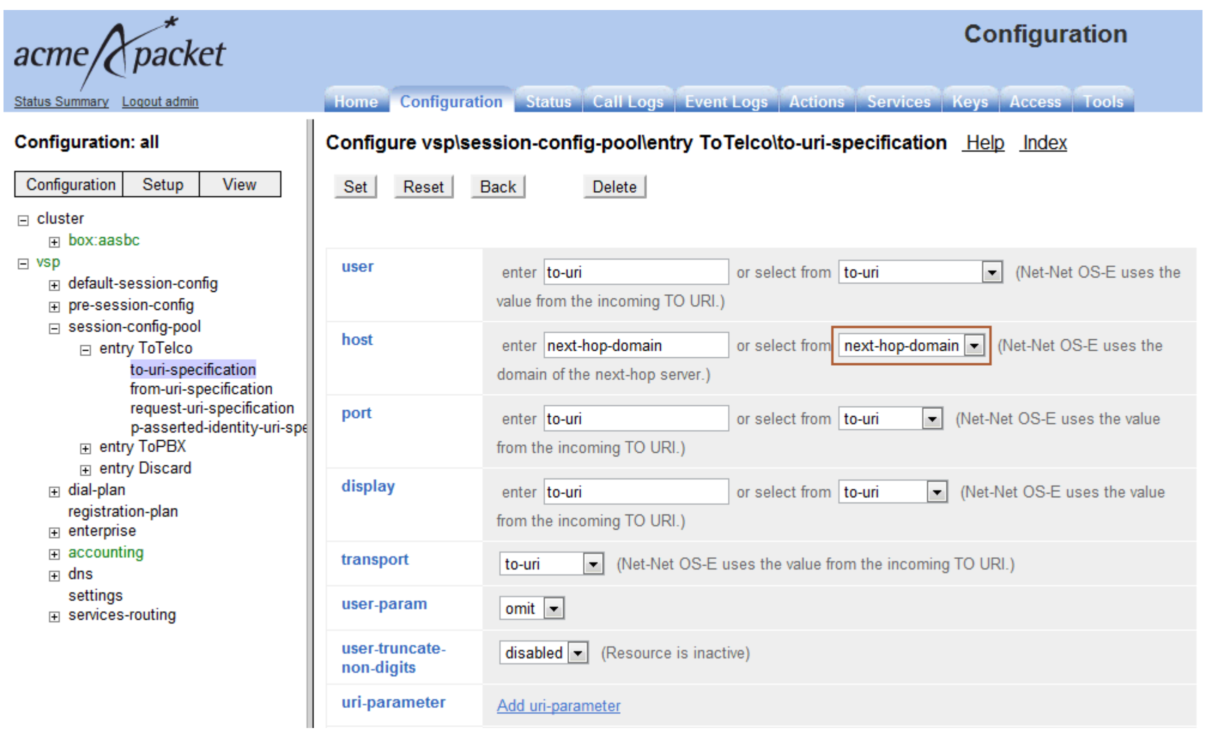


7.3. Administer Outbound SIP Header Manipulation Rules

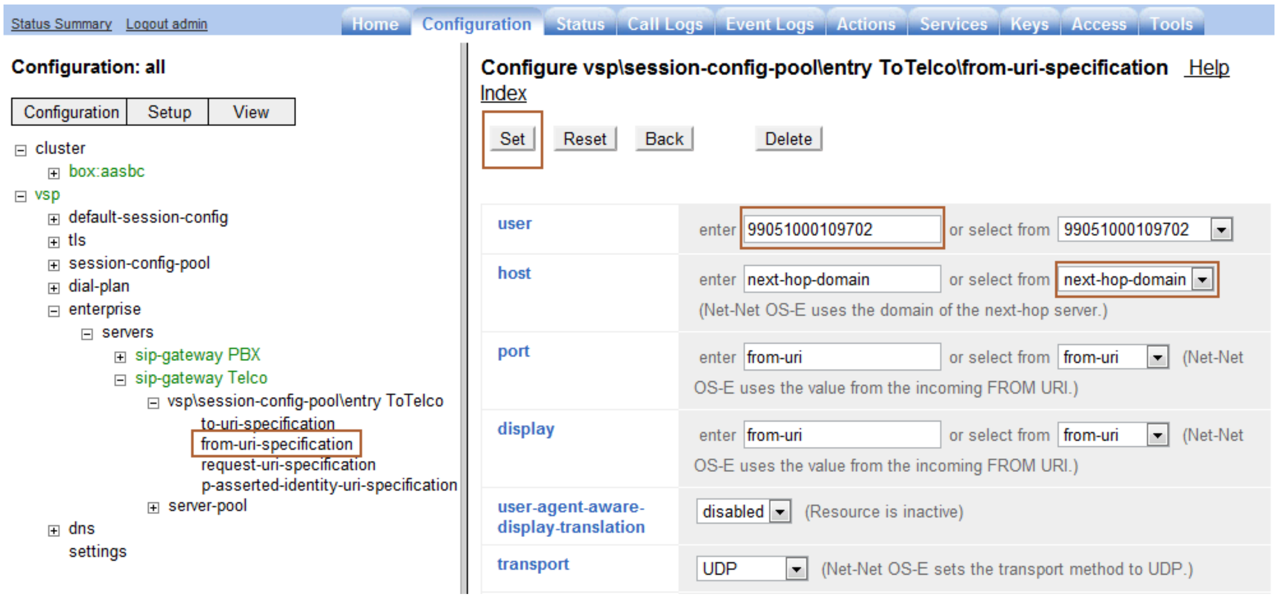
The following steps translate the SIP domain names created in **Section 7.2** in the To, From, Request-URI and PAI headers and assign the Skype ID to the user part of the From header as it is required for outbound calls to work. The **Configuration** page is displayed. Expand **vsp → session-config-pool → entry ToTelco → to-uri-specification** on the left pane.



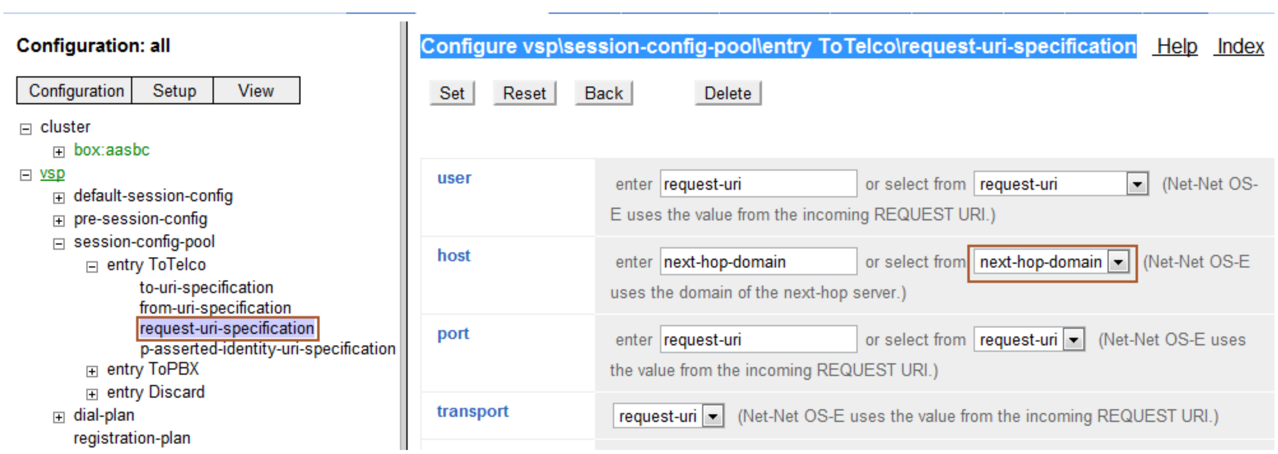
Under **host** select **next-hop-domain** from the drop-down list box then click **Set**.



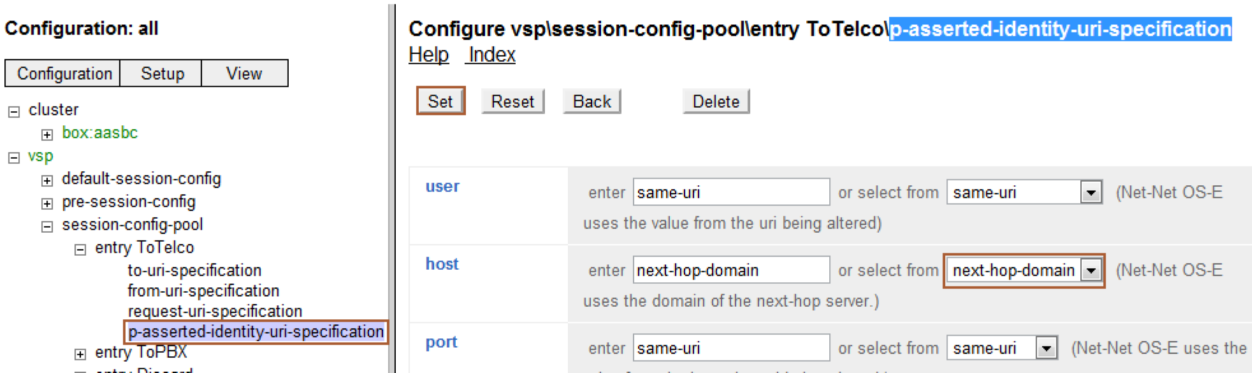
Select **from-uri-specification** from the left pane. The **Configure vsp\session-config-pool\entry ToTelco\from-uri-specification** page is displayed. Type the Skype Manager assigned **SIP User** from **Section 6.3** in the **user** field. Under **host** select **next-hop-domain** from the drop-down list box then click **Set**.



Select **request-uri-specification** from the left pane. The **Configure vsp\session-config-pool\entry ToTelco\request-uri-specification** page is displayed. Under **host** select **next-hop-domain** from the drop-down list box then click **Set**.

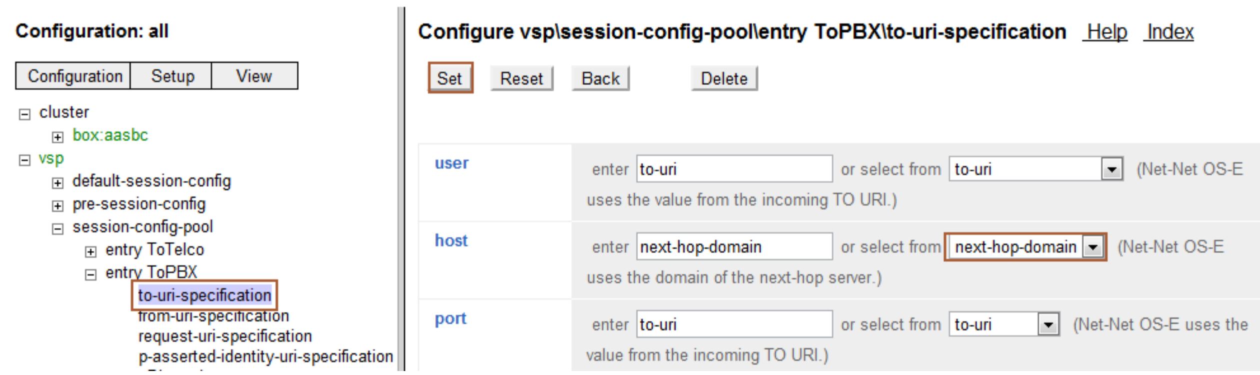


Select **p-asserted-identity-uri-specification** from the left pane. The **Configure vsp\session-config-pool\entry ToTelco\ p-asserted-identity-uri-specification** page is displayed. Under **host** select **next-hop-domain** from the drop-down list box then click **Set**.

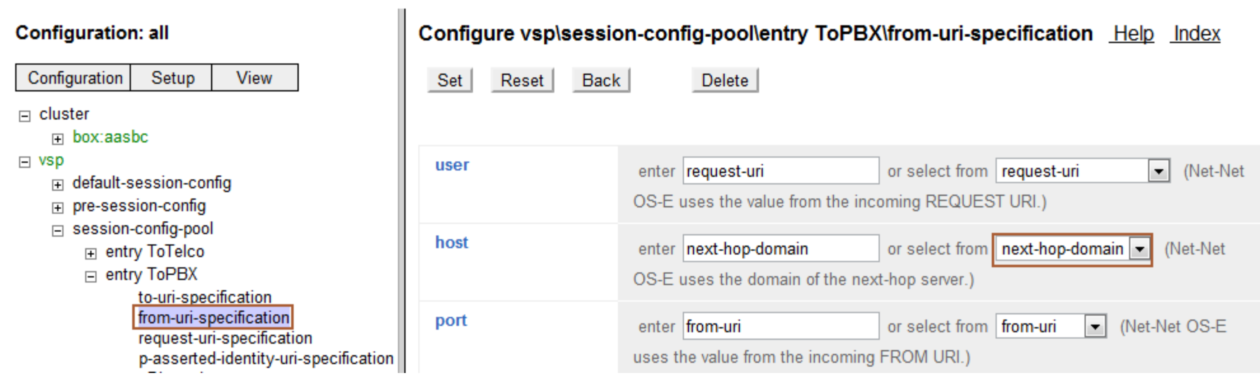


7.4. Administer Inbound SIP Header Manipulation Rules

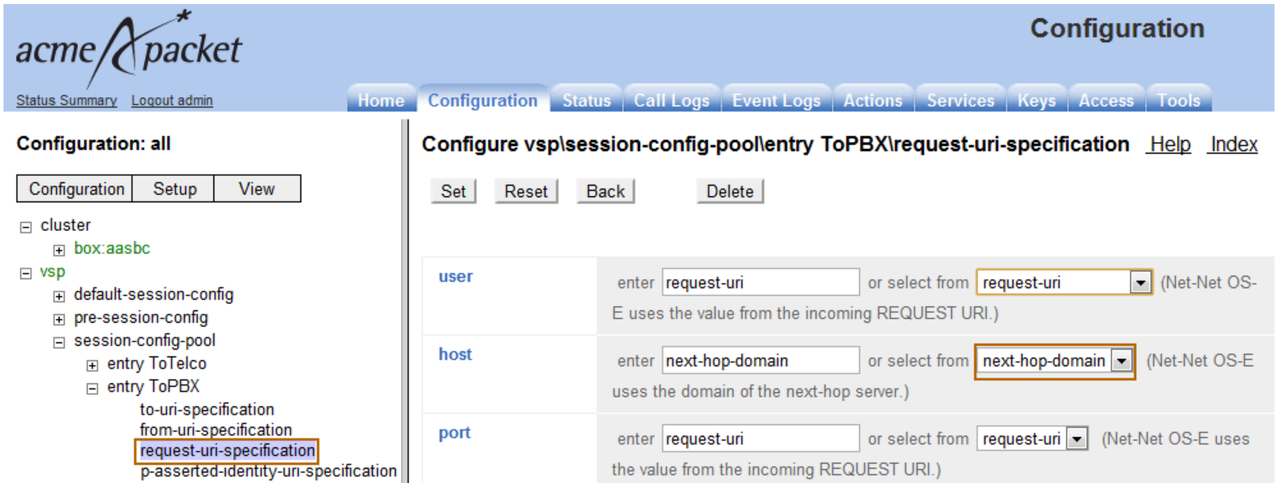
The following steps translate the SIP domain names created in **Section 7.2** in the To, From, Request-URI and PAI headers from sip.skype.com to avaya.com. Expand **vsp → session-config-pool → entry ToPBX → to-uri-specification** on the left pane. The **Configure vsp\session-config-pool\entry ToPBX\to-uri-specification** page is displayed. Under **host** select **next-hop-domain** from the drop-down list box.



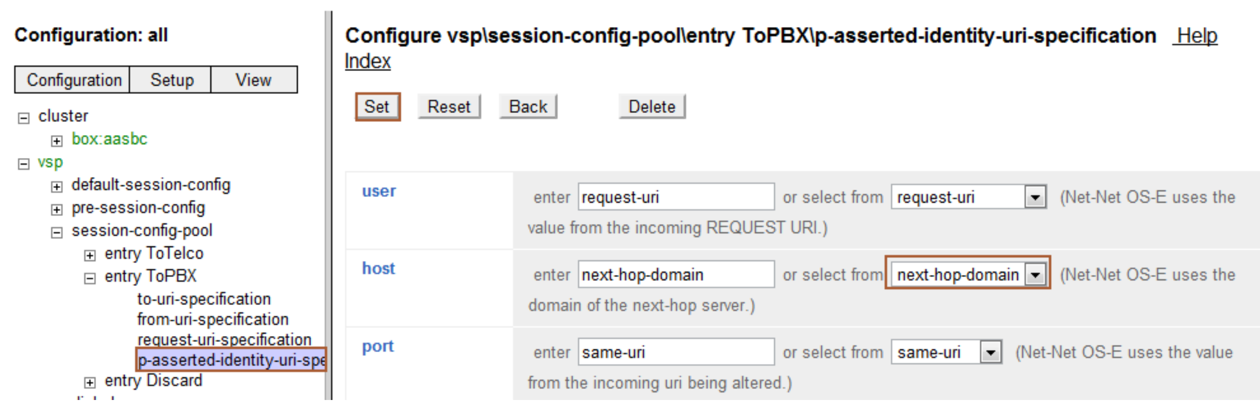
Select **from-uri-specification** from the left pane. The **Configure vsp\session-config-pool\entry ToPBX\from-uri-specification** page is displayed. Under **host** select **next-hop-domain** from the drop-down list box then click **Set**.



Select **request-uri-specification** from the left pane. The **Configure vsp\session-config-pool\entry ToPBX\request-uri-specification** page is displayed. Under **host** select **next-hop-domain** from the drop-down list box then click **Set**.

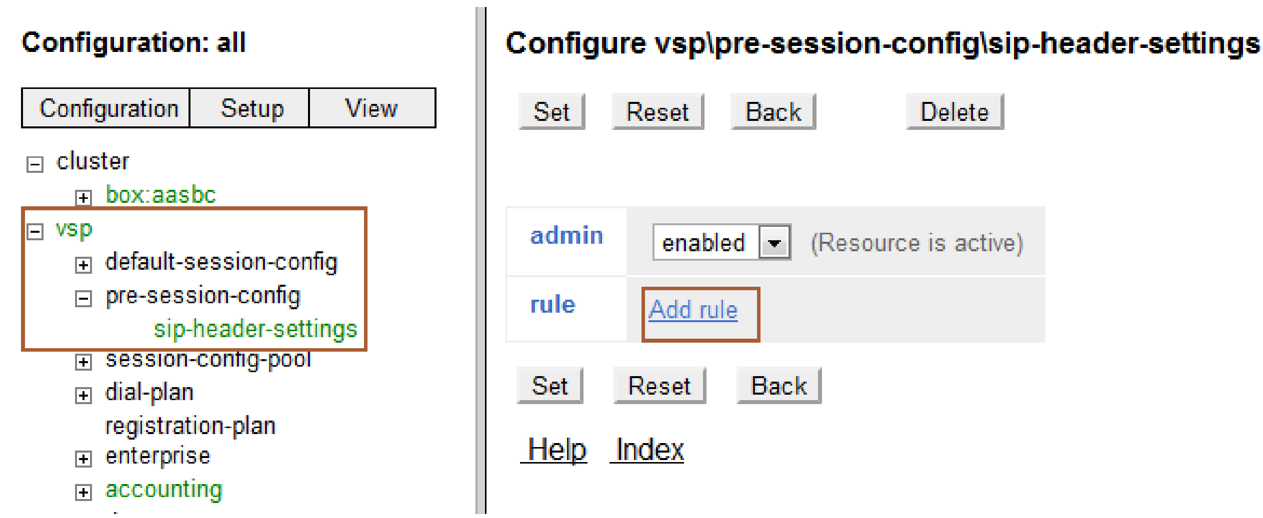


Select **p-asserted-identity-uri-specification** from the left pane. The **Configure vsp\session-config-pool\entry ToPBX\p-asserted-identity-uri-specification** page is displayed. Under **host** select **next-hop-domain** from the drop-down list box then click **Set**.

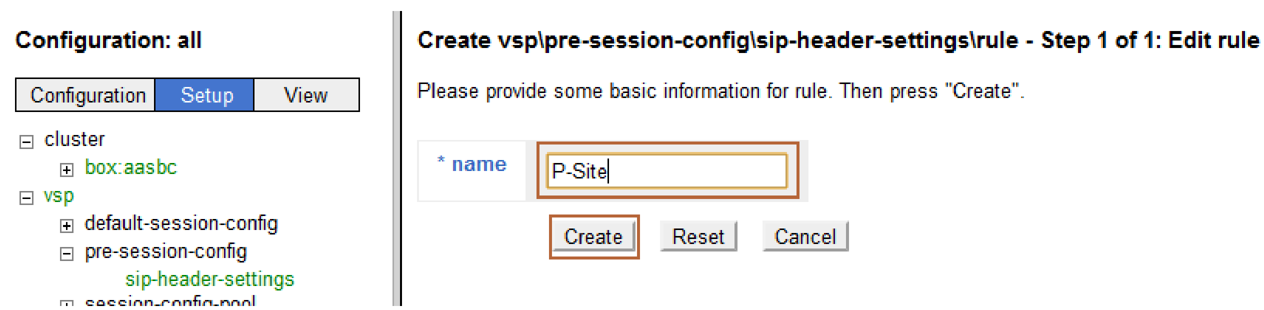


7.5. Administer SIP Header Rules for Topology Hiding

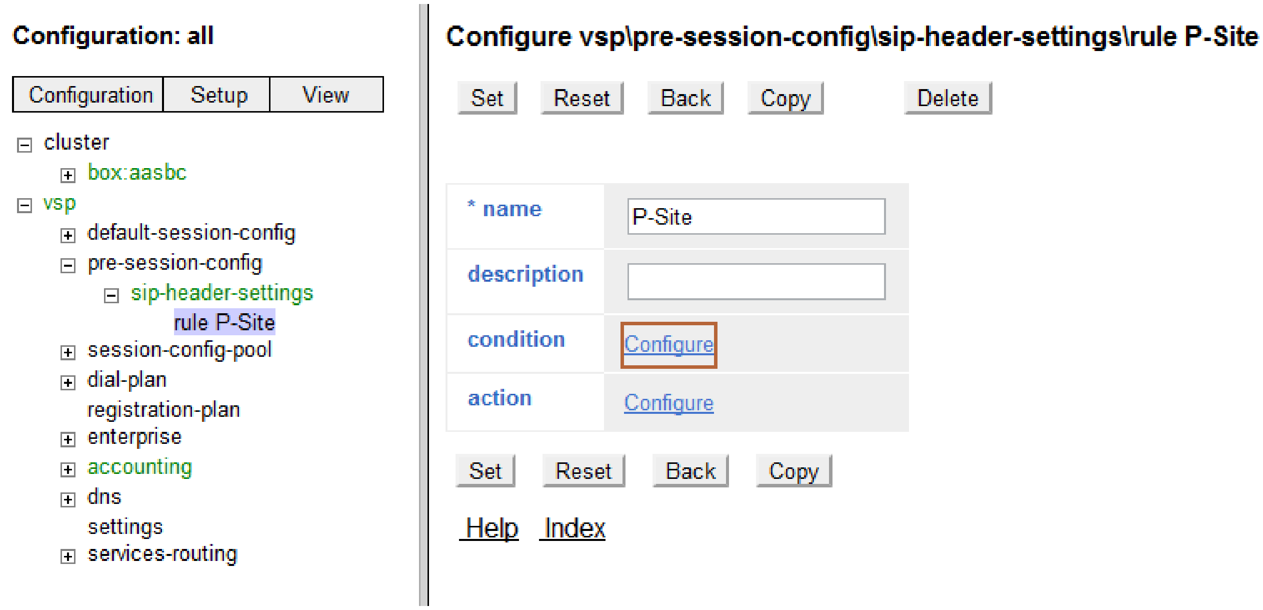
In the outgoing INVITE message to Skype Connect the Avaya Aura™ Session Manager inserts the P-Site proprietary with the private IP address of Avaya Aura™ System Manager thereby exposing the private IP addressing scheme to the public Internet. This section describes a SIP header manipulation rule which strips the P-Site header from the outgoing INVITE. Expand **vsp → pre-session-config → sip-header-settings**. The **Configure vsp\pre-session-config\sip-header-settings** page is displayed. Click **Add rule**.



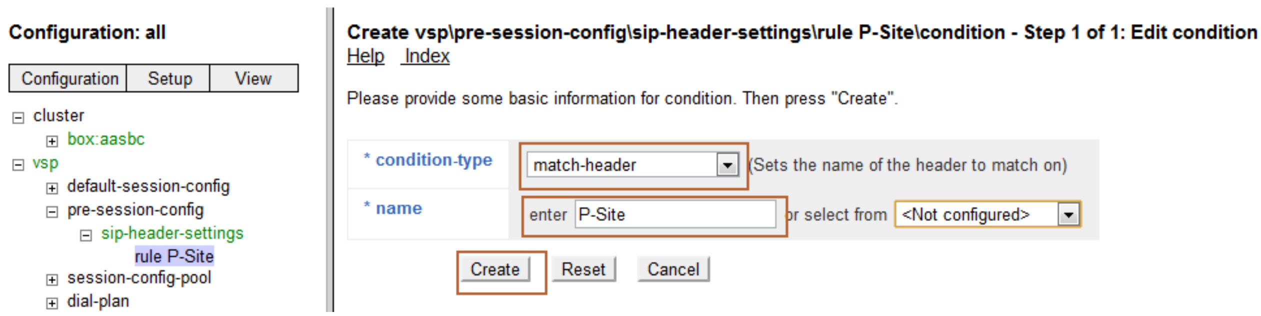
Type a descriptive name for the rule and click **Create**.



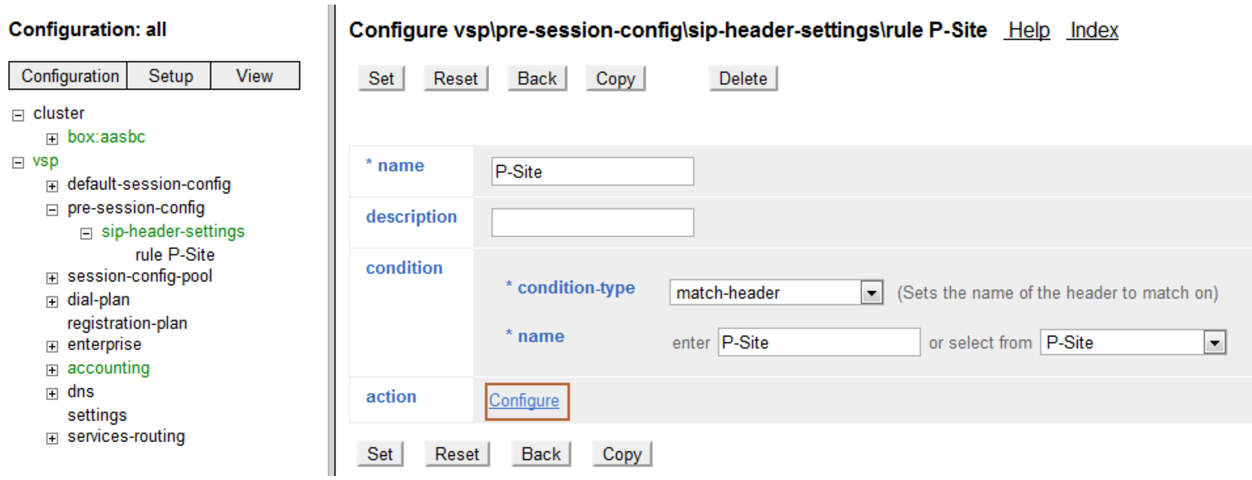
The **Configure vsp\pre-session-config\sip-header-settings\rule P-Site** page is displayed. Click **Configure**.



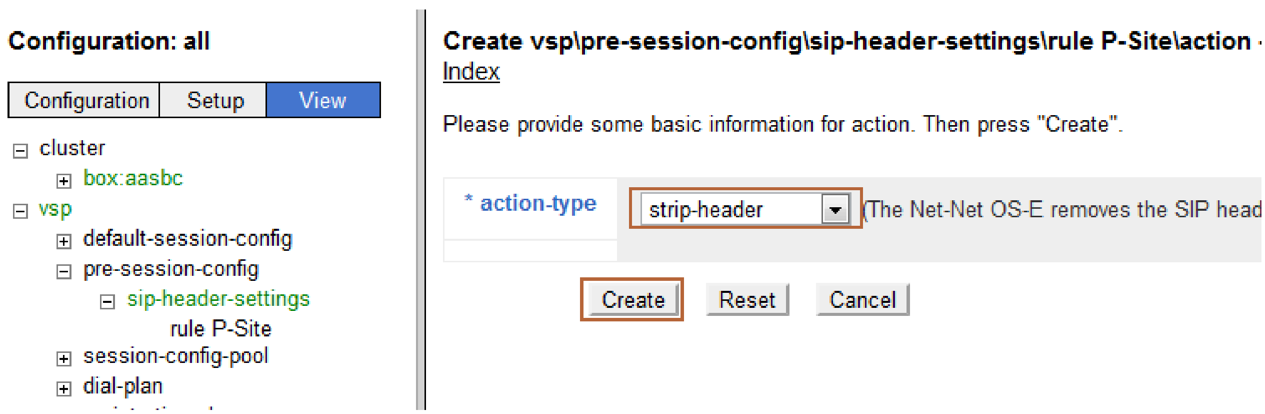
The **Create vsp\pre-session-config\sip-header-settings\rule P-Site\condition** page is displayed. Under **condition-type** select **match-header**. Under **name** type **P-Site** then click Create.



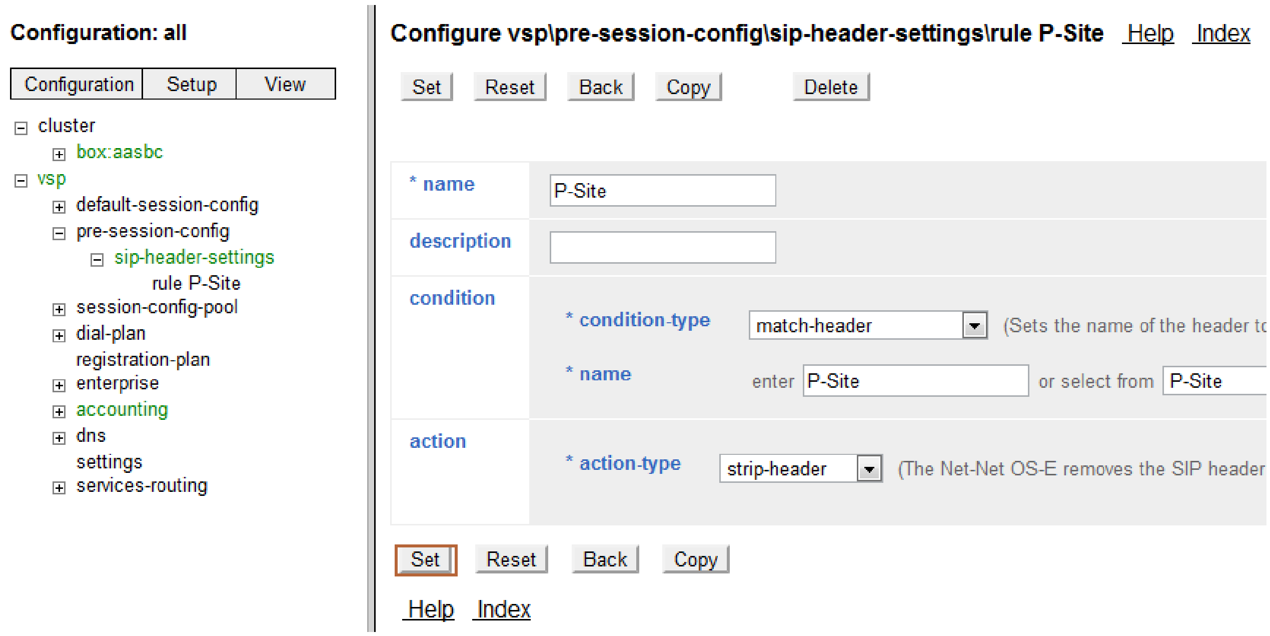
The **Configure vsp\pre-session-config\sip-header-settings\rule P-Site** page is displayed. Click **Configure**.



The **Create vsp\pre-session-config\sip-header-settings\rule P-Site\action** page is displayed. Under **action-type** select **strip-header** then click **Create**.

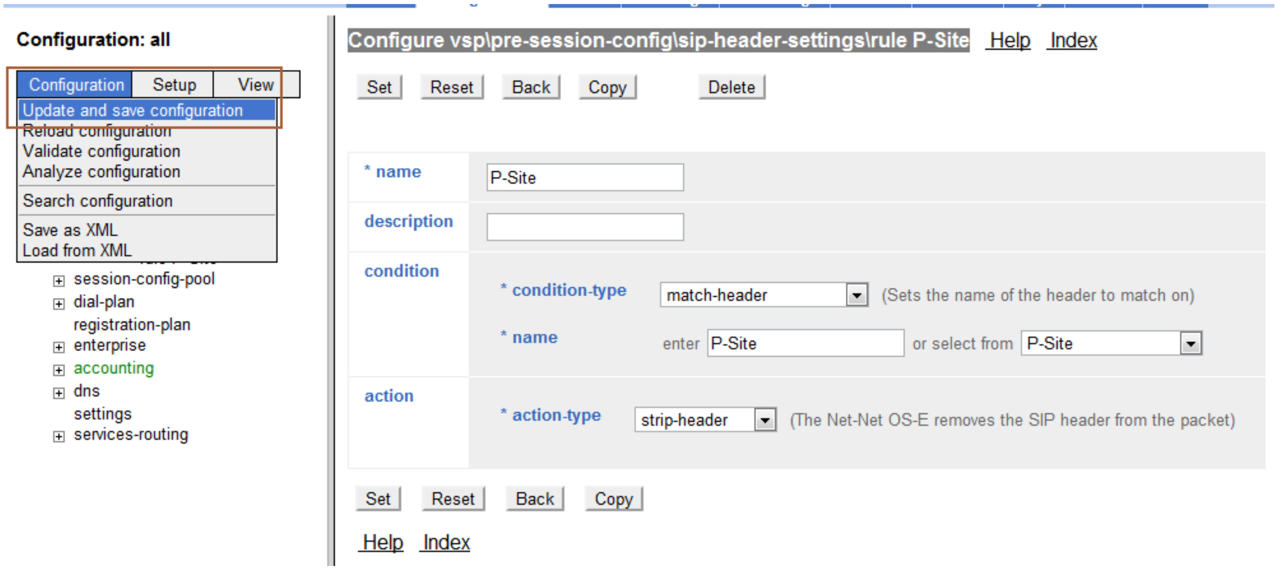


The **Configure vsp\pre-session-config\sip-header-settings\rule P-Site** page is displayed. Click **Set**.

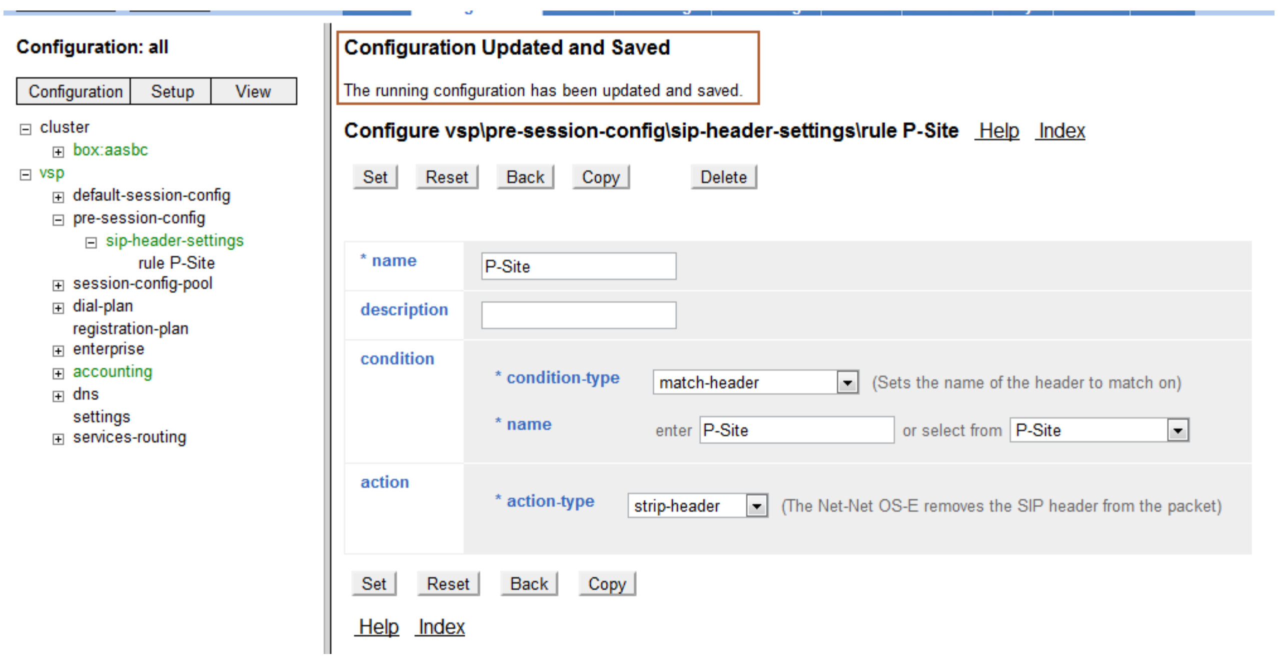


7.6. Save the Configuration

Click **Configuration** on the left pane then select **Update and save configuration**.



Once the configuration is written to disk the **Configuration Updated and Saved** message is displayed.



8. General Test Approach and Test Results

The test environment consisted of:

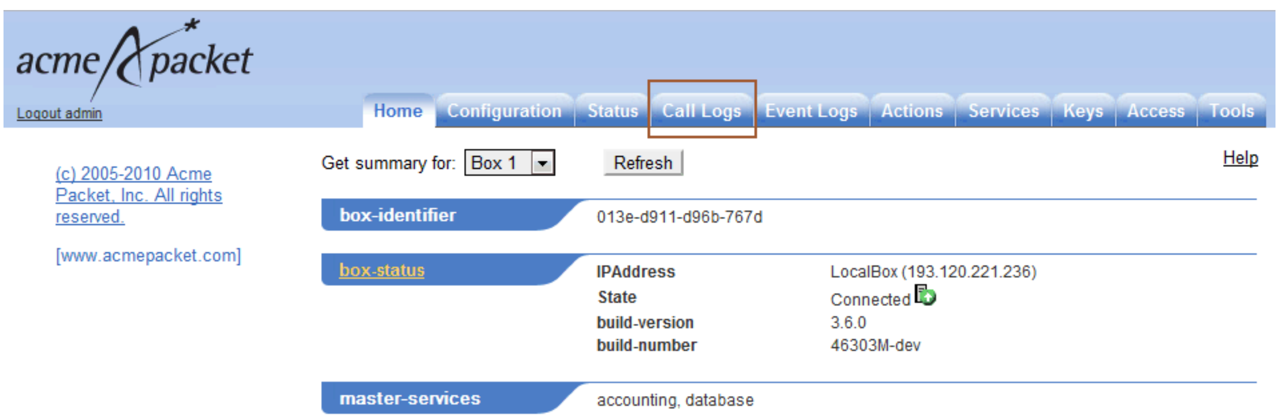
- A simulated enterprise with Avaya Aura™ System Manager, Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager, Avaya phones, Avaya Aura™ Session Border Controller, and Avaya Modular Messaging.
- A production version of the Skype Connect service, to which the simulated enterprise was connected.

The main test objectives were to verify the following features and functionality:

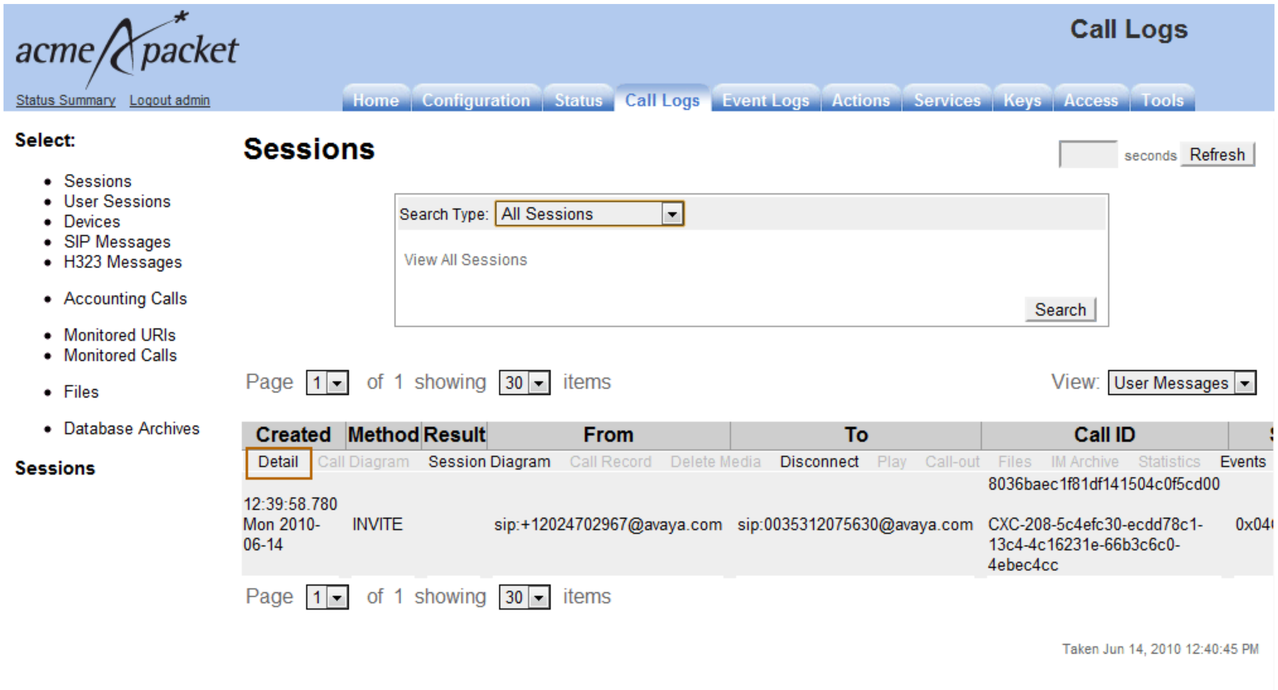
- Inbound and outbound PSTN and Skype P2P service calls from the simulated enterprise site via Skype Connect.
- Basic supplementary telephony features such as hold, resume, transfer, and conference.
- G.729 and G.711 codecs.
- DTMF tone transmission using RFC 2833.
- Inbound Skype Connect service calls that are directly routed to stations, and unanswered, can be covered to Avaya Modular Messaging.
- Long duration calls.

9. Verification Steps

The Avaya Aura™ Session Border Controller stores the SIP signaling traces of each test call in the Call Log database. Log in to the Avaya Aura™ Session Border Controller through the GUI and click on Call Logs.



The Sessions page is displayed. Calls can be filtered by Call ID and called/calling number. Click on **Detail** once a particular call is selected to display the SIP message trace (not shown).



9.1. Troubleshooting Tools

The Communication Manager **list trace station**, **list trace tac**, and **status trunk-group** commands are helpful diagnostic tools to verify correct operation and to troubleshoot problems.
The logging and reporting functions within the Avaya Aura™ Session Border Controller Avaya Aura™ System Manager Common Console may be used to examine the details of SIP calls. In addition, if port monitoring is available, a SIP protocol analyzer such as Wireshark (a.k.a. Ethereal) can be used to capture SIP traces at the various interfaces. SIP traces can be instrumental in understanding SIP protocol issues resulting from configuration problems.

10. Conclusion

As illustrated in these Application Notes, Avaya Aura™ Session Manager R6, Avaya Aura™ Communication Manager R6, Avaya Aura™ Session Border Controller R6 and Avaya Modular Messaging R5.2 can be configured to interoperate successfully with the Skype Connect service. This solution provides users of Avaya Aura™ Communication Manager R6 the ability to support inbound and outbound calls and basic supplementary features over a public SIP trunk to Skype Connect. These Application Notes further demonstrated that the Avaya Aura™ Session Border Controller could be utilized to remove P-Site header information on egress SIP messages to the Skype Connect service as well as provide required domain name conversion for inbound and outbound calls.

The reference configuration shown in these Application Notes is representative of a basic enterprise customer configuration and is intended to provide configuration guidance to supplement other Avaya product documentation.

11. References

The Avaya product documentation is available at http://support.avaya.com unless otherwise noted.

[1] *Configuring Avaya Modular Messaging as a Centralized Messaging Solution for the Avaya CS1000E, Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager - Feature Server & Access Element 5.2.1 – Issue 1.0*

[2] *Administering Avaya Aura™ Session Manager – 09-Aug-2010 - Doc ID 03-603324*

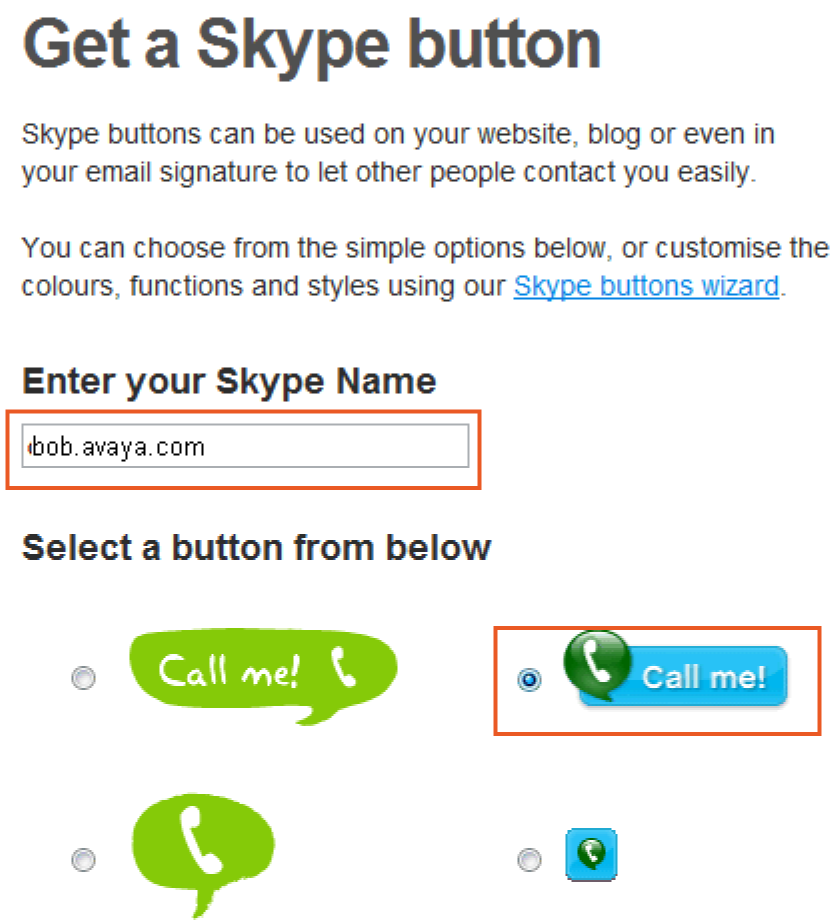
The Skype product documentation is available at http://www.skype.com/intl/en-us/business/ unless otherwise noted.

[3] *Skype for SIP product datasheet, Version 2.0,* 2010.

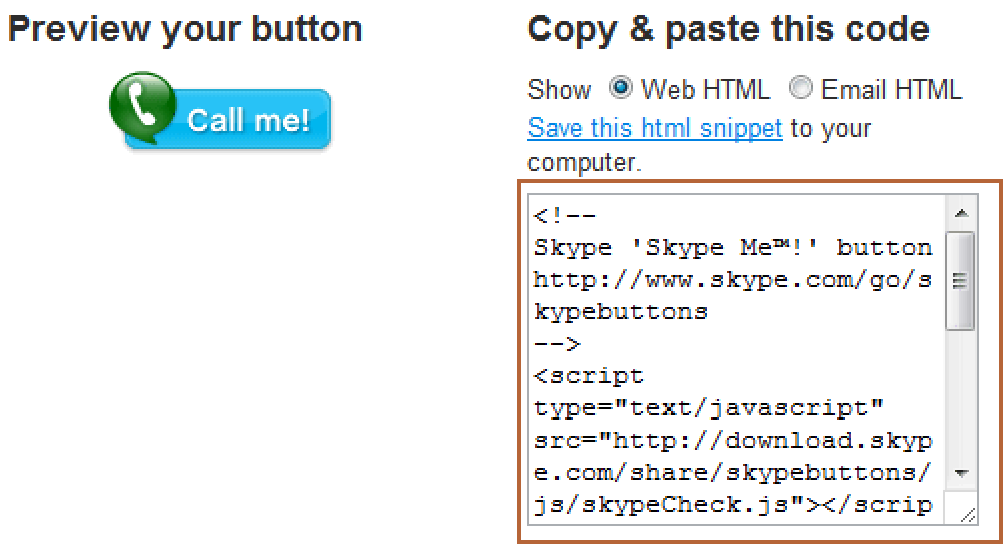
Appendix A

In **Section 6.7** the provisioning of a Business User is discussed. The Business User is associated with a unique Skype Name which can be dialed from the Skype Client application or can be used in click-to-call applications. This section describes the steps for embedding a Skype Button in HTML code to allow Internet users to dial the Skype Name of a Business User.

Go to http://www.skype.com/intl/en-us/tell-a-friend/get-a-skype-button/ . The **Get a Skype Button** page is displayed. Enter the **Skype Name** from **Section 6.7** and select a Skype Button.



Scroll down to the bottom of the page. Copying and pasting the HTML snippet from the text box into an existing webpage will allow Internet users to initiate a call to Business Users.



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