

Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Voice Portal 5.1, Avaya Aura® Session Manager 6.0, Avaya Aura® Session Border Controller 6.0 with Skype Connect™ 2.0 – Issue 1.0

Abstract

These Application Notes describe the steps to configure Avaya Voice Portal 5.1 and the Avaya Aura® R6 SIP reference architecture with the Skype Connect™ SIP trunking service.

Skype Connect™ allows PSTN users and Skype's 600 million registered user community to access the Avaya Voice Portal self-service platform hosted on a secure enterprise network through click-to-call applications 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 Avaya Voice Portal 5.1 and the Avaya Aura® R6 SIP reference architecture with the Skype Connect™ SIP trunking service. Skype Connect™ allows PSTN users and Skype's 600 million registered user community to access the Avaya Voice Portal self-service platform hosted on a secure enterprise network through click-to-call applications 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.

Avaya Voice Portal 5.1 runs on a single Avaya S8800 server and is connected to Avaya Aura® Session Manager over a TCP-based SIP trunk while DTMF is transmitted within the RTP stream using RFC2833 compliant messages. Both Avaya Voice Portal 5.1 and Skype Connect™ 2.0 support G711Mu-law, G711A-law and G729 codecs.

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™, provides telephony features such as Call Forward, Transfer and Call Pickup as well as Automatic Call Distribution for phones belonging to a contact center agent group. 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.

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1.1. Design Principles and Assumptions

The service offer described in these Application Notes is designed for business customers using Avaya Voice Portal, Avaya Aura® Communication Manager and Avaya Aura® Session Manager on a private protected enterprise network who wish to provide their customers and partners with publicly routable E.164 self-service access numbers and/or Skype Names that could be used to access the Avaya Voice Portal self-service platform from the Skype P2P Network through the Internet. It is 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. A basic dial-plan was provisioned on the Avaya Aura® Session Border Controller in the sample configuration to allow it to handle REFER and INVITE with REPLACES SIP messages. Call routing and number modification logic was configured on Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Hardware, software resilience and failover between the various Avaya components are not covered in these Application Notes.

1.2. Interoperability Compliance Testing

The compliance testing was based on a test plan provided by the Avaya Solution and Interoperability Test Lab and the following features were tested as part of this effort:

- Avaya Voice Portal access from the PSTN and the Skype P2P Network
- Basic call scenarios using G.711 and G.729 codecs
- DTMF transmission
- Blind Transfer from Avaya Voice Portal to Avaya Aura® Communication Manager
- Consultative transfer from Avaya Voice Portal to Avaya Aura® Communication Manager using the following two SIP methods:
    - INVITE with REPLACES
    - REFER
- Bridged Transfer from Avaya Voice Portal to Avaya Aura® Communication Manager

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
DoS	Denial of Service
PE	Processor Ethernet
P2P	Peer-to-peer
AOR	Address of record
DNIS	Dialed Number Identification Service

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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.

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1.5. Reference Configuration

Figure 1 illustrates how the Avaya Aura® R6 SIP reference architecture, Avaya Voice Portal 5.1 and Skype Connect™ 2.0 were used for the 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 HQ site an Avaya Aura® Session Border Controller acts as a B2BUA for SIP calls and provides the following services:

- Topology hiding through the termination and re-origination of SIP calls using its own inside (private) and outside (public) IP addresses.
- Network security through its DoS and packet filtering module.
- SIP signaling channel conversion from UDP to TCP for inbound and vice-versa for outbound calls.
- REFER message interception and conversion in call flows that involve blind or bridged transfer of PSTN or Skype P2P callers to an ACD agent on Avaya Aura® Communication Manager as Skype Connect™ 2.0 does not support the REFER method (RFC 3515).
- REPLACES header (RFC 3891) support for call flows that involve consultative transfer of PSTN or Skype P2P callers to an ACD agent on Avaya Aura® Communication Manager.

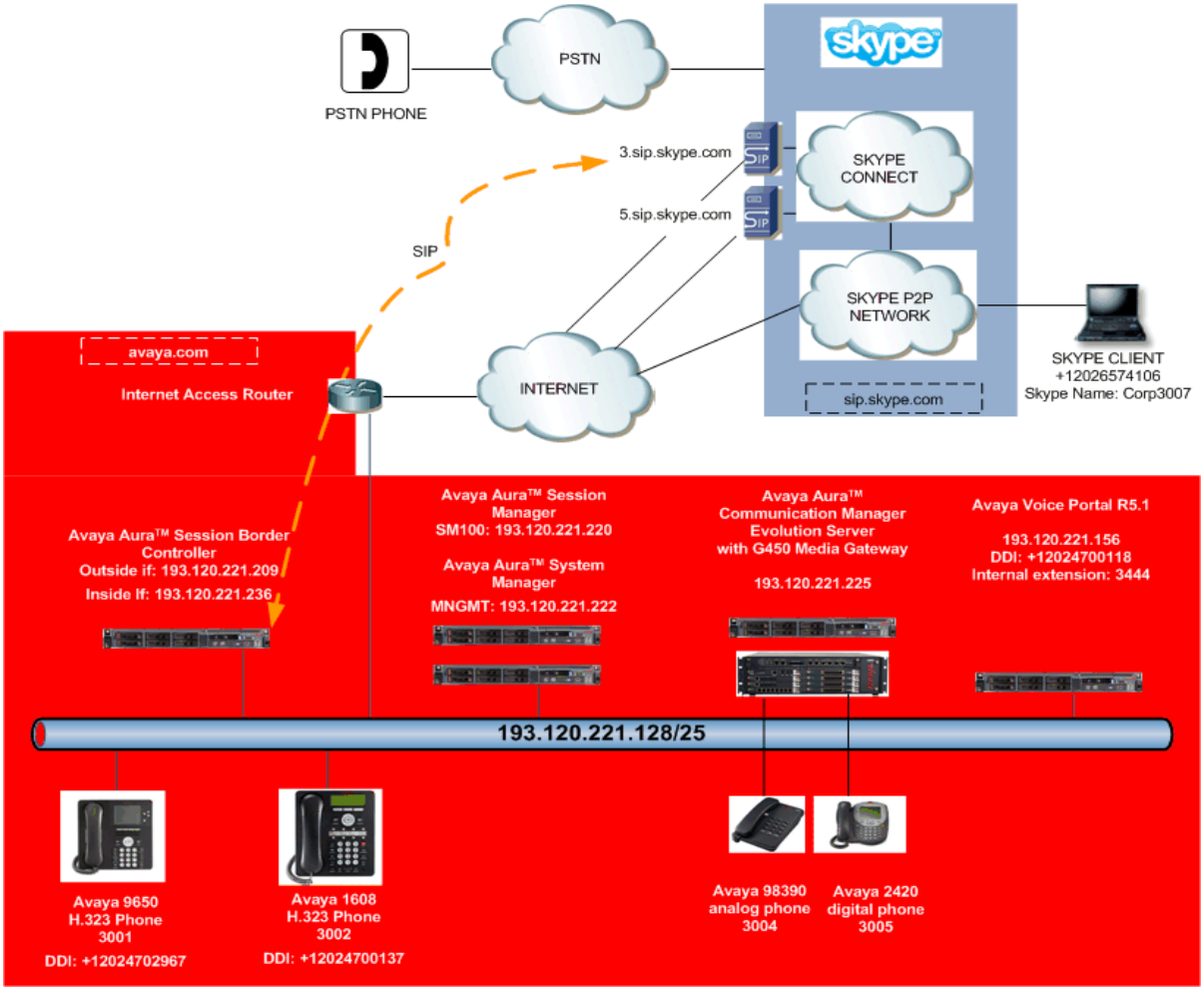


Figure 1: Avaya Interoperability Test Lab Reference Configuration

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On the private enterprise network Avaya Aura® Session Manager acts as a SIP proxy and provides the following services:

- Routing of both On-net and Off-net calls using a centralized dial plan.
- E.164 to internal number normalization using the DigitConversionAdapter Adaptation module where the 11 digit public DDIs are mapped to 4 digit internal extensions.

Avaya Aura® Communication Manager provides the following services:

- Telephony features and functionality for test phones and contact center agents.
- ACD functionality using vectors and VDNs.
- Media services such as call processing tones and music on hold. Detailed administration of Avaya Aura® Communication Manager will not be described in this Application Note (see the appropriate documentation listed in **Section 11).**

Avaya Voice Portal provides the following services:

- Blind, bridged and consultative (aka supervised) call transfers to/from PSTN callers, PBX users and ACD agents.
- Text to Speech and Automatic Speech Recognition for inbound Skype Connect™ and P2P network calls through the use of an external (Nuance) TTS and ASR server.
- DTMF recognition.
- UUI data transfer to ACD agents.

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

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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 – R6.0.0.0.4
Avaya S8800 Server	Avaya Aura® Communication Manager – Evolution Server - R016x.00.0.345.0
Avaya S8510 Server	Avaya Voice Portal R5.1 Build 5.1.0.0.4201
Avaya G450 Media Gateway	R30.12.1
Avaya 9650 H.323 Phone	R3.1.1
Avaya 1608 H.323 Phone	R1.2
Avaya 2420 Digital Phone	N/A
Avaya 98390 Analog Phone	N/A
Test PC1	Nuance Realspeak 4.5
Test PC2	Skype Client v4.2.0.169
Skype Connect™	Version 2.0

Table 1: Equipment and Software Used in the Reference Configuration

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3. Call Flows

To understand how inbound and outbound calls are handled by enterprise network elements two general call flows are described in this section.

3.1. Blind Transfer of PSTN caller to Internal Extension using the REFER method

The first call scenario illustrated in **Figure 2** is a blind transfer of a PSTN caller from Voice Portal to an internal extension on Communication Manager. Please see **Appendix C** for the ladder diagram of the following call.

1. PSTN user dials the Voice Portal self-service DDI (+12024700118).
2. The Voice Portal self-service DDI was provisioned in Skype Manager as an Online Number associated with the HQ site's SIP Profile therefore the Class 4 PSTN switches on the SS7 network route the call to the entry point into the Skype Connect service network.
3. The SS7 call is converted into SIP before it is sent to the publicly routable IP address of the outside interface of the Session Border Controller using the Skype Connect service.
4. The call is delivered to the Session Border Controller using SIP over UDP.
5. The Session Border Controller performs SIP header modifications, transport protocol conversion (UDP to TCP) and routes the call to Session Manager.
6. Session Manager performs number normalization by converting the Voice Portal self-service DDI to the Voice Portal internal extension (+12024700118 to 3444) using the adaptation module and sends the call to the Voice Portal MPP.
7. The PSTN caller is connected to Voice Portal, a voice menu prompt is played and the caller is requested to input a target extension (3001) for blind transfer. The target extension is an Avaya 9650 H.323 phone registered to Communication Manager. Voice Portal executes the blind transfer by sending a REFER request as part of a new dialog to Session Manager with the target extension in the Refer-To header.
8. Session Manager proxies the REFER to the Session Border Controller.
9. The Session Border Controller is programmed to handle the REFER locally by sending an INVITE to Communication Manager with the target extension number in the Request URI as Skype Connect 2.0 does not currently support call transfers using the REFER method. When Communication Manager receives the INVITE, it rings the target extension while it is answered. Finally the PSTN caller hangs up the call.

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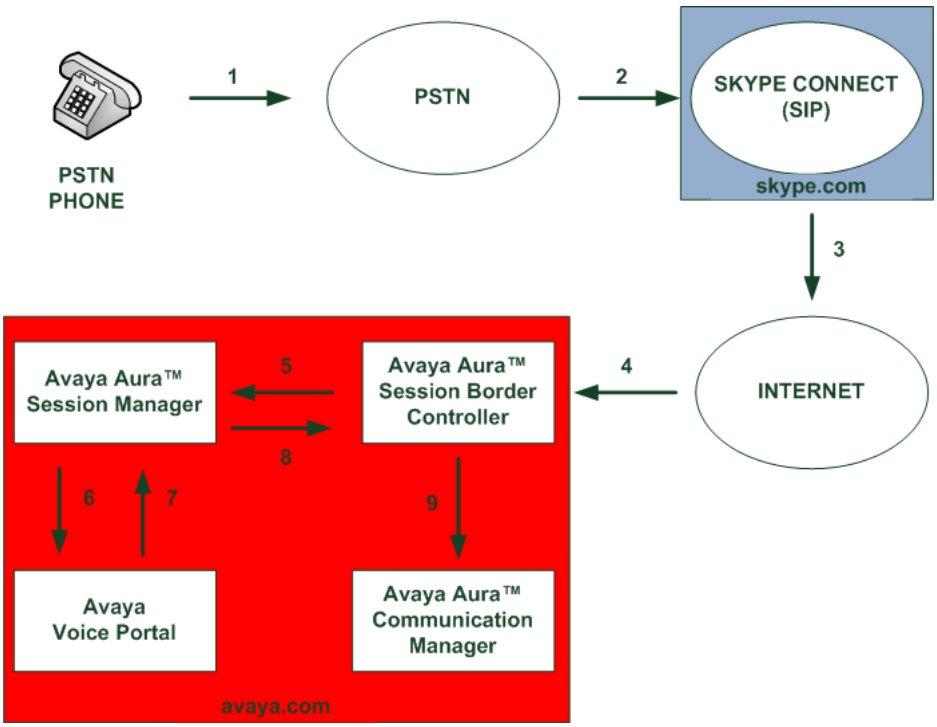


Figure 2: Blind Transfer of PSTN caller to Internal Extension

3.2. Consultative Transfer of Inbound Skype P2P Client to Internal Extension using INVITE with REPLACES

The second call scenario illustrated in **Figure 3** is an inbound call from a Skype Client on the Internet, registered to a super node on the Skype P2P network. A Skype Name (cc.avaya.com) was created and associated with the Voice Portal self-service access number using Skype Manager. Please see **Appendix C** for the ladder diagram of the following call.

1. The Skype Client user initiates a call to a Skype Name associated with Voice Portal self-service access number (3444).
2. The call is routed from the Skype P2P Network to the Skype Connect service network.
3. The call is sent to the Avaya Aura® Session Border Controller using SIP over UDP.
4. The Avaya Aura® Session Border Controller performs SIP header modifications, transport protocol conversion (UDP to TCP) and routes the call to Session Manager.
5. Session Manager sends the call to Avaya Voice Portal as no called number normalization is required (3444 was sent from Skype Connect in the Request-URI).
6. The caller is connected to Voice Portal, a voice menu prompt is played and the caller is requested to input a target extension (3001) for consultative transfer. The target extension is an Avaya 9650 H.323 phone registered to Communication Manager.
7. Voice Portal executes the consultative transfer by sending an INVITE to Session Manager with the target extension in the Request-URI header.
8. Session Manager routes the call to the target extension on Communication Manager. An additional call leg is established when the target extension answers the call. It is important to note that the codec does not change from what was initially negotiated between Skype Connect and Voice Portal.

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9. Voice Portal attempts to send a REFER message back to the Skype Connect network via Session Manager with the Skype P2P Clients SIP handle in the Request-URI <sip:Anonymous@avaya.com> and the target extension (3001) in the Refer-To header.
10. The REFER request is proxied to the Avaya Aura® Session Border Controller.
11. The Avaya Aura® Session Border Controller intercepts the REFER as it is not supported by Skype Connect and converts it into an INVITE request with a REPLACES header as per RFC3891) which includes the dialog identifier (Call-ID, To-tag, From-tag) of the SIP dialog that was created in **steps 7 and 8**.
12. Avaya Aura® Session Border Controller sends a re-INVITE to the Skype Connect network to allow the Skype P2P Client to join the call leg created in **step 11.** At this stage the signaling and audio path is established between the Skype P2P Client (caller) and the internal extension on Communication Manager.

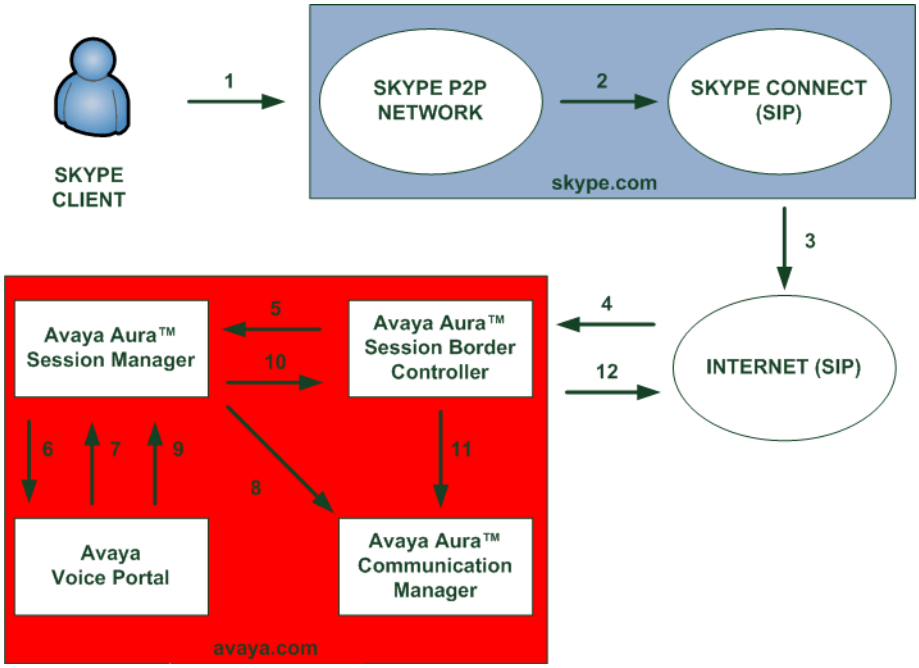


Figure 3: Consultative Transfer of Skype Client to Internal Extension

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4. Configure Avaya Aura® Session Manager

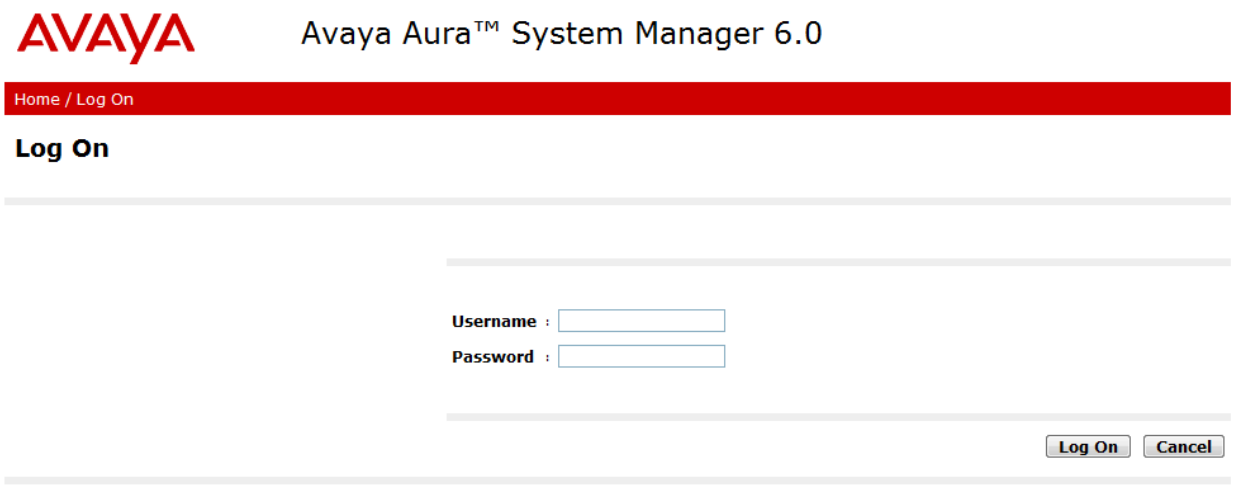
This section provides the procedures for configuring the Session Manager and includes the following items:

- 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 [1]**.

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

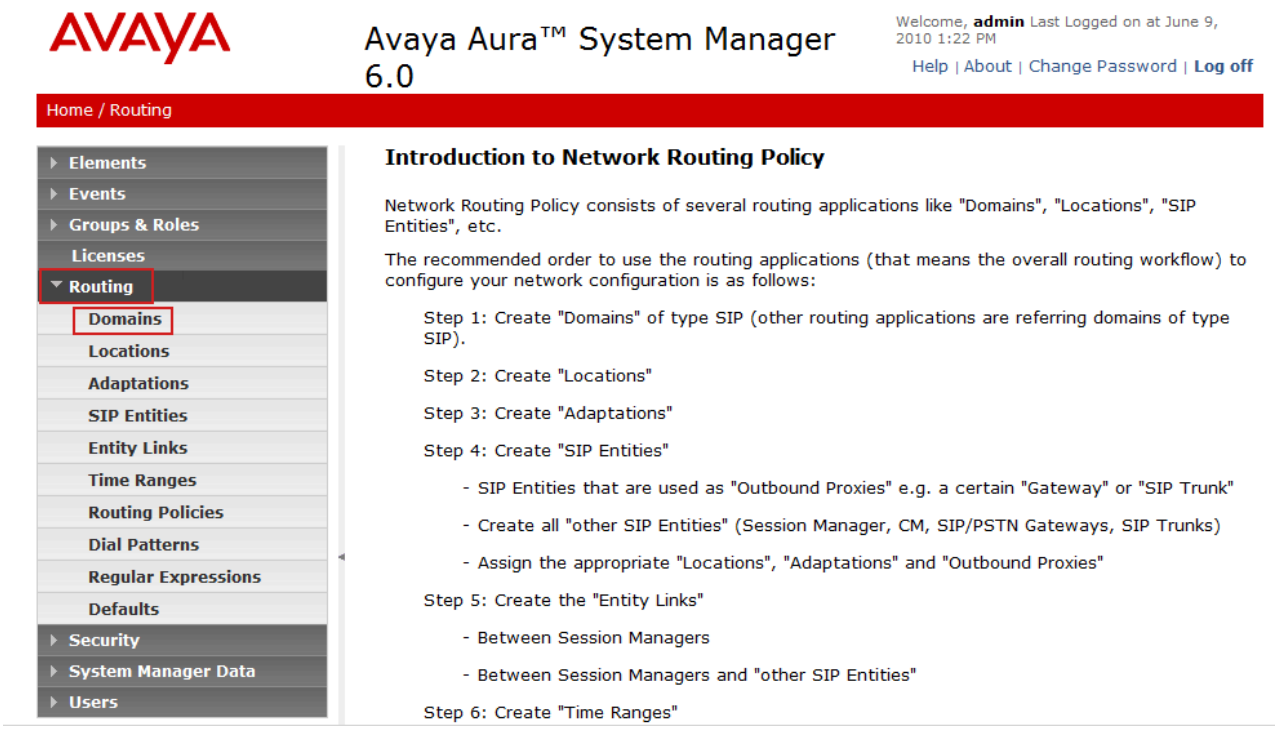
Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL "**https://<ip-address>/SMGR**", where "<ip-address>" is the IP address of System Manager. Log in with the appropriate credentials.



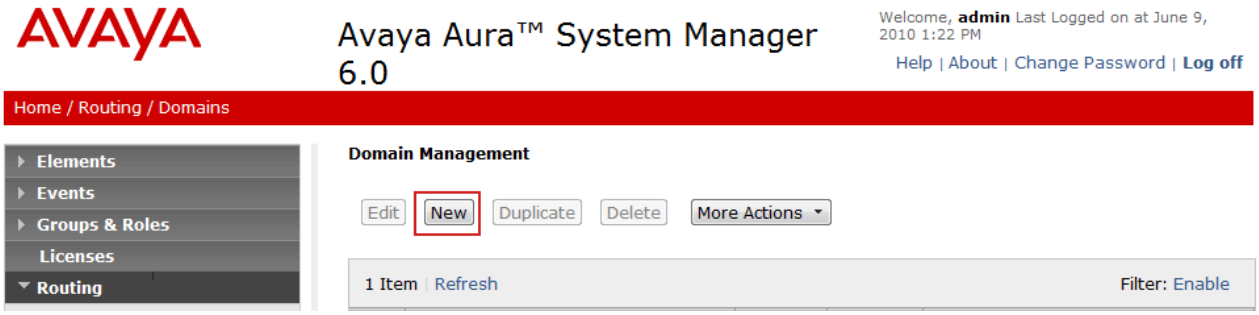
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4.2. Administer SIP Domain

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

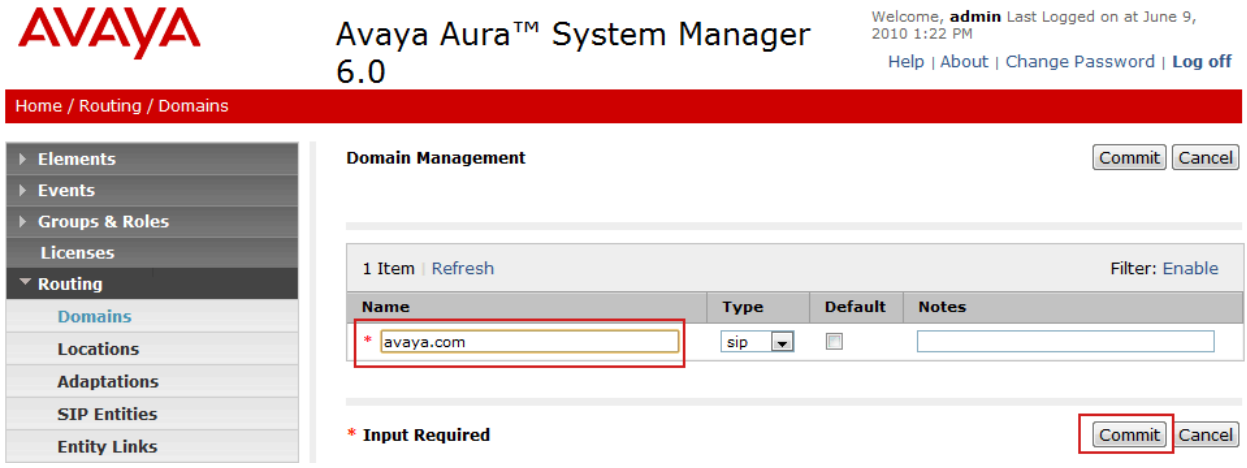


Click **New.**



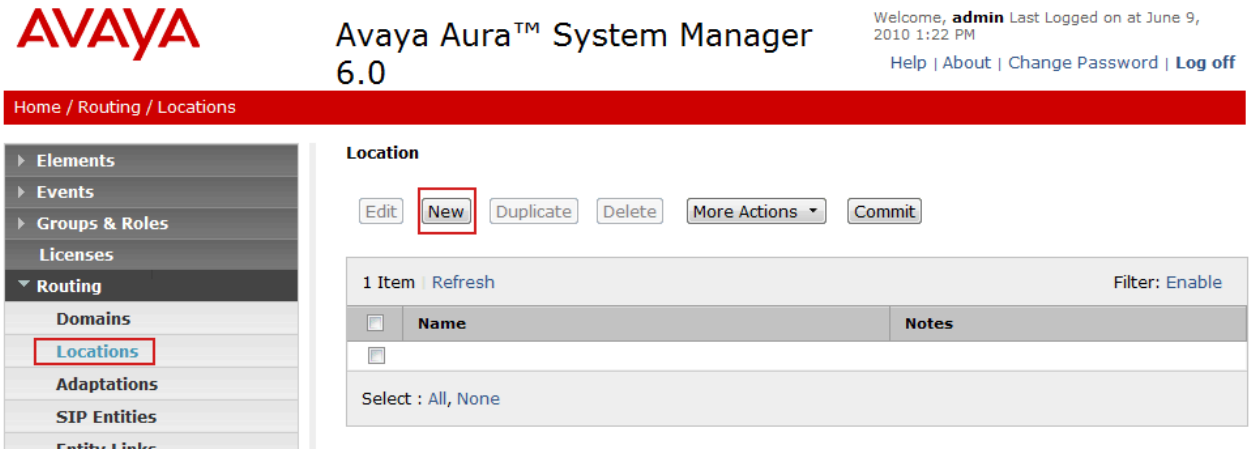
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On the **Domain Management** screen under **Name** add a descriptive name. Retain the default values for the remaining fields. Click **Commit** to save.



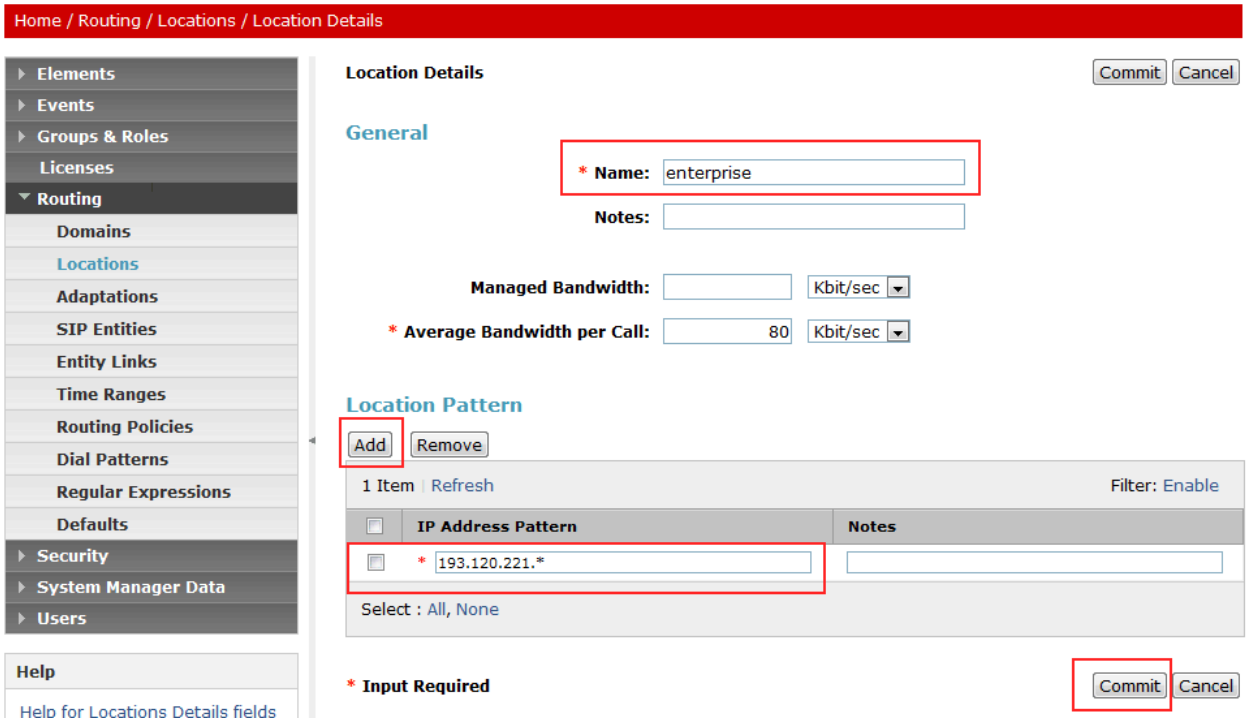
4.3. Define a Location

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.**



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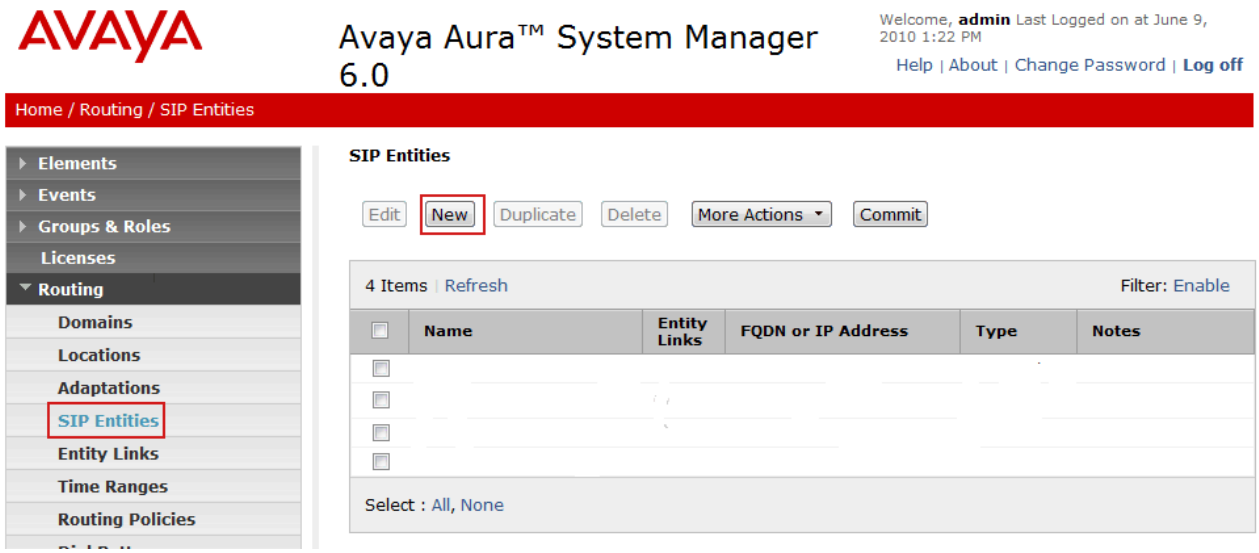
In the **General** section, under **Name** add a descriptive name. Click **Add**. In the **Location Pattern** section under **IP Address Pattern** enter IP address of the subnet on which the SIP Entities (CM and AASBC) reside. Click **Commit** to save.



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4.4. Define SIP Entities

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

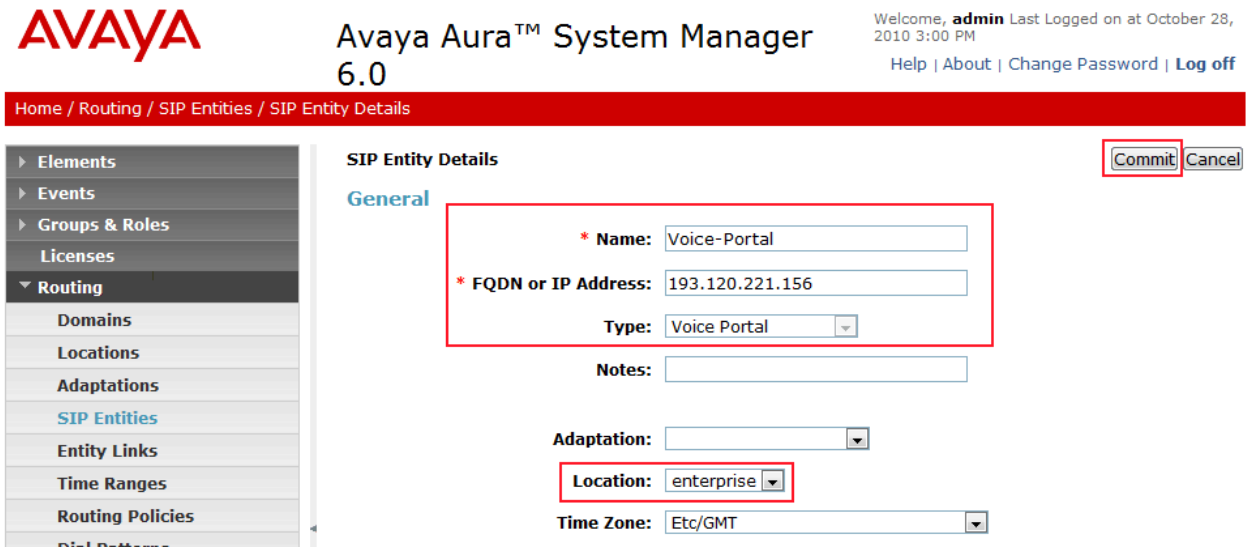


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The **SIP Entity Details** screen is displayed.
Under **General**:

- **Name** Type in a descriptive name
- **FQDN or IP Address** Type IP address of Voice Portal
- **Type** Select **SIP Trunk**
- **Notes** (Optional) Type in description
- **Location** Select the Location created in **Section 4.3**

Click **Commit**.

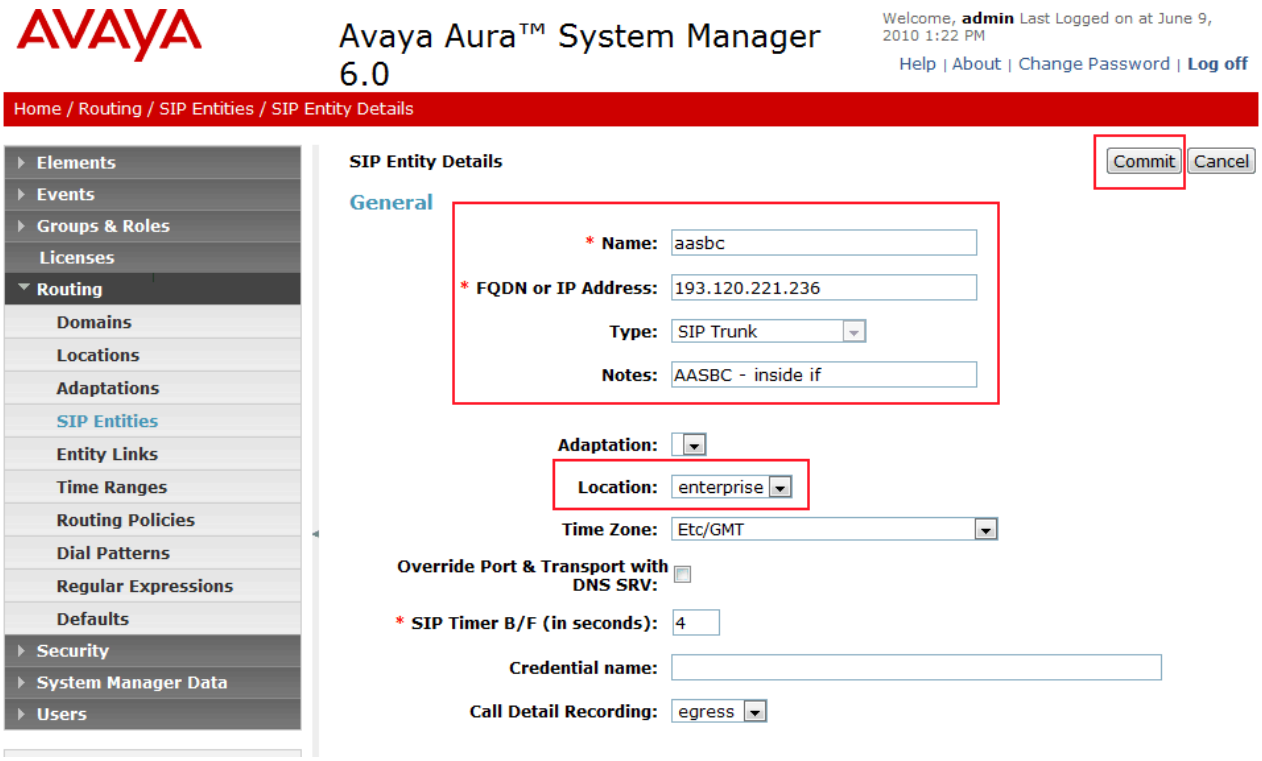


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Repeat the steps from the previous section. Click **New** on the **SIP Entities** page. The following screen shows addition of the Session Border Controller as a SIP entity.
Under **General**:

- **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 **Section 4.3**

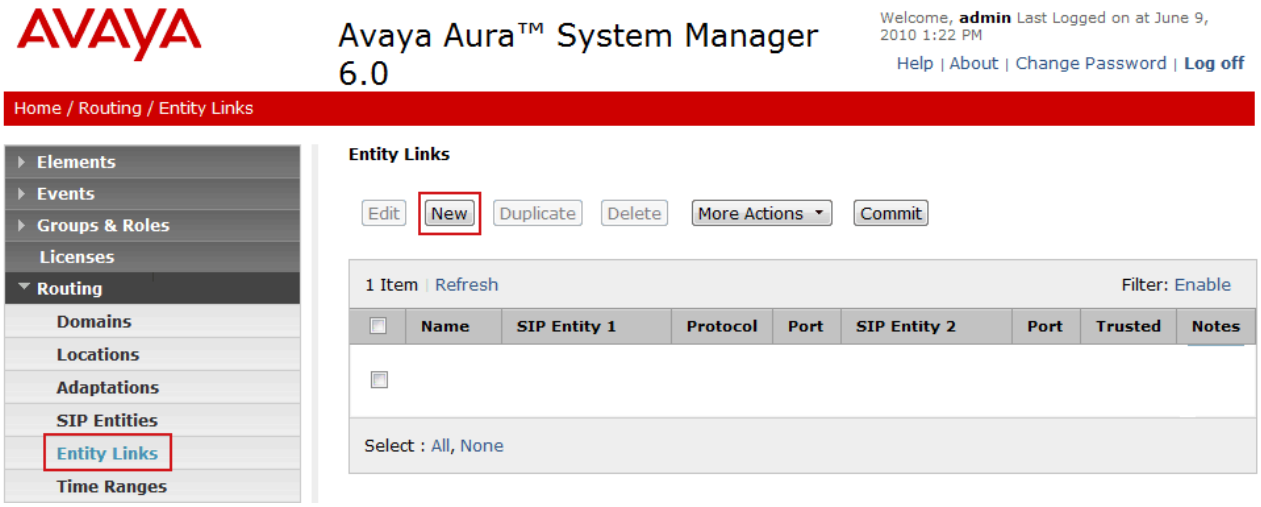
Click **Commit**.



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4.5. Define Entity Links

SIP trunks between Session Manager and SIP Entities are described by Entity Links. To add an Entity Link for Voice Portal, select **Entity Links** on the left pane then click **New**.

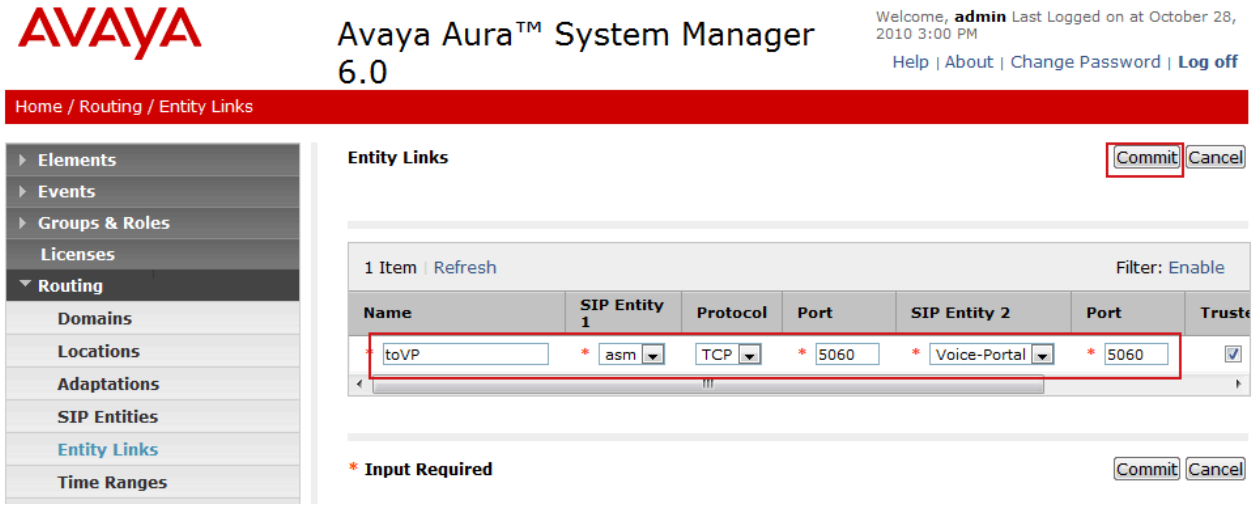


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The **Entity Links** screen is displayed. Fill in the following fields in the new row that is displayed:

- **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 Voice Portal SIP Entity
- **Port:** Type **5060**. Voice Portal listens for SIP requests on TCP port 5060.
- **Trusted:** Check this box, otherwise SIP calls will be denied to/from Voice Portal

Click **Commit**.

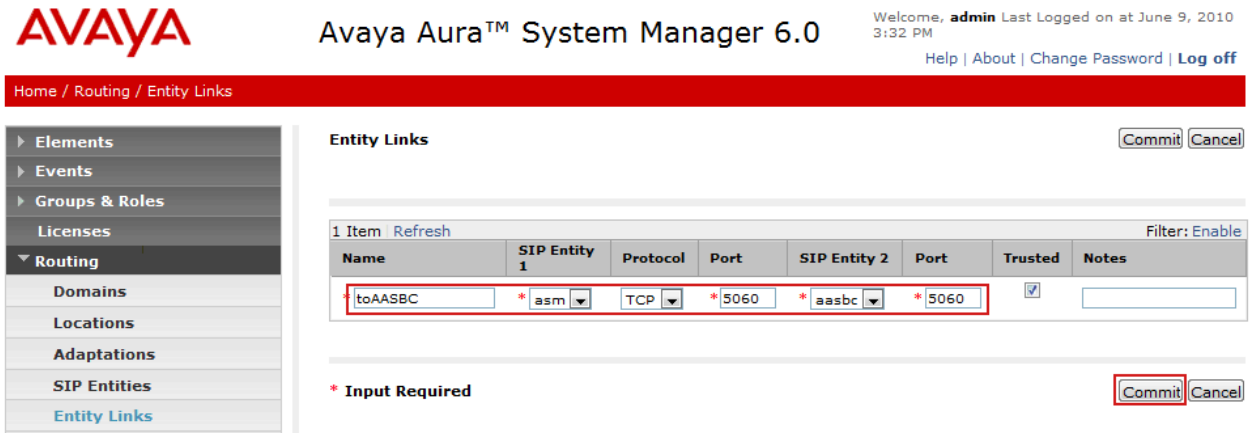


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Repeat the steps from the previous section. Click **New** on the **Entity Links** page to add a link to the Session Border Controller. The **Entity Links** screen is displayed. Fill in the following fields in the new row that is displayed:

- **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
- **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 Avaya Aura® Session Border Controller

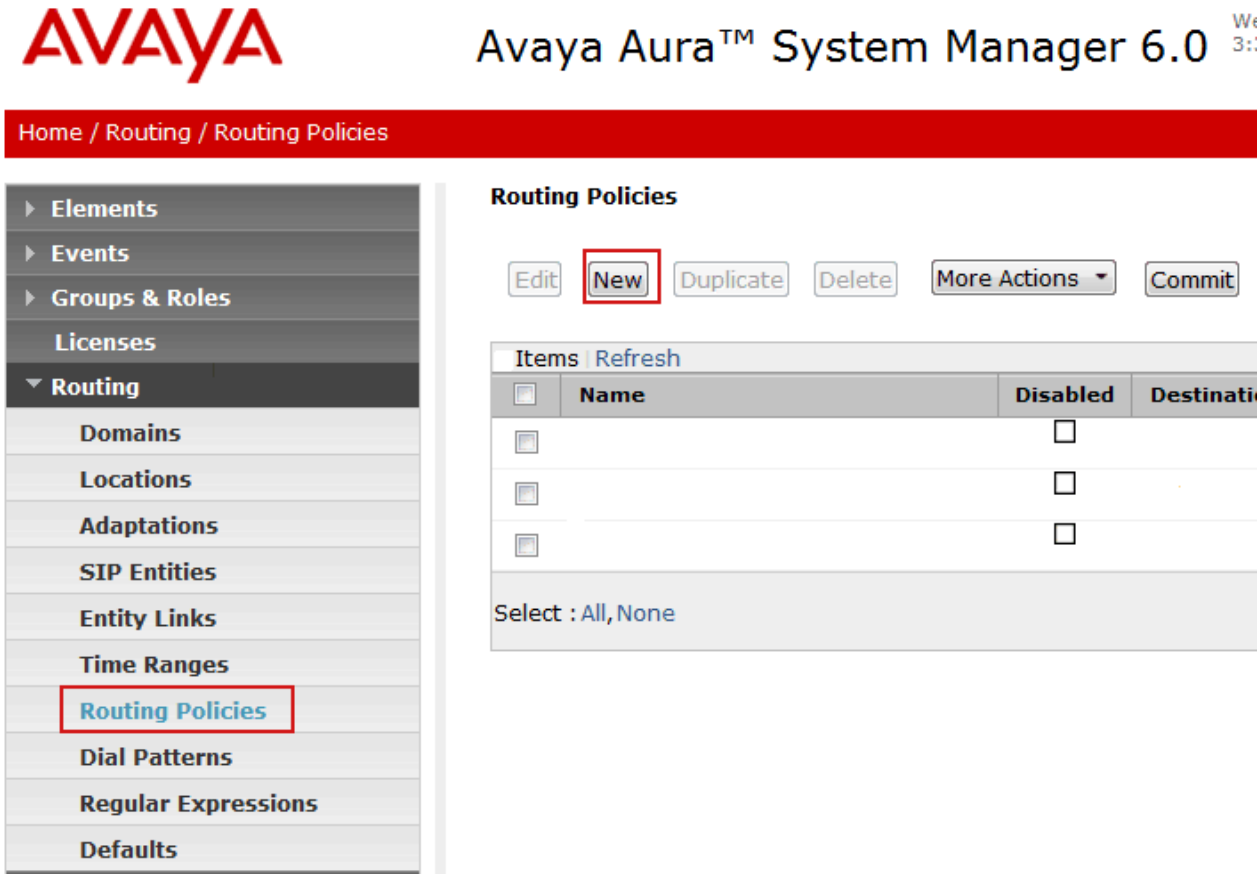
Click **Commit**.



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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 Voice Portal, select **Routing Policies** on the left pane then click **New**.



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

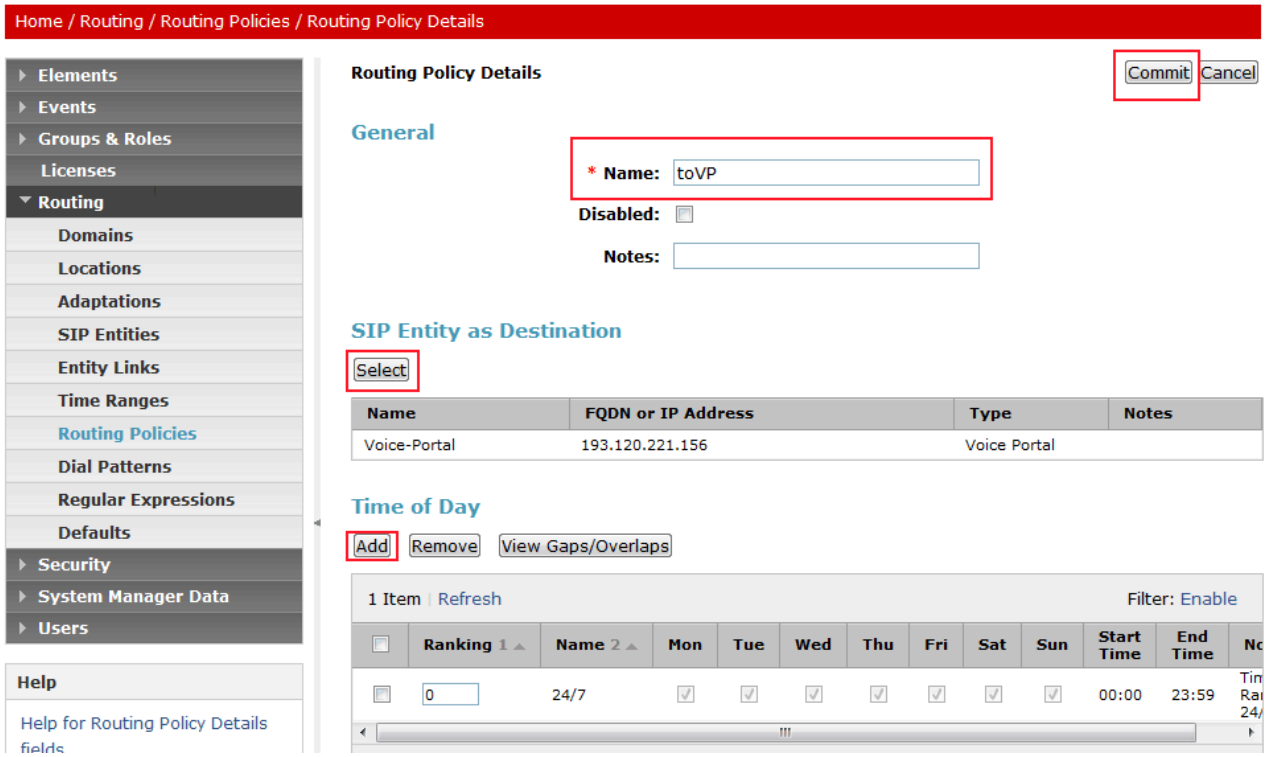
- **Name** Descriptive name.

Under **SIP Entity as Destination**:

- Click **Select**, and then select the Voice Portal 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.**



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

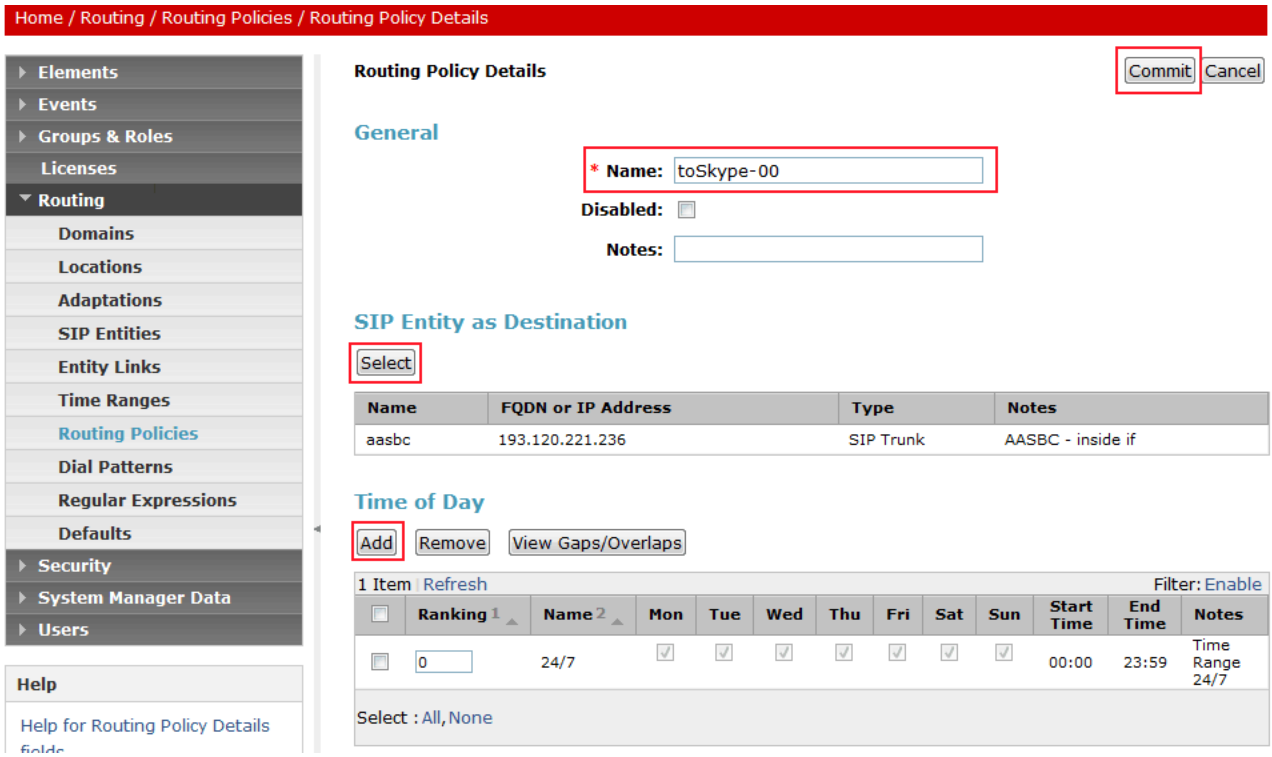
- **Name** Descriptive name.

Under **SIP Entity as Destination**:

- Click **Select**, and then select the Session Border Controller SIP Entity to which this routing policy applies.

Under **Time of Day**:

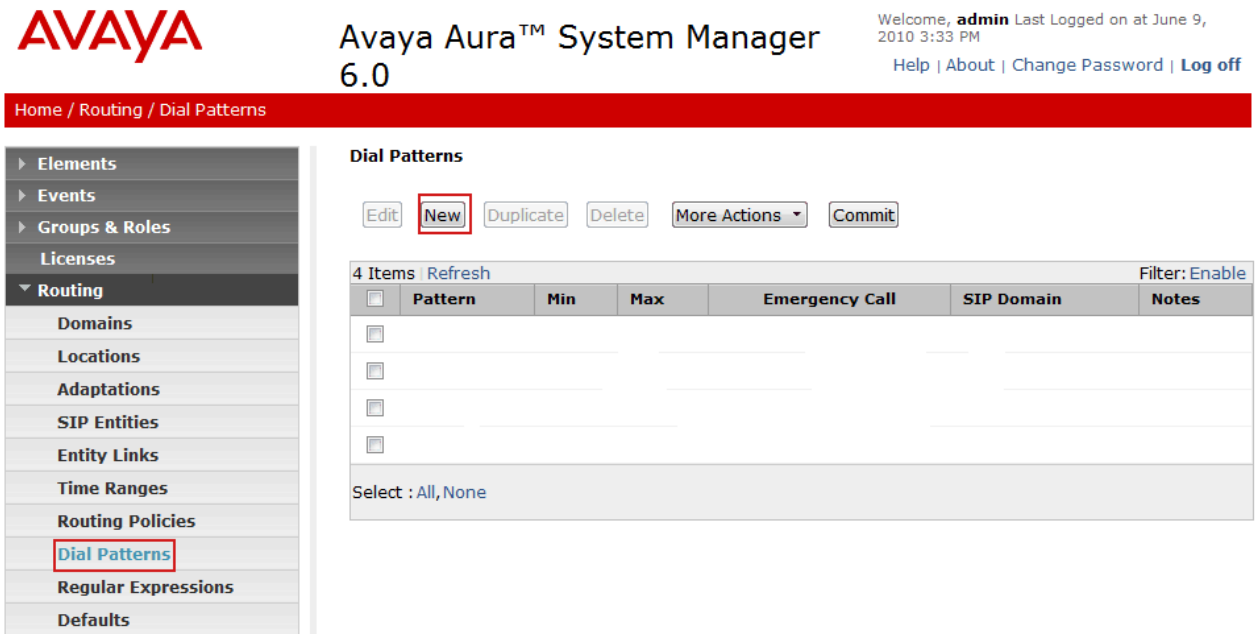
- Click **Add**, and select the default **24/7** time range. Defaults can be used for the remaining fields. Click **Commit.**



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4.7. Define Dial Patterns

Define dial patterns to direct calls to the appropriate SIP Entity. Calls to 3444 and 12024700118 are routed to Voice Portal. To add a dial pattern, select **Dial Patterns** on the left pane then click **New**.



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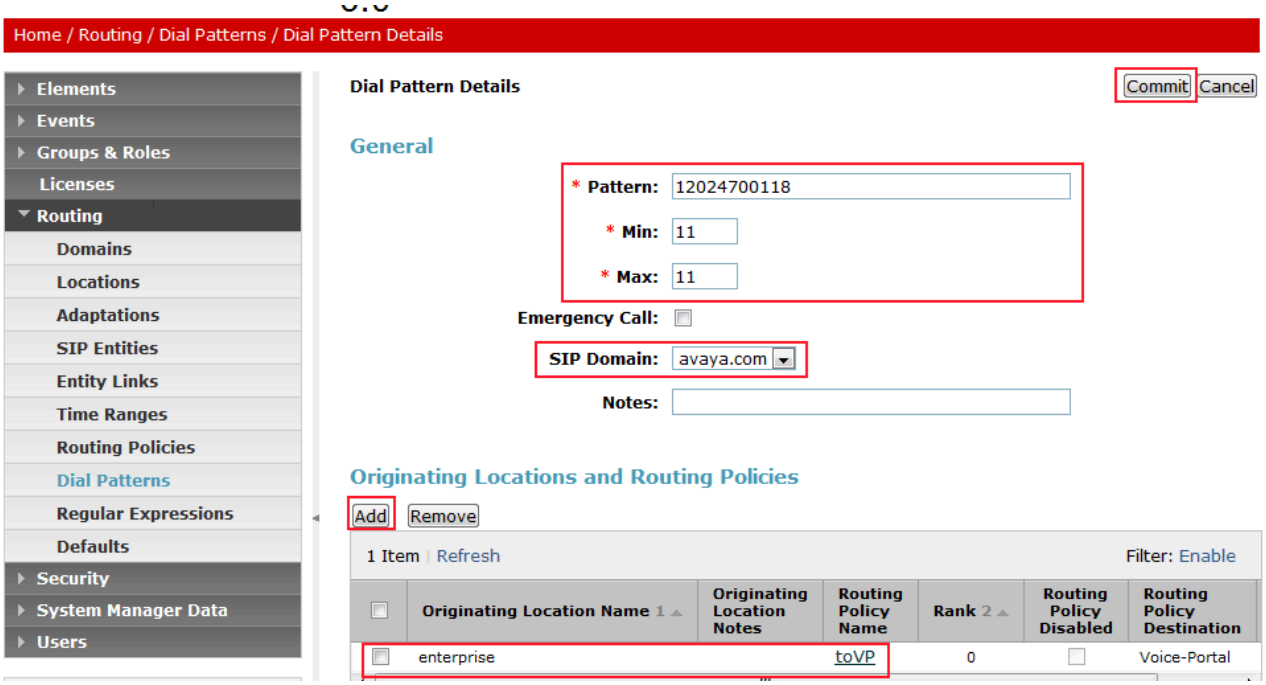
The **Dial Pattern Details** screen is displayed. Under **General** fill in the following fields to add an inbound dial pattern from Skype Connect:

- **Pattern:** Type **12024700118** as PSTN users would be dialling this number to access the Voice Portal.
- **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 **toVP,** the Routing Policy created in **Section 4.6**

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



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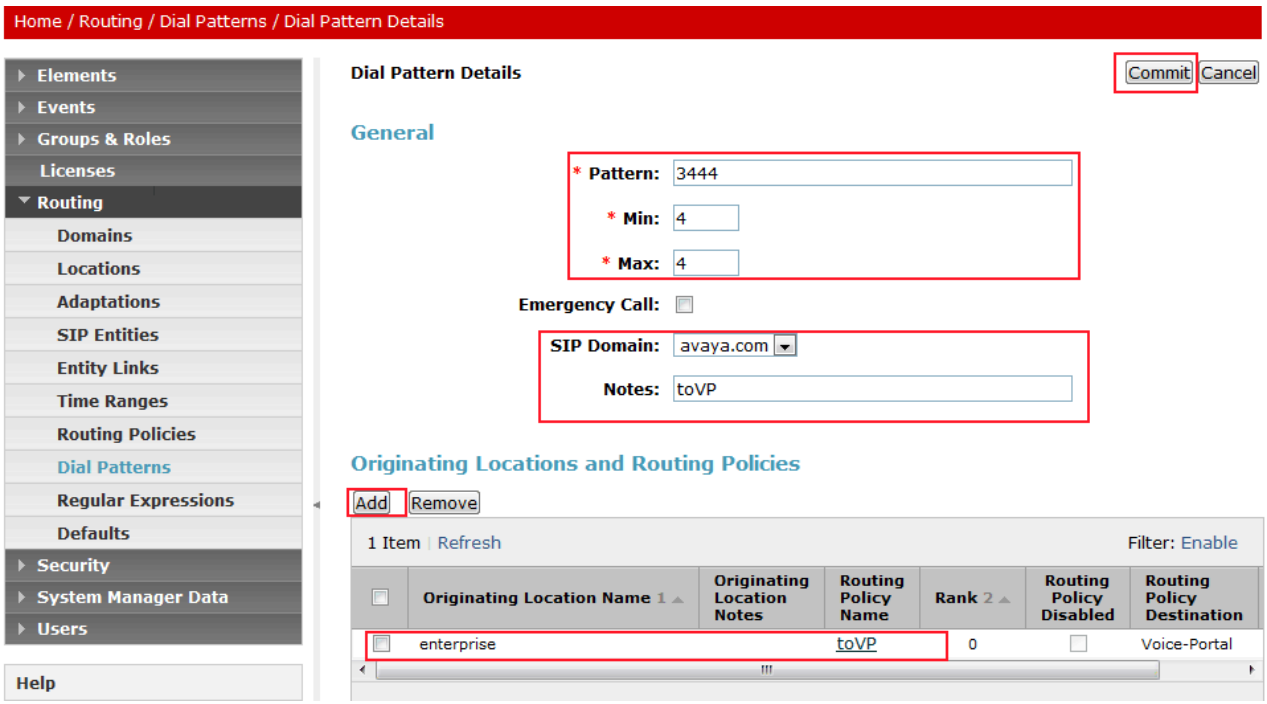
Repeat the steps from the previous section to add an inbound dial pattern for users calling from Avaya endpoints registered to Communication Manager and the Skype P2P Network and until the **Dial Pattern Details** screen is displayed.

- **Pattern:** Type **3444** as this was used to associate an extension to the cc.avaya.com Skype Name in **Section 5.8**
- **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 **toVP,** the Routing Policy created in **Section 4.6**

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



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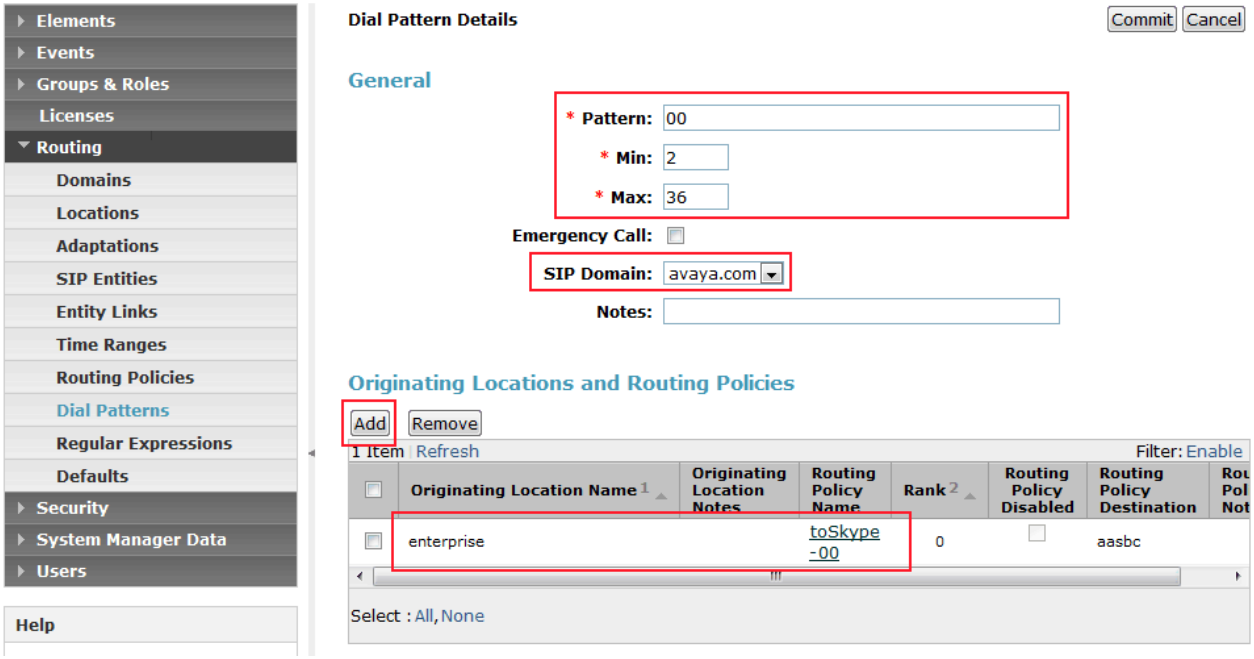
Repeat the steps from the previous section to add an outbound dial pattern to Skype Connect until the **Dial Pattern Details** screen is displayed.

- **Pattern:** Type **00** as the International numbering format is used outbound calls 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.



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5. Configure Skype Manager

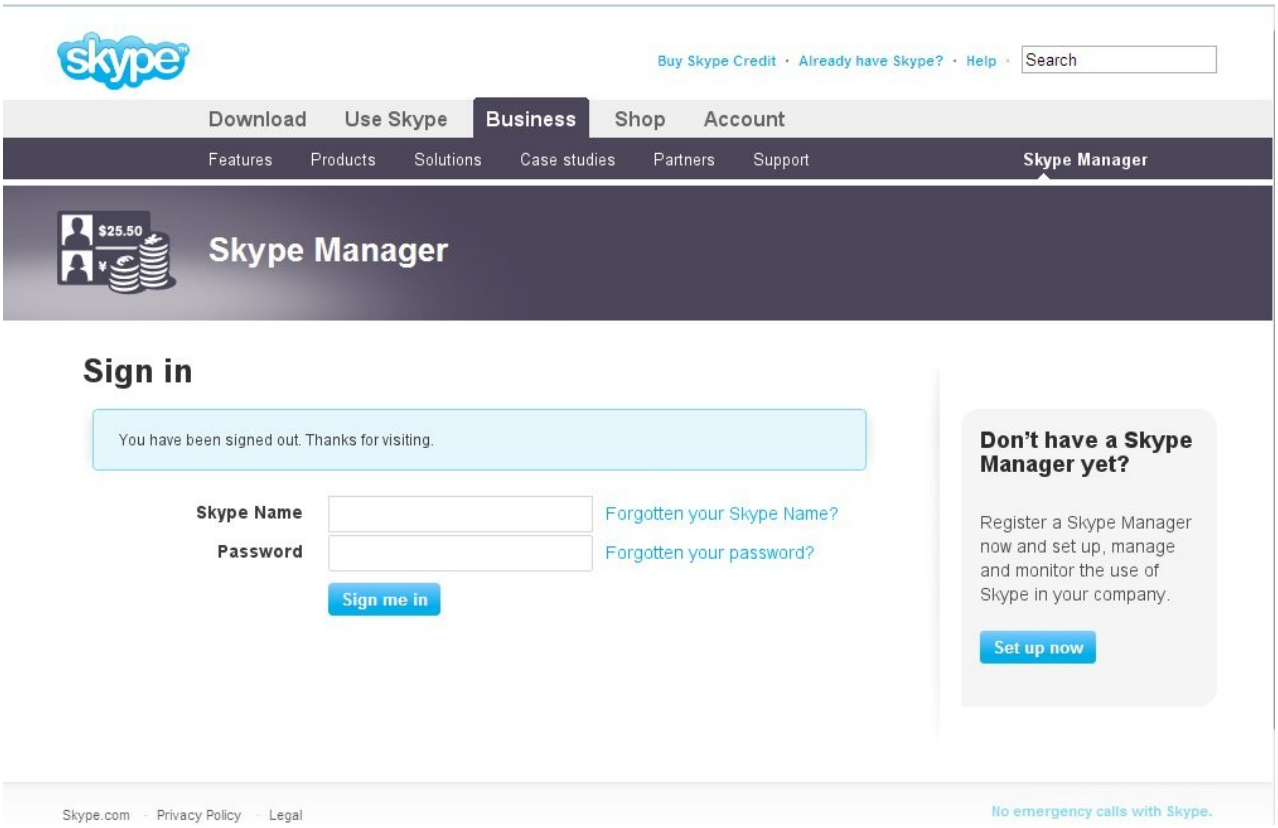
This section provides the procedures for configuring Skype Manager and includes the following items:

- 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.

5.1. Log in to Skype Manager

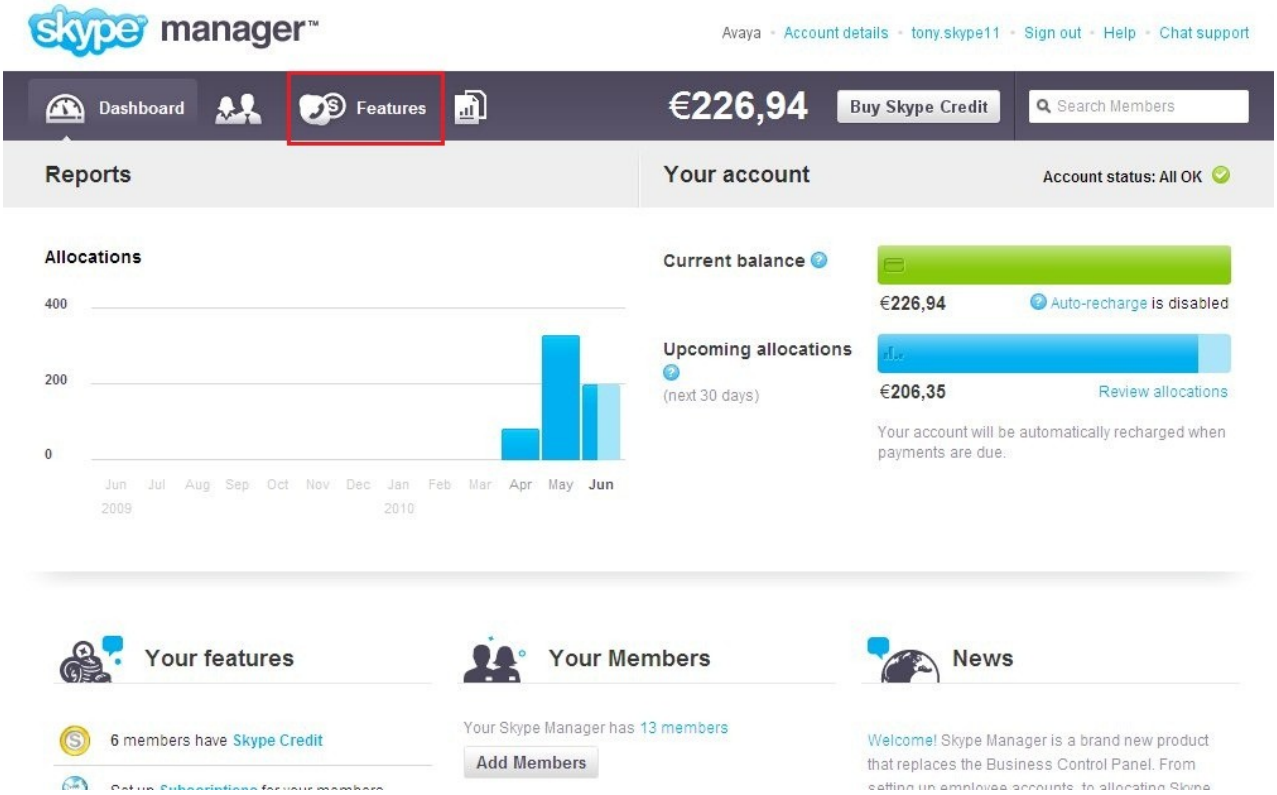
Configuration of a new SIP Profile and Business Users is accomplished by accessing the browser-based GUI of Skype Manager at **http://manager.skype.com**. Log in with the appropriate credentials.



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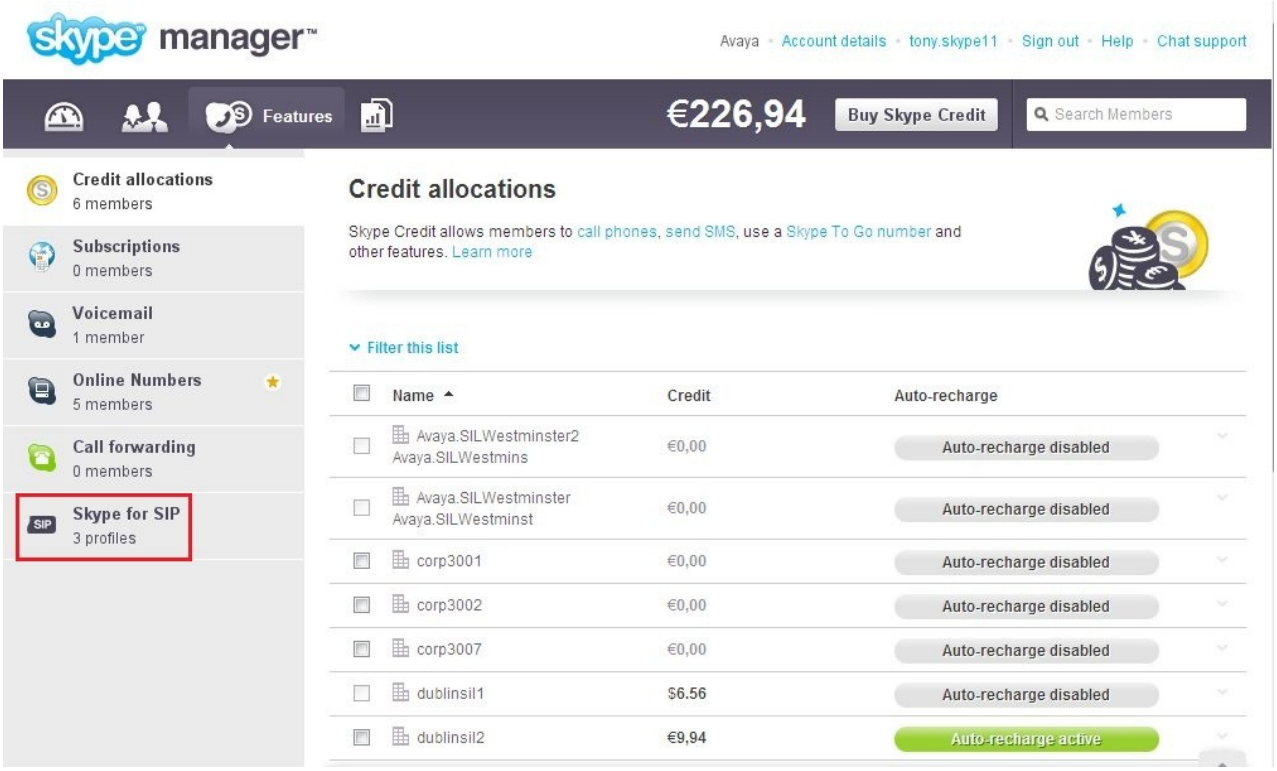
5.2. Create SIP Profile

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**.

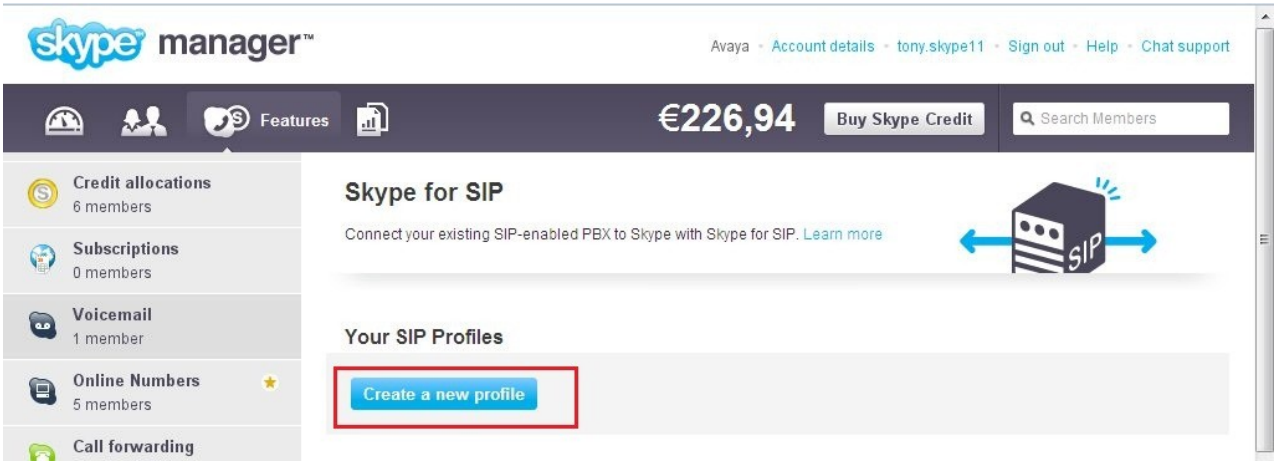


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The **Features** page is displayed. Click **Skype for SIP** on the left pane.

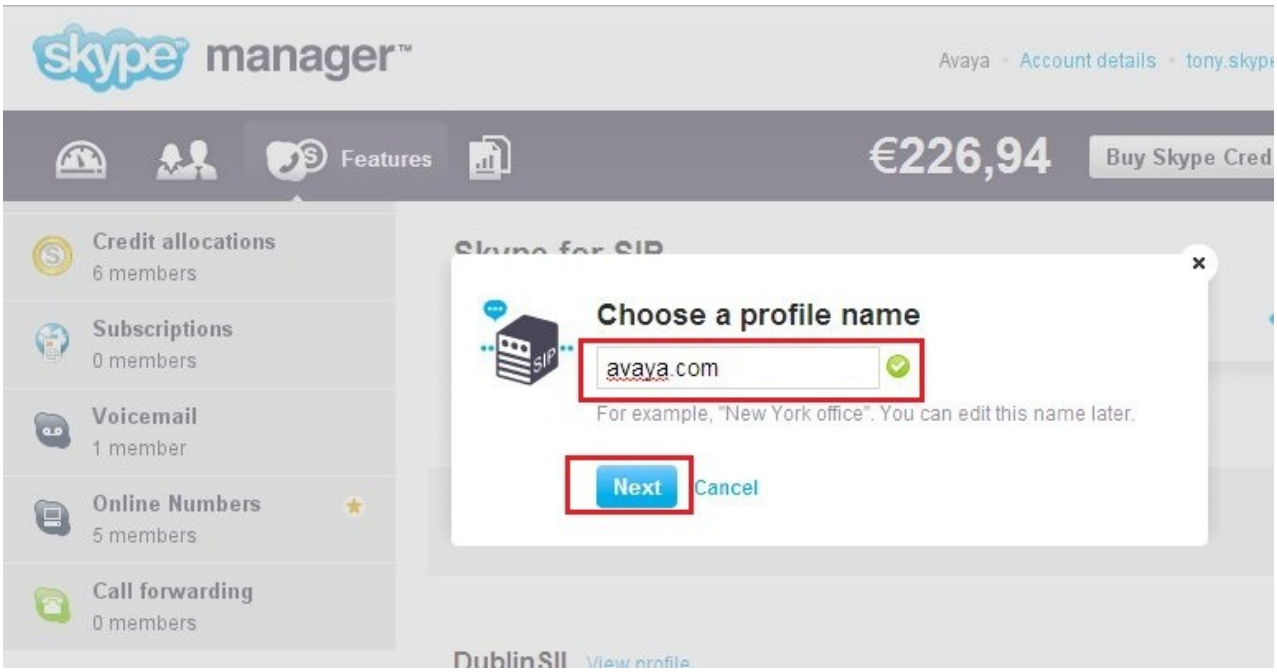


The **Skype for SIP** page is displayed. Click **Create a new profile**.



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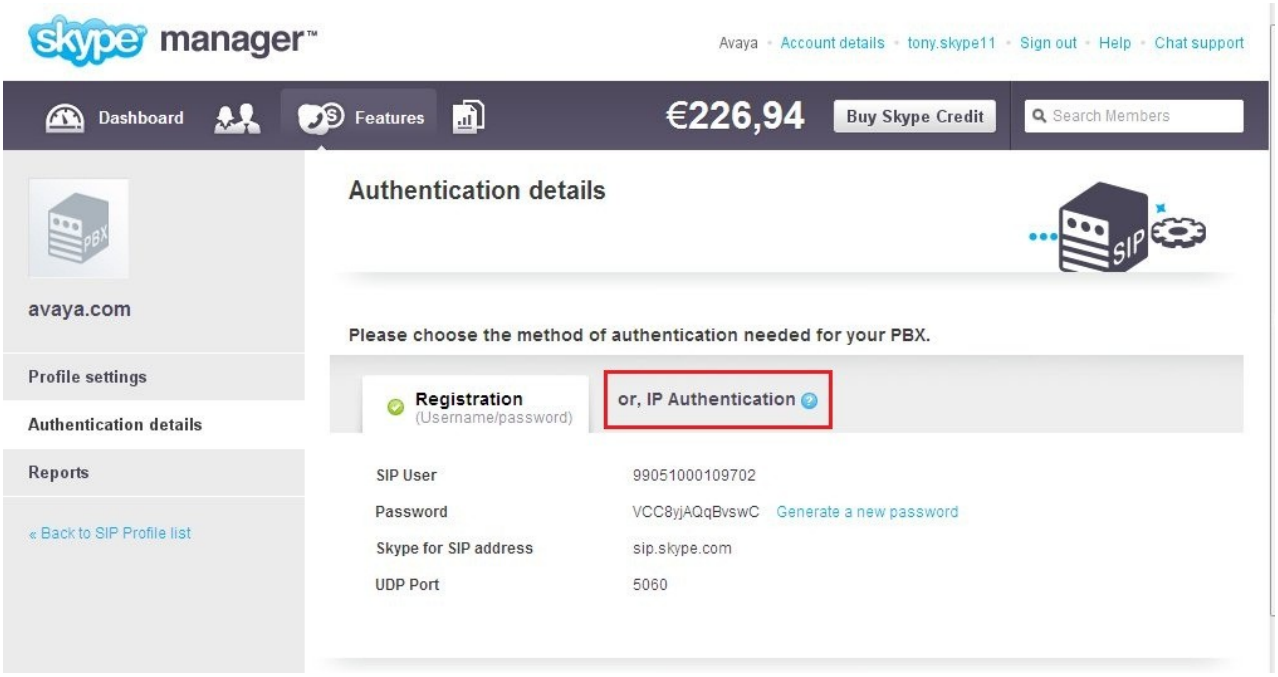
The **Choose a profile name** pop-up window is displayed. Type the name of your SIP Profile and click **Next**.



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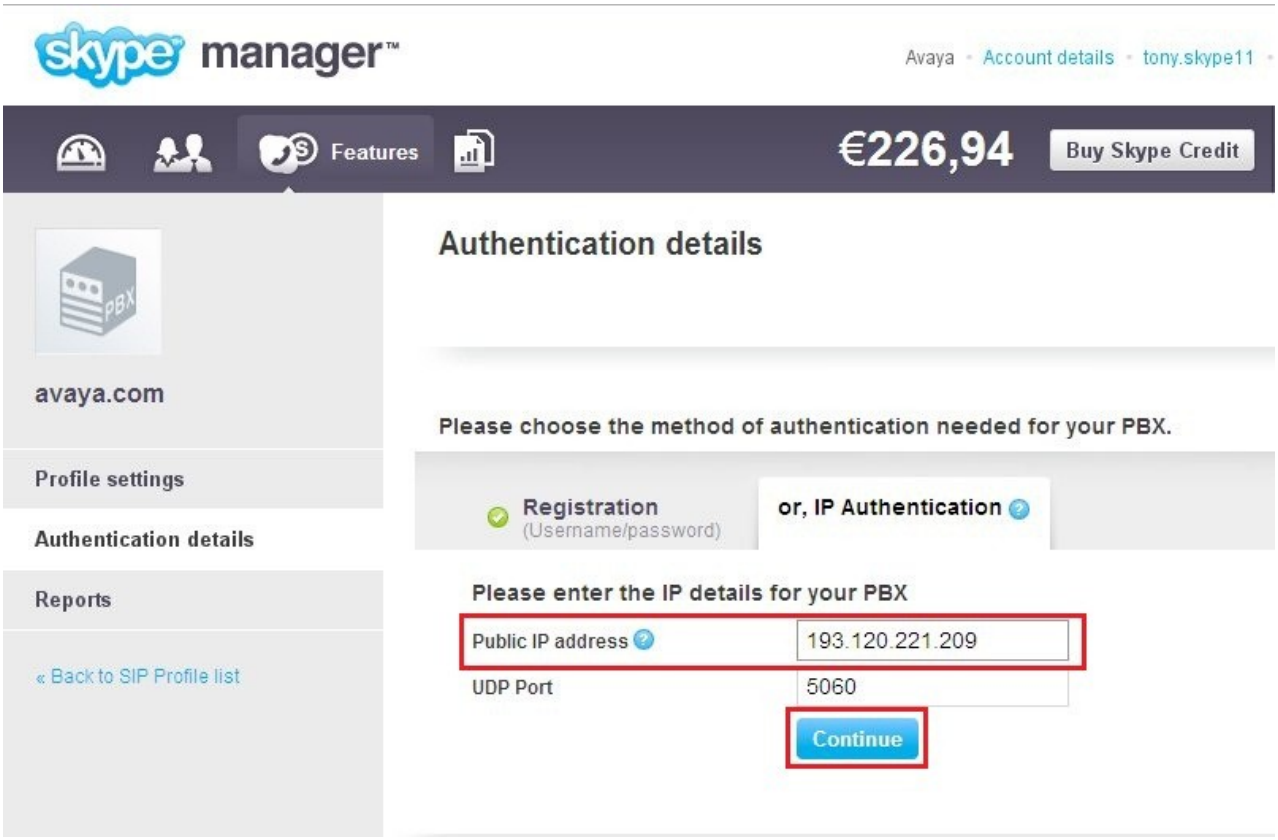
5.3. Administer Authentication Method

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



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

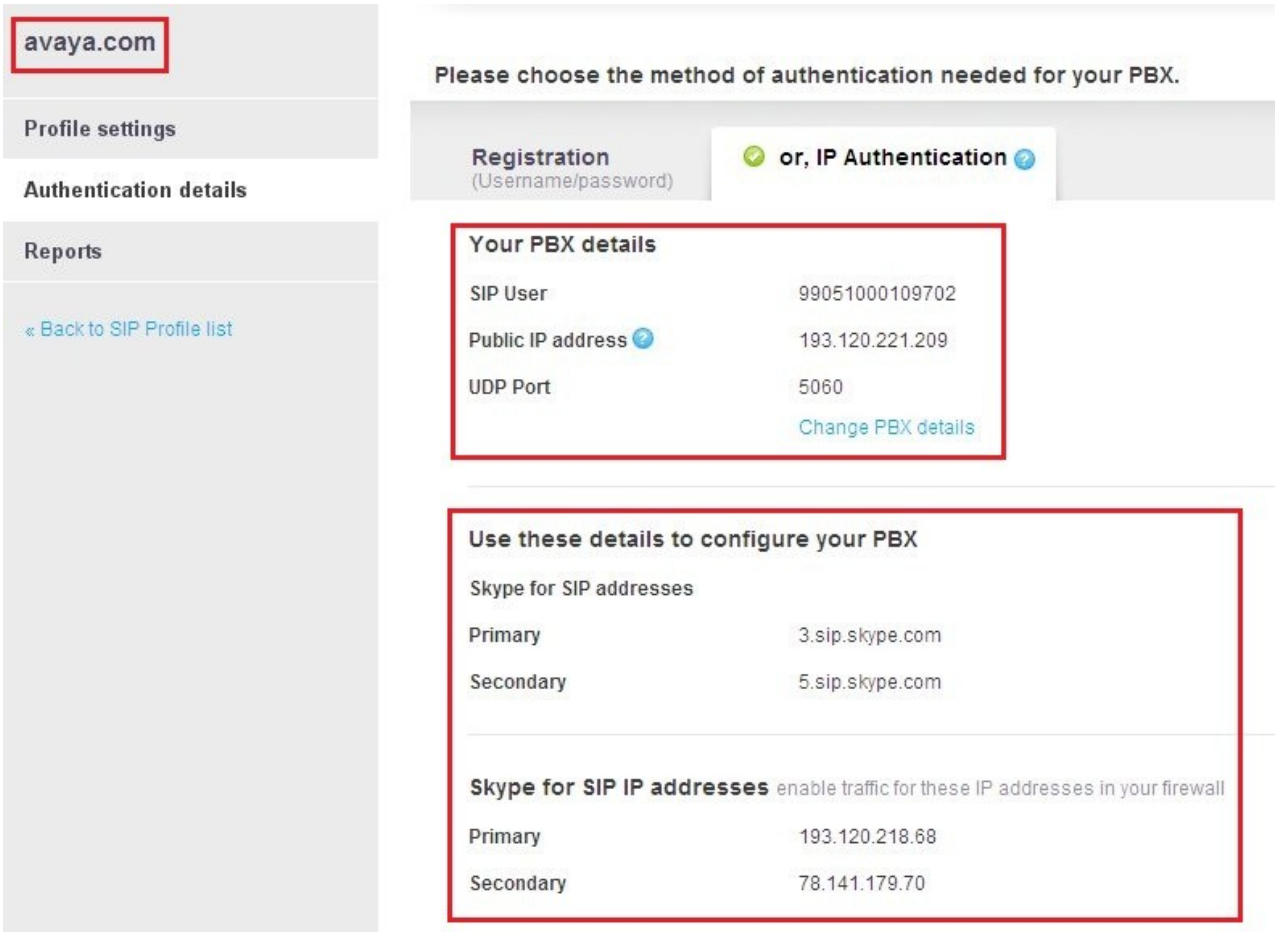


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Skype Manager automatically generates a unique SIP User and associates it with the newly created **avaya.com** SIP Profile. Calls originating from the Avaya Aura® Session Border Controller must include the SIP User and the sip.skype.com domain in the 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

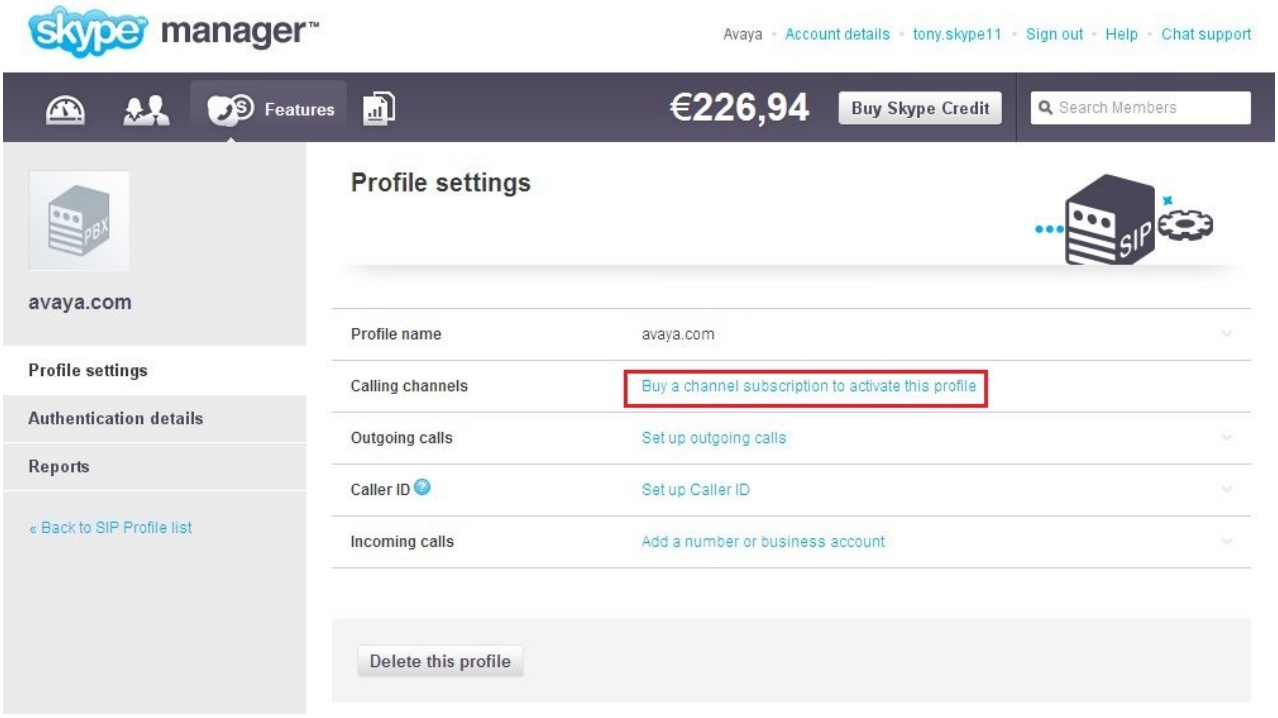
Skype Manager allocates a Primary and Secondary **Skype for SIP address**. Click **Profile settings**.



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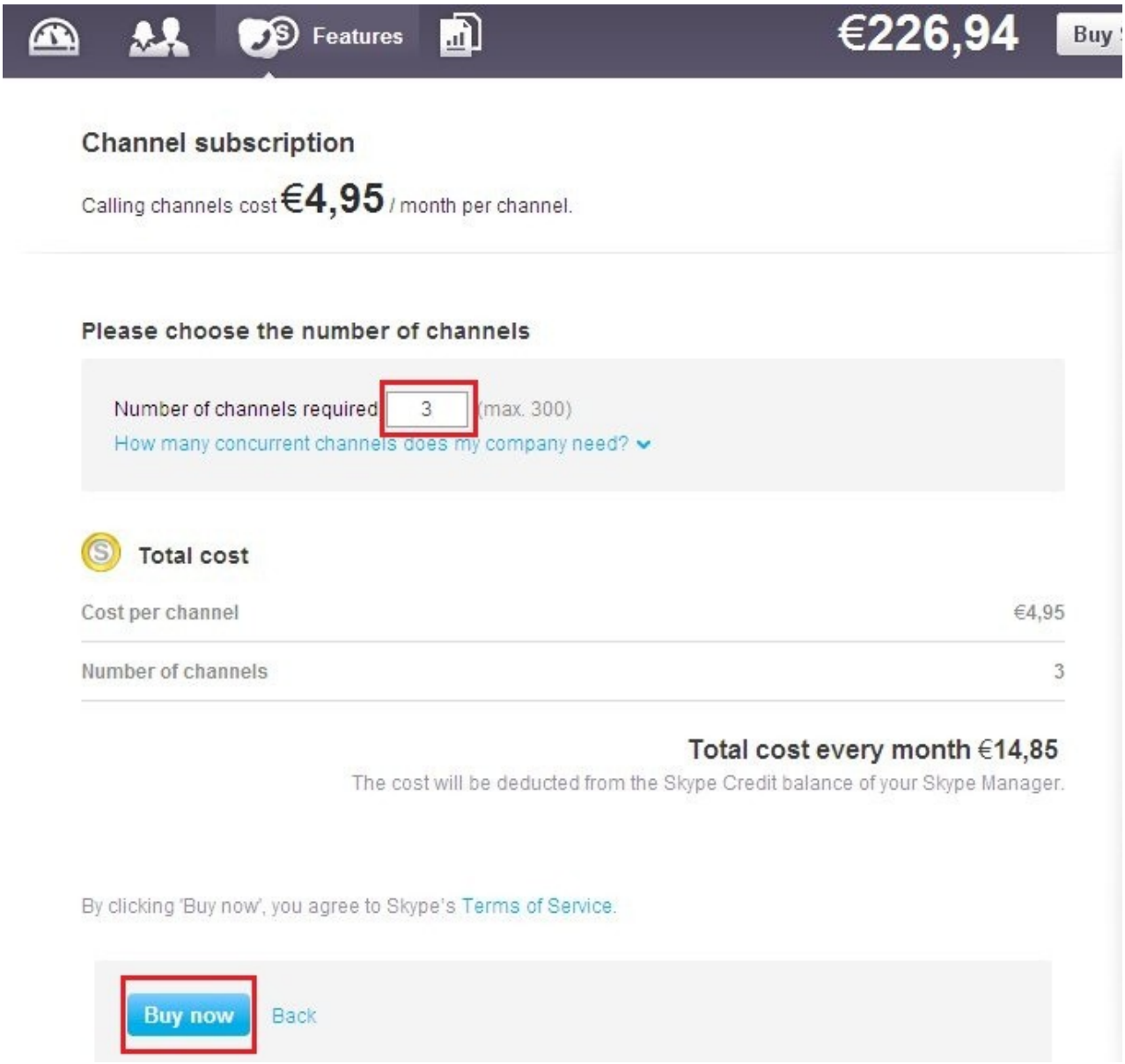
5.4. Administer Maximum Simultaneous Calls

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



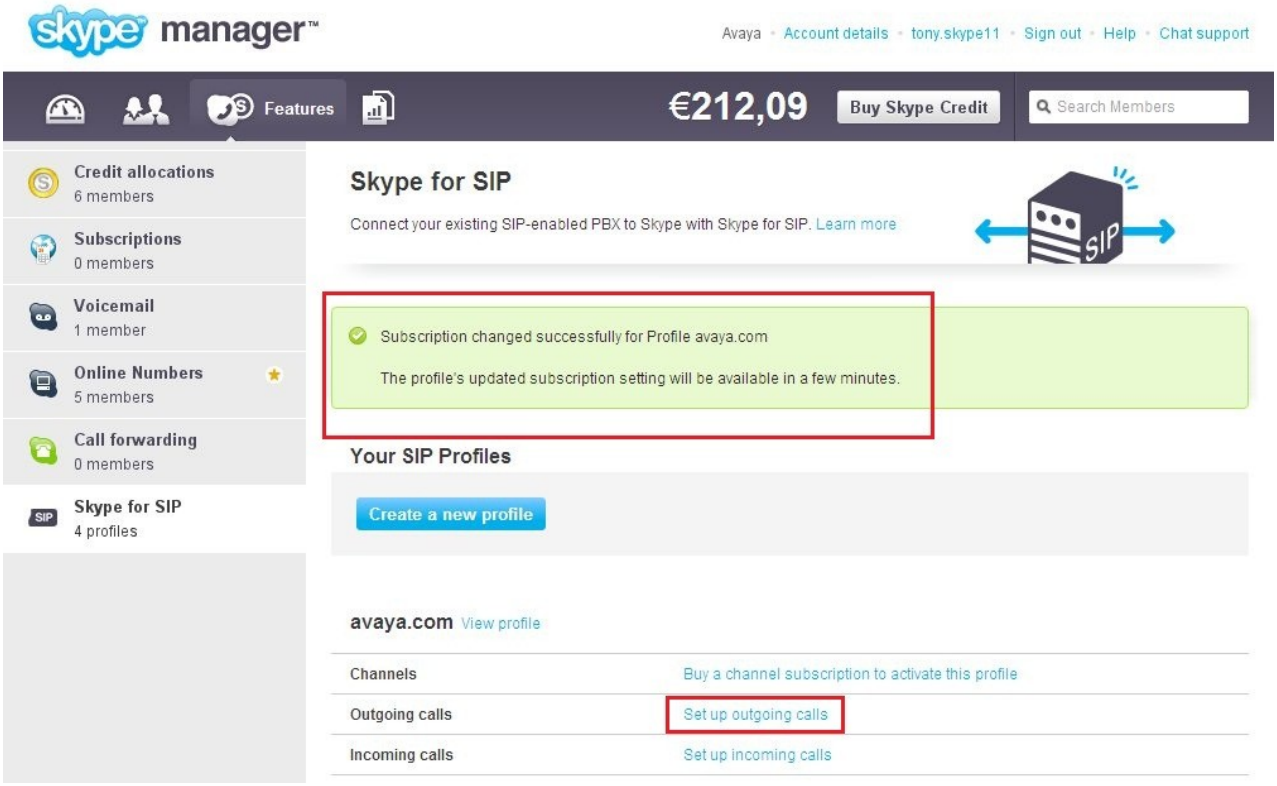
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The **Channel subscription** page is displayed. Type the maximum number of simultaneous calls that your business users are expected to make. Click **Buy now**.



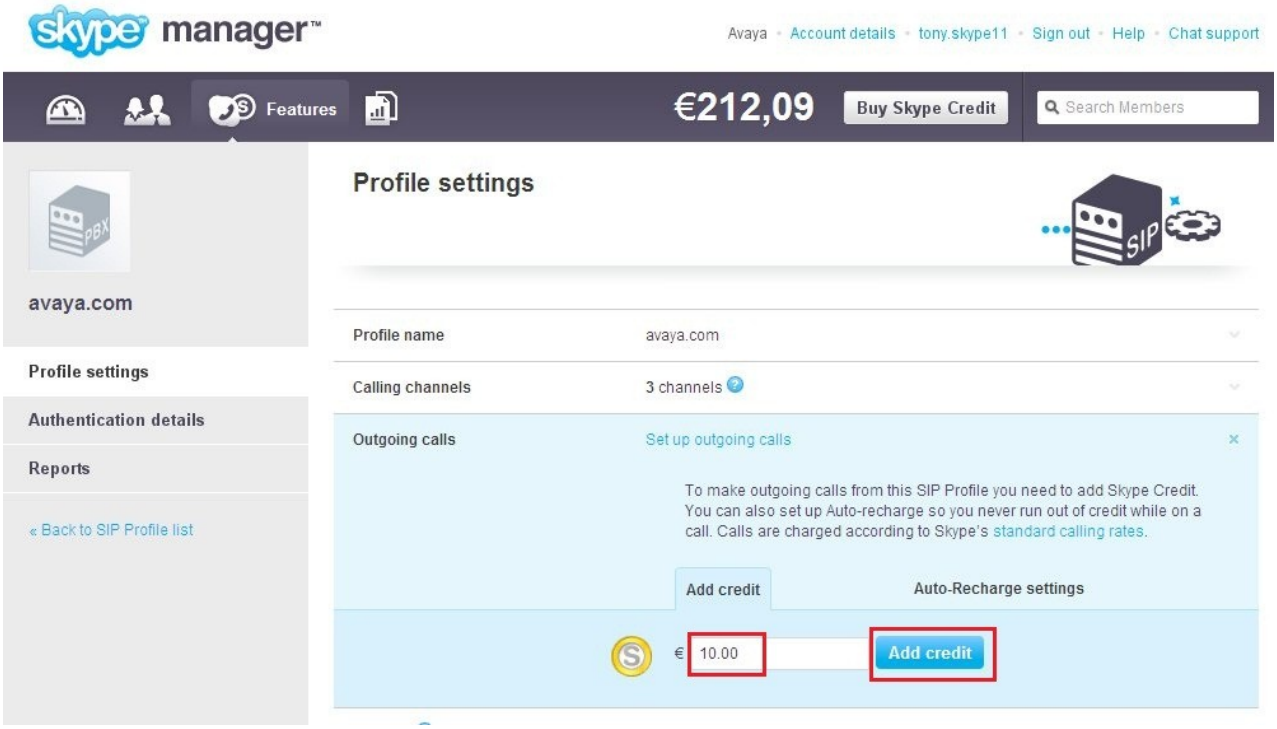
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The below status message is displayed to confirm that the channel subscription was successful. Click **Set up outgoing calls**.



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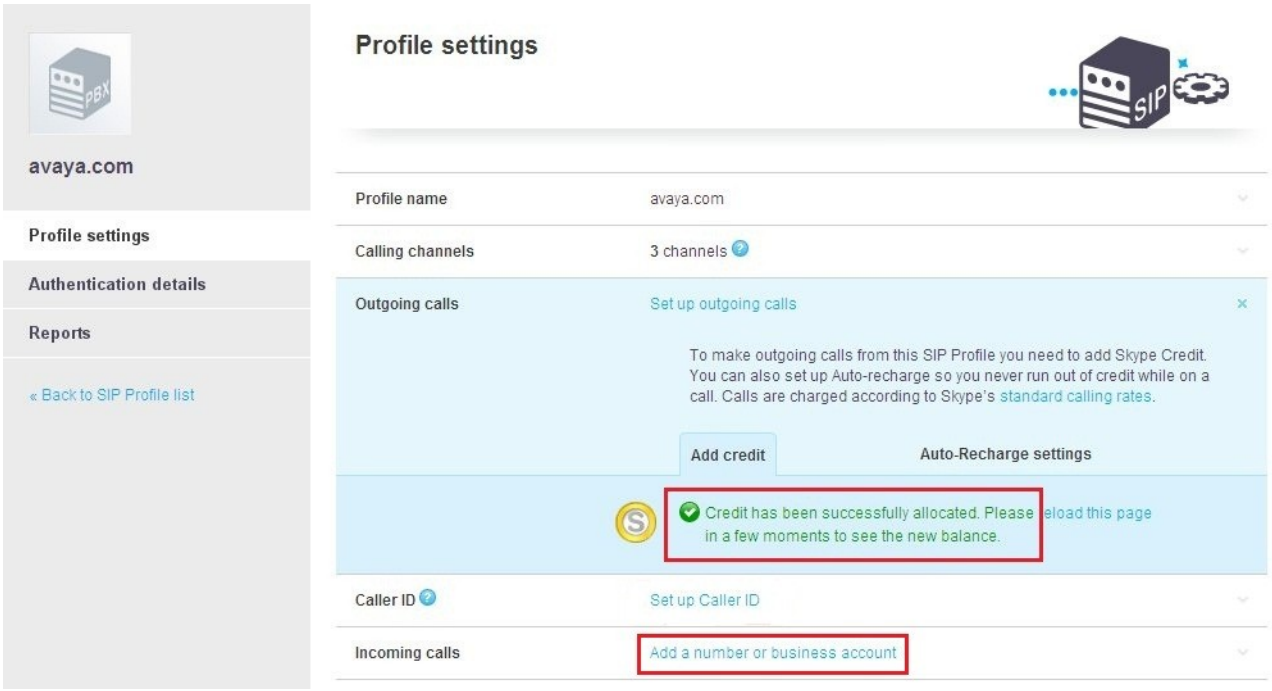
The **Profile settings** page is displayed. On the **Add credit** tab type the amount of credit you'd like to allocate for outbound calls. Note that calls to other SIP Profiles or Online Numbers of Skype Clients on the P2P network are free of charge. Click **Add credit**.



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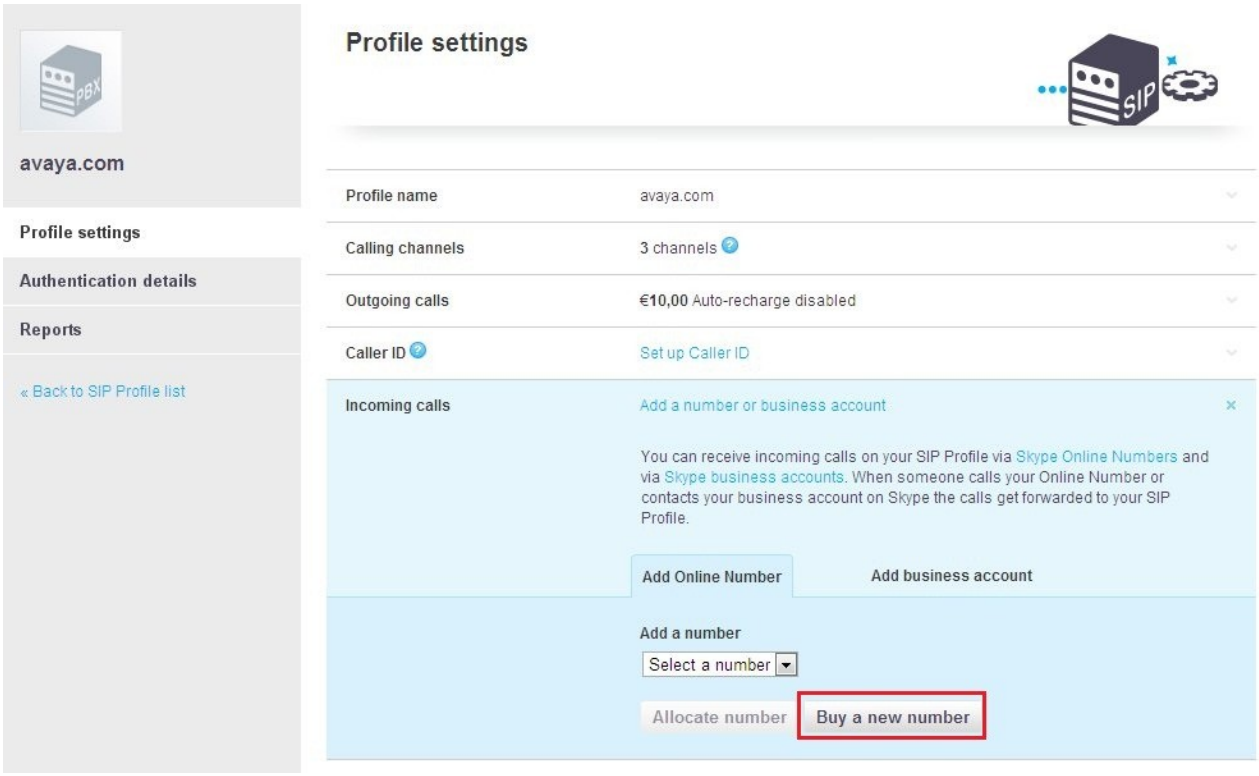
5.5. Administer Online Numbers

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**.



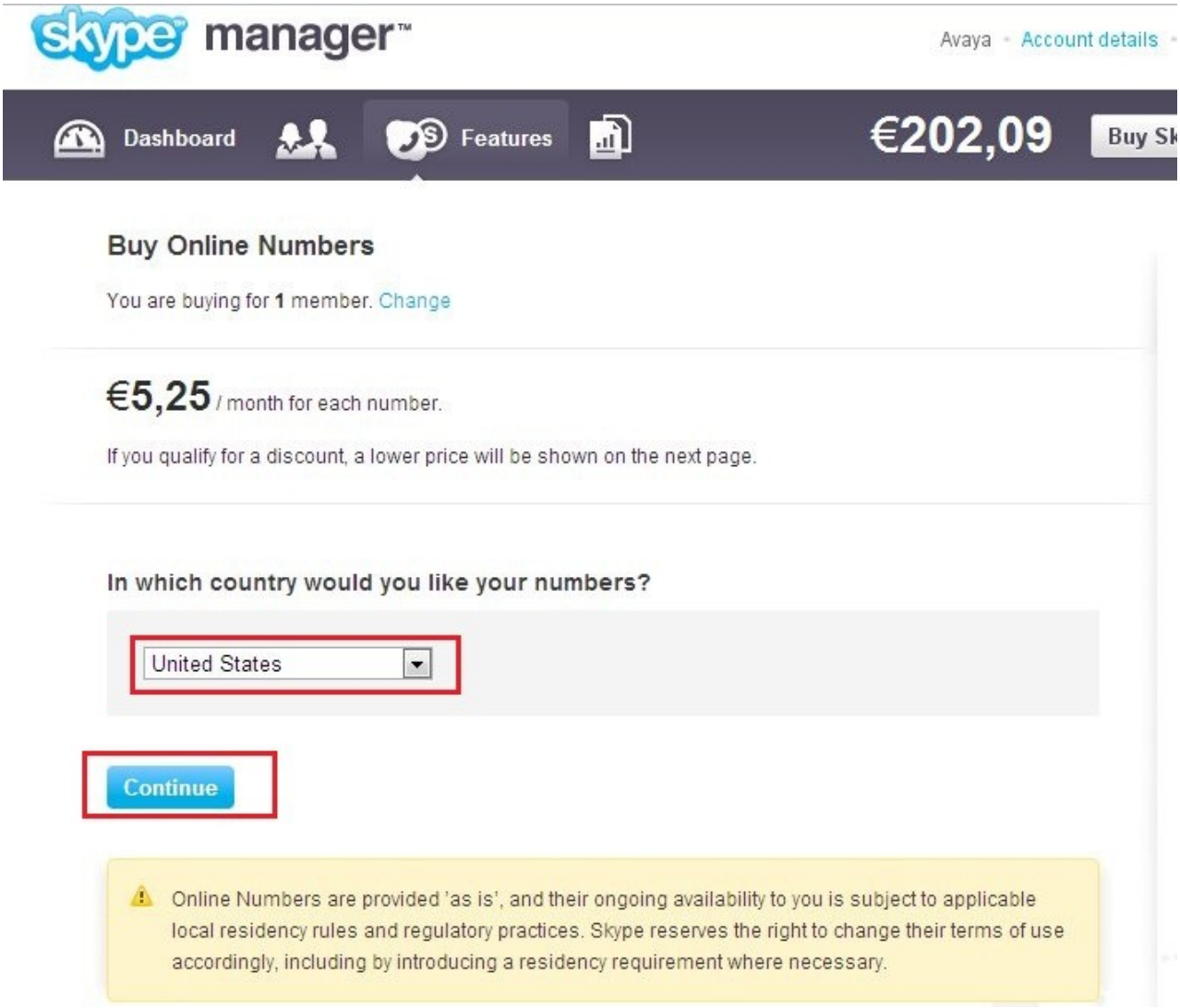
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The **Incoming calls** tab is expanded. Click **Buy a new number**.



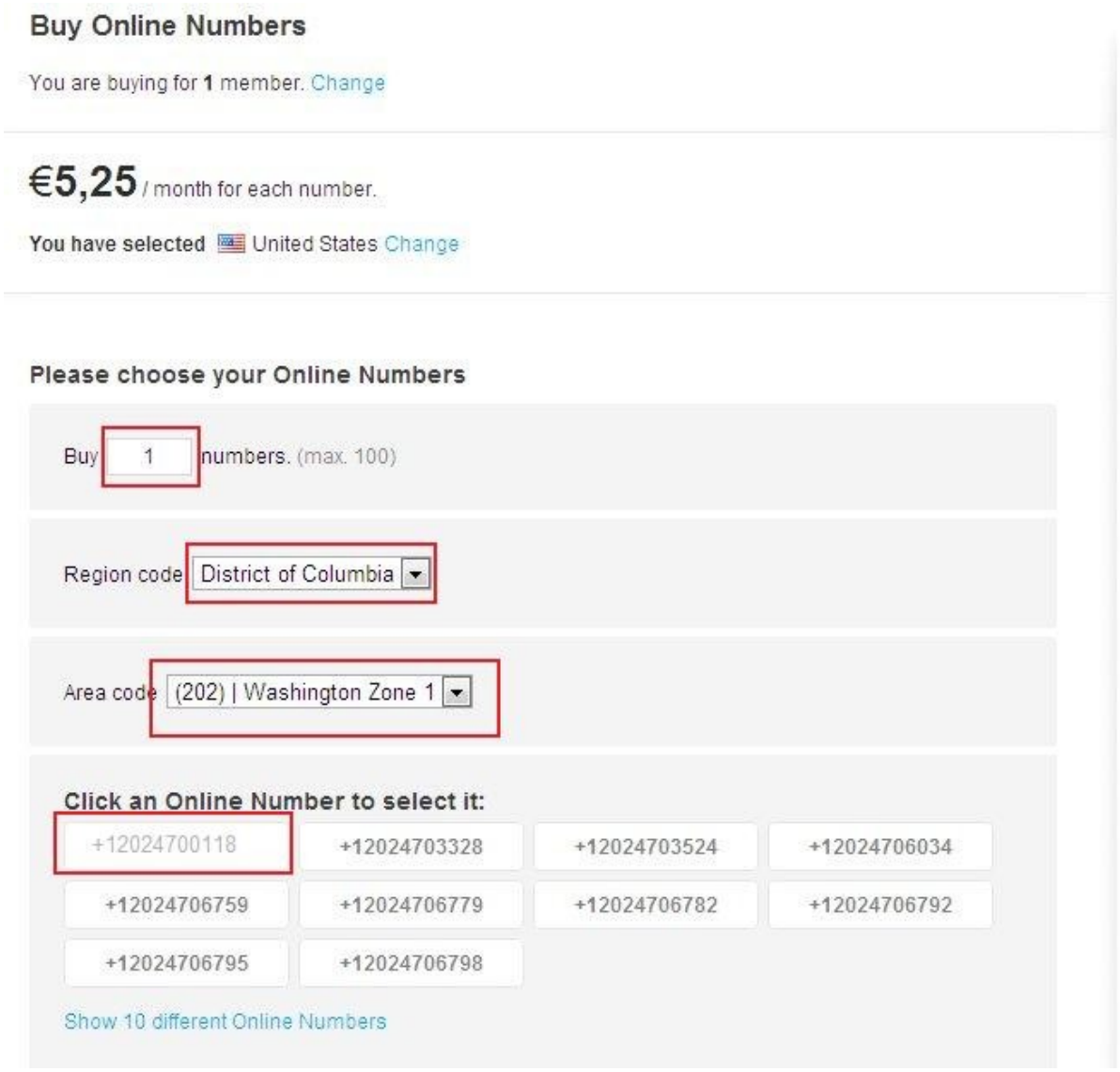
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The **Buy Online Numbers** page is displayed. Select **a country** from the drop-down list box and click **Continue**.



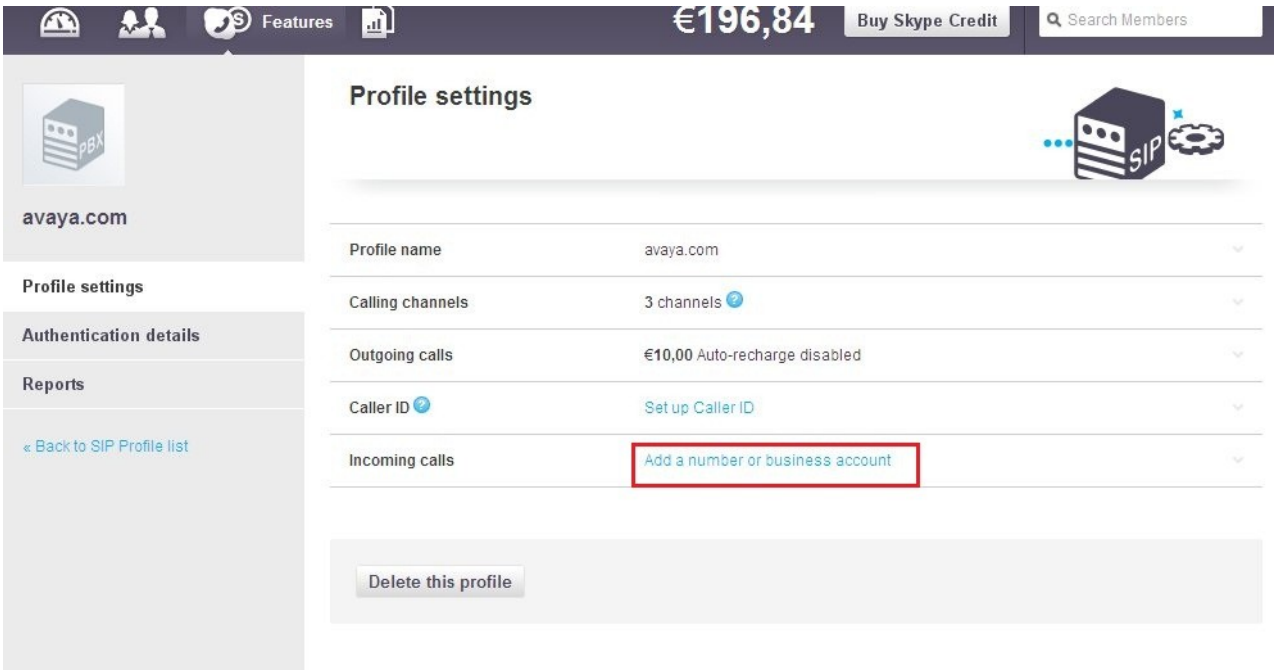
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The **Buy Online Numbers** page is displayed. Under **Please choose your Online Numbers** type the number of required Online Numbers which are used as DIDs in conjunction with the SIP Profile. Select **a state or county** from the **Region Code** drop-down 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).

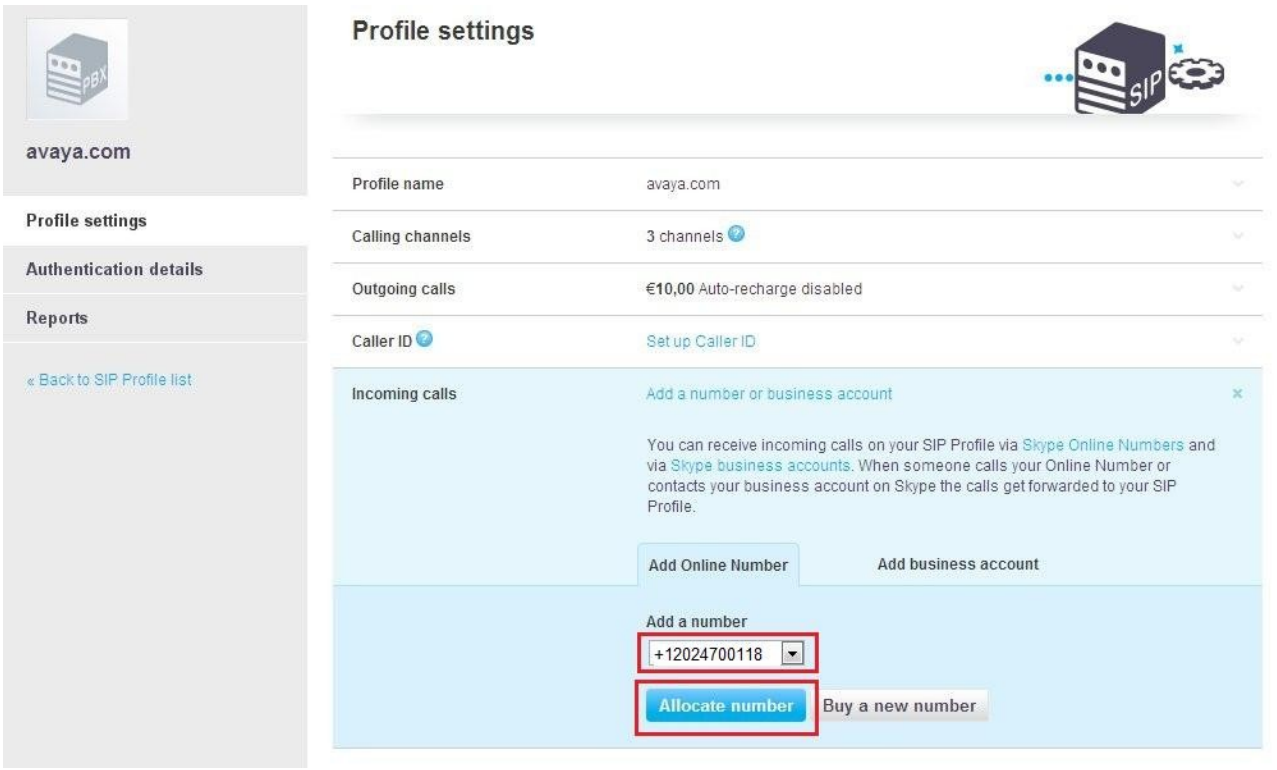


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The **Profile settings** page is displayed. Click **Add a number or business account**.



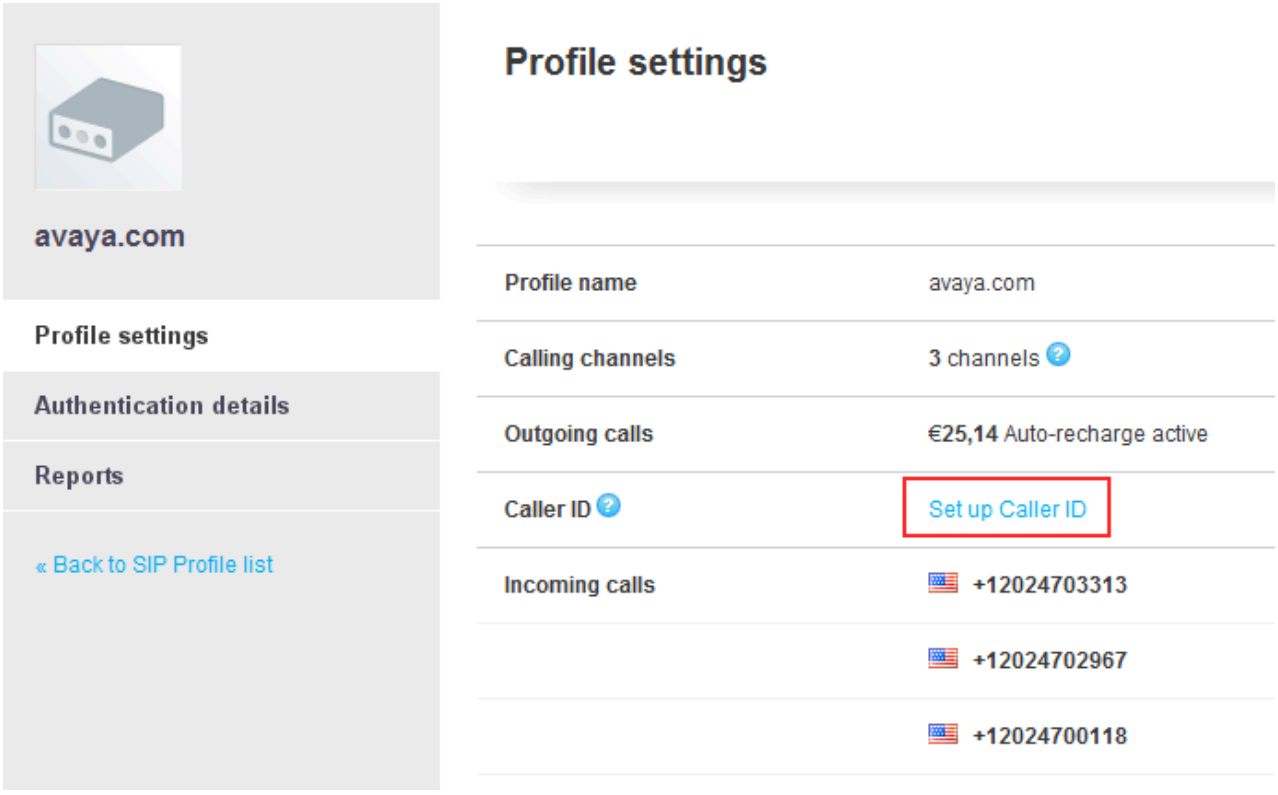
Select the newly purchased Online Number from the **Add a number** drop-down list box then click **Allocate number**.



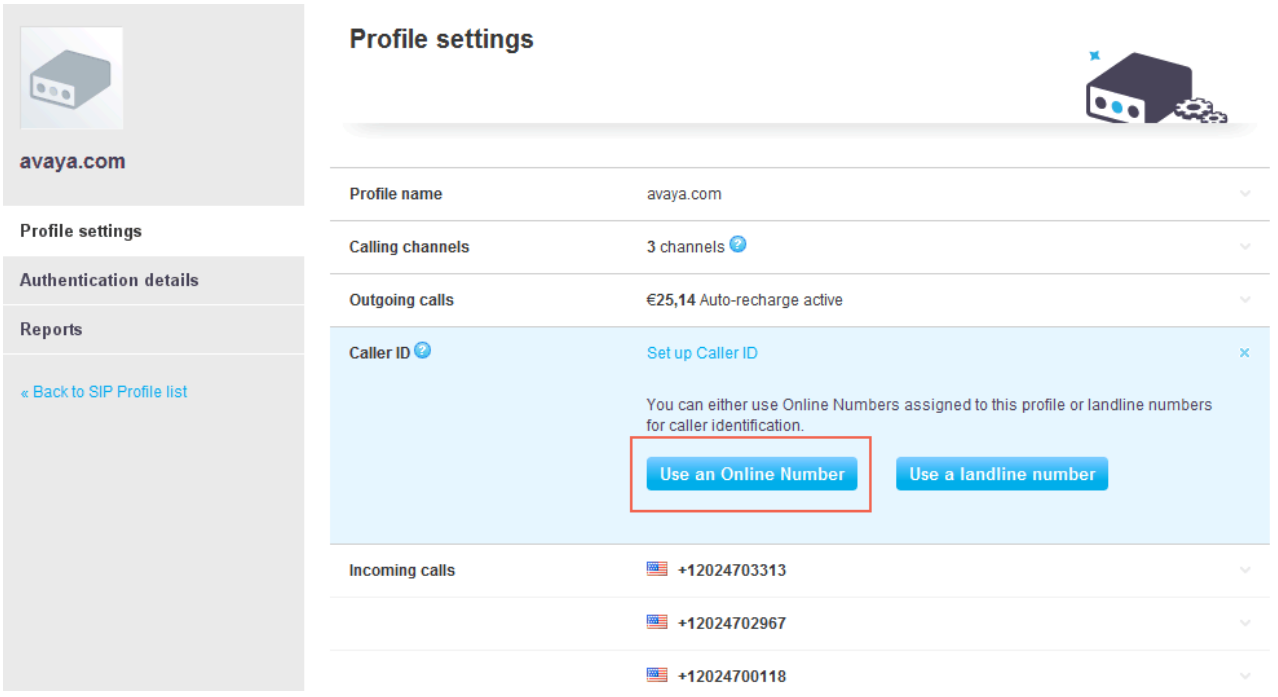
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5.6. Administer Caller ID

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

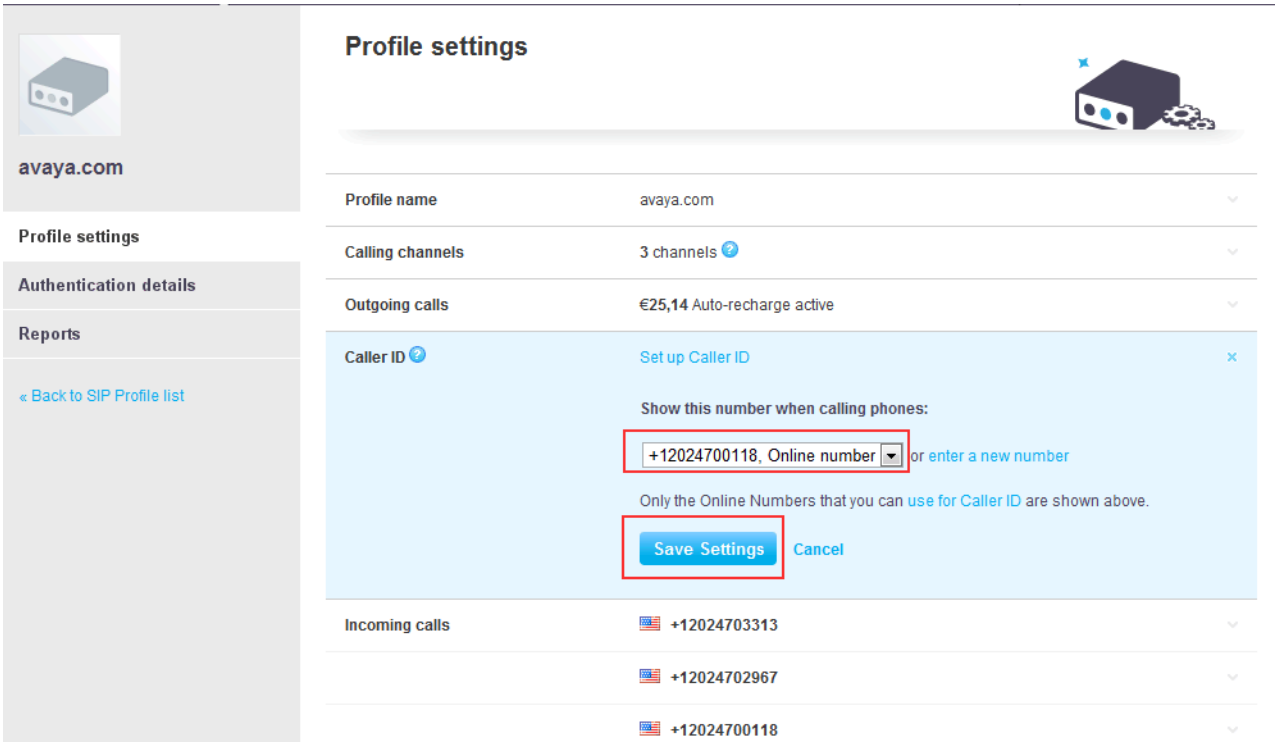


Click **Use an Online Number**.

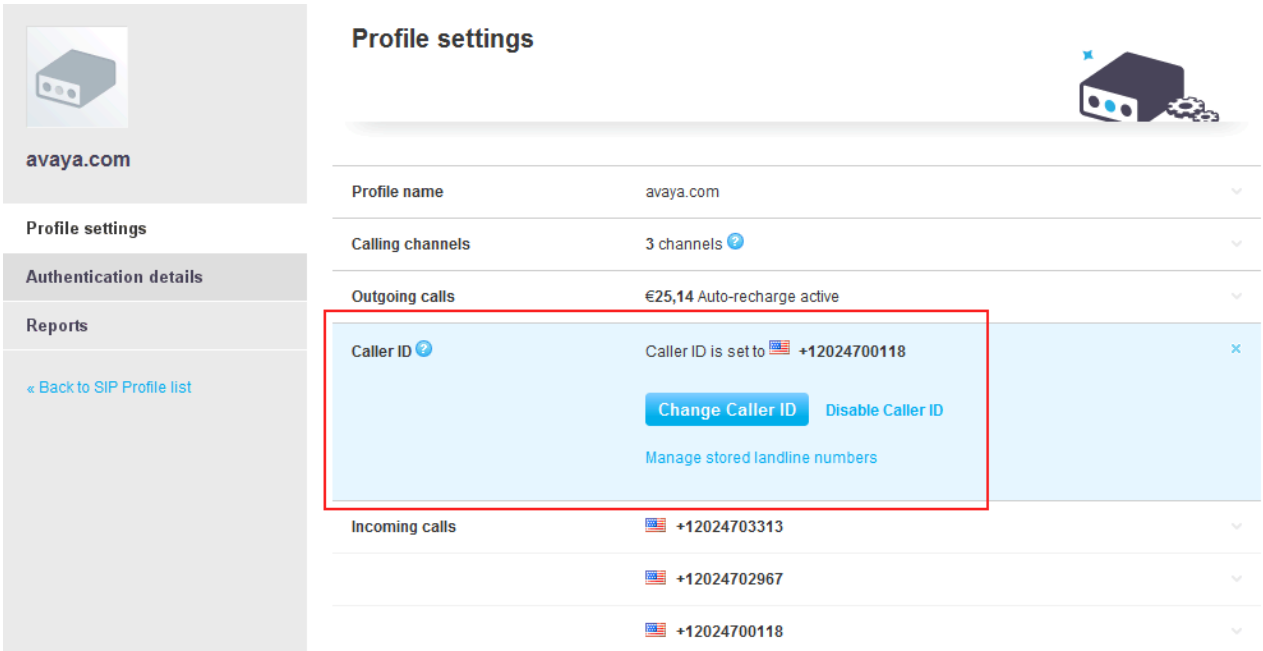


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Select an Online Number from the **Show this number when calling phones:** drop-down list box than click **Save Settings**.



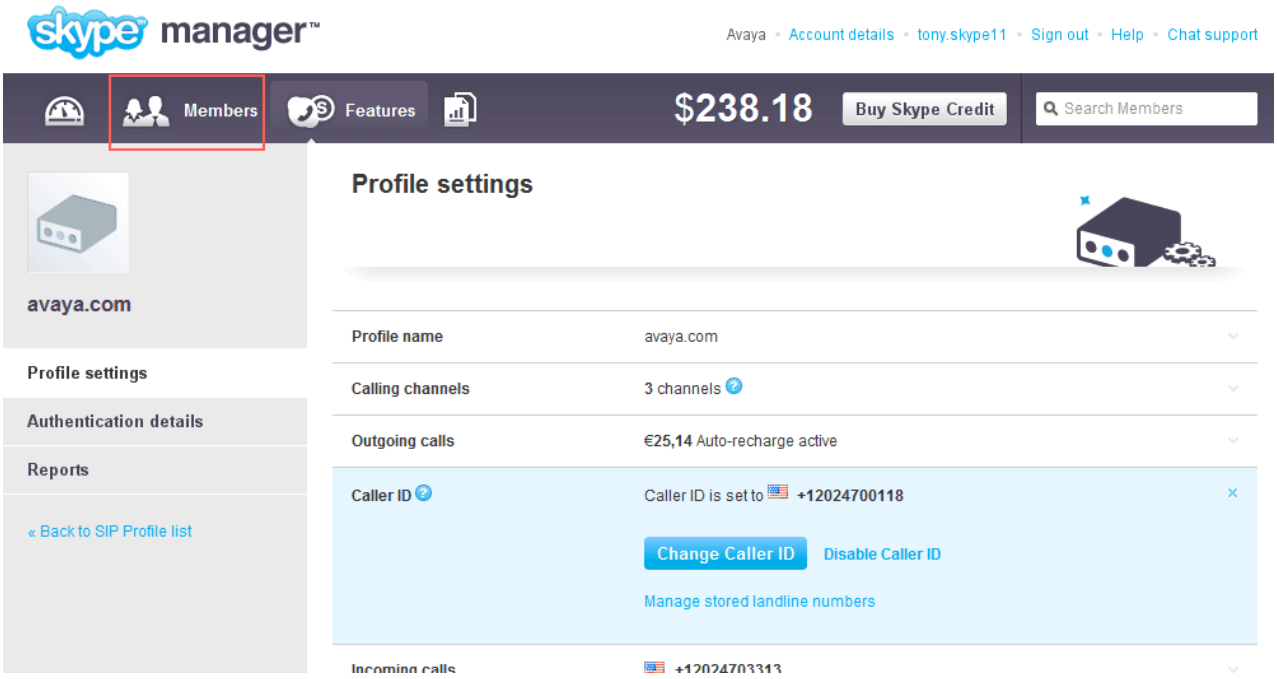
The following screenshot displays a sample Profile with both the **Caller ID** and the **Incoming calls** (DID) set to the same Online Number.



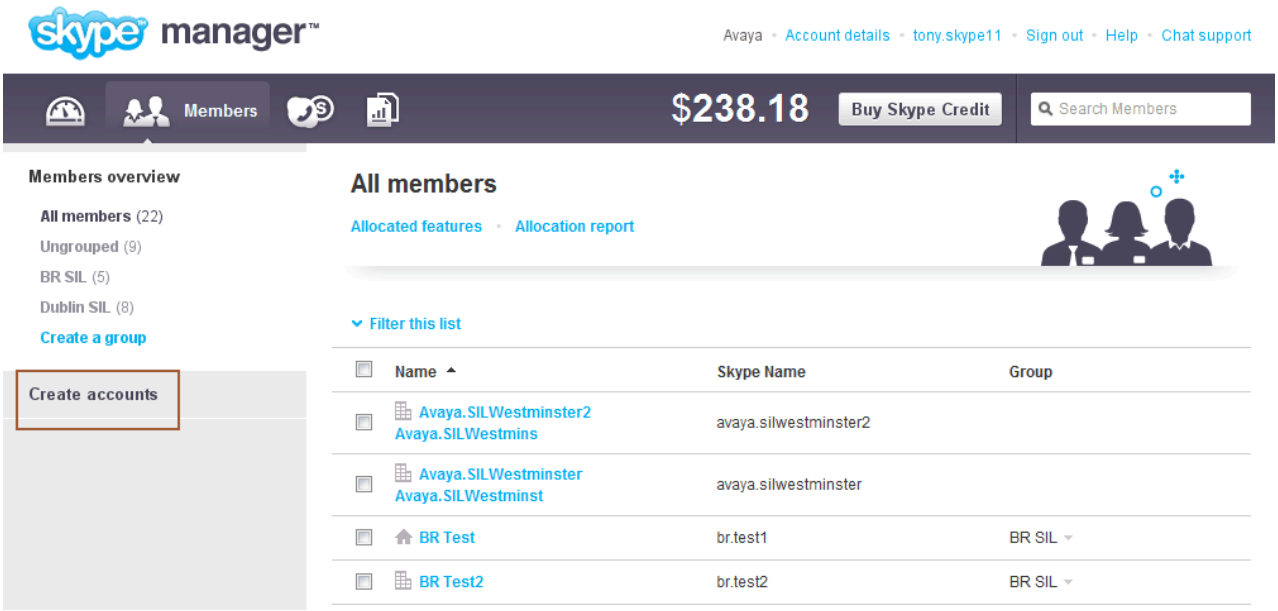
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5.7. Create a Business User

A Business User with a unique Skype Name needs to be created and associated with the internal Voice Portal self-service access number (3444) for click-to-call functionality to work from the Skype P2P Network. Click **Members** on the toolbar.

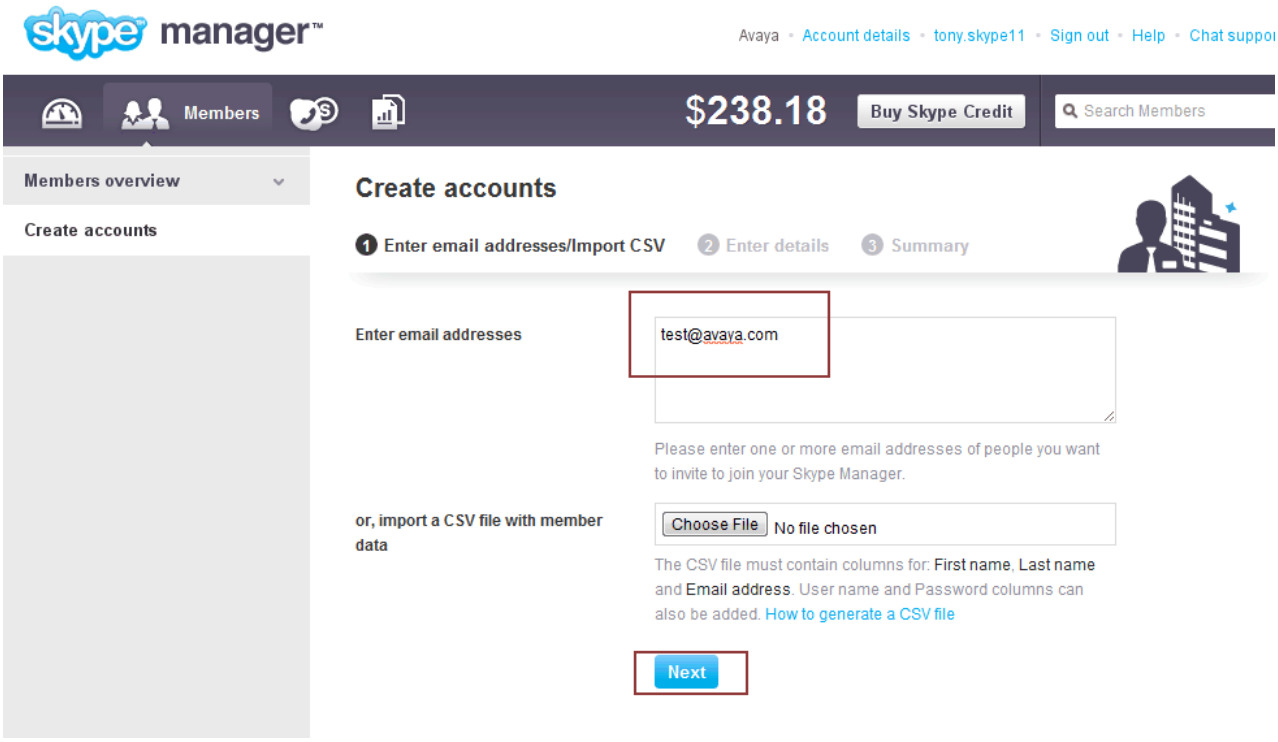


The **All Members** page is displayed. Click **Create accounts**.



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The **Create accounts** page is displayed. The **Create 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**.

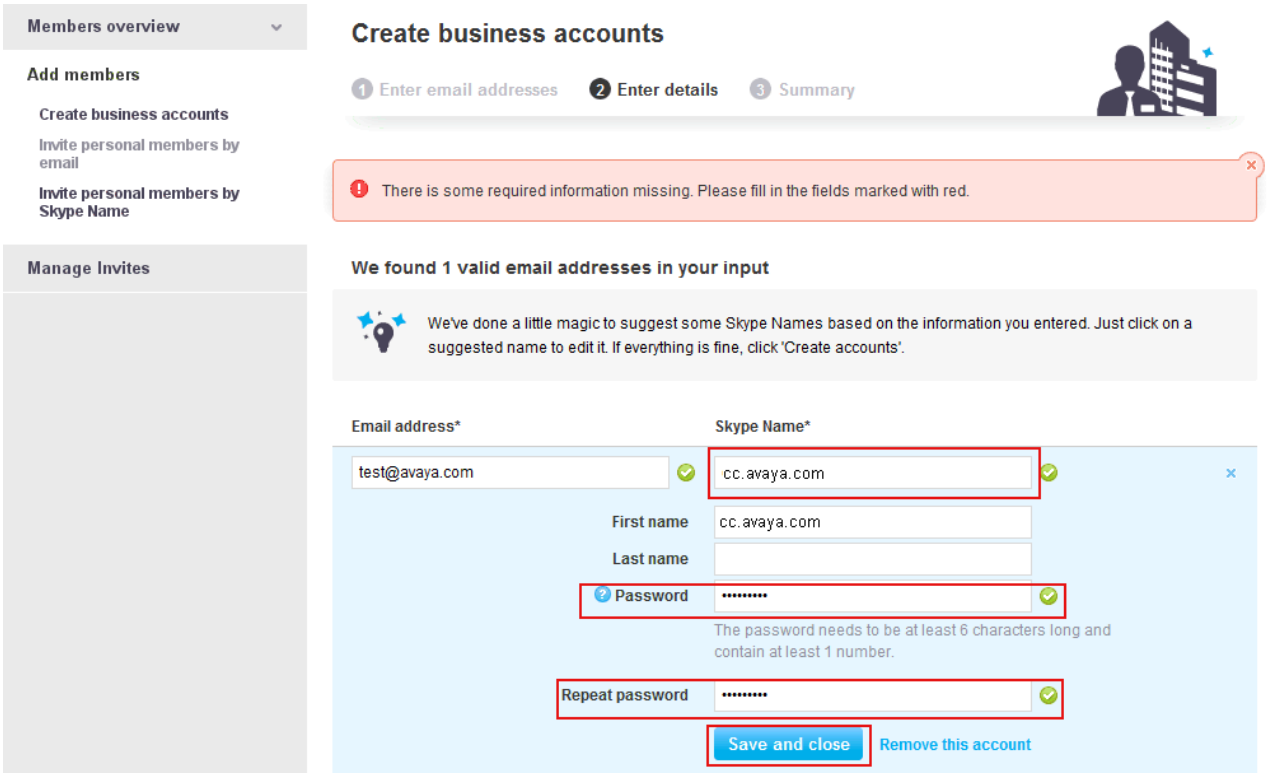


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The **Create business accounts** page is displayed. Under **Enter details** fill in the following fields:

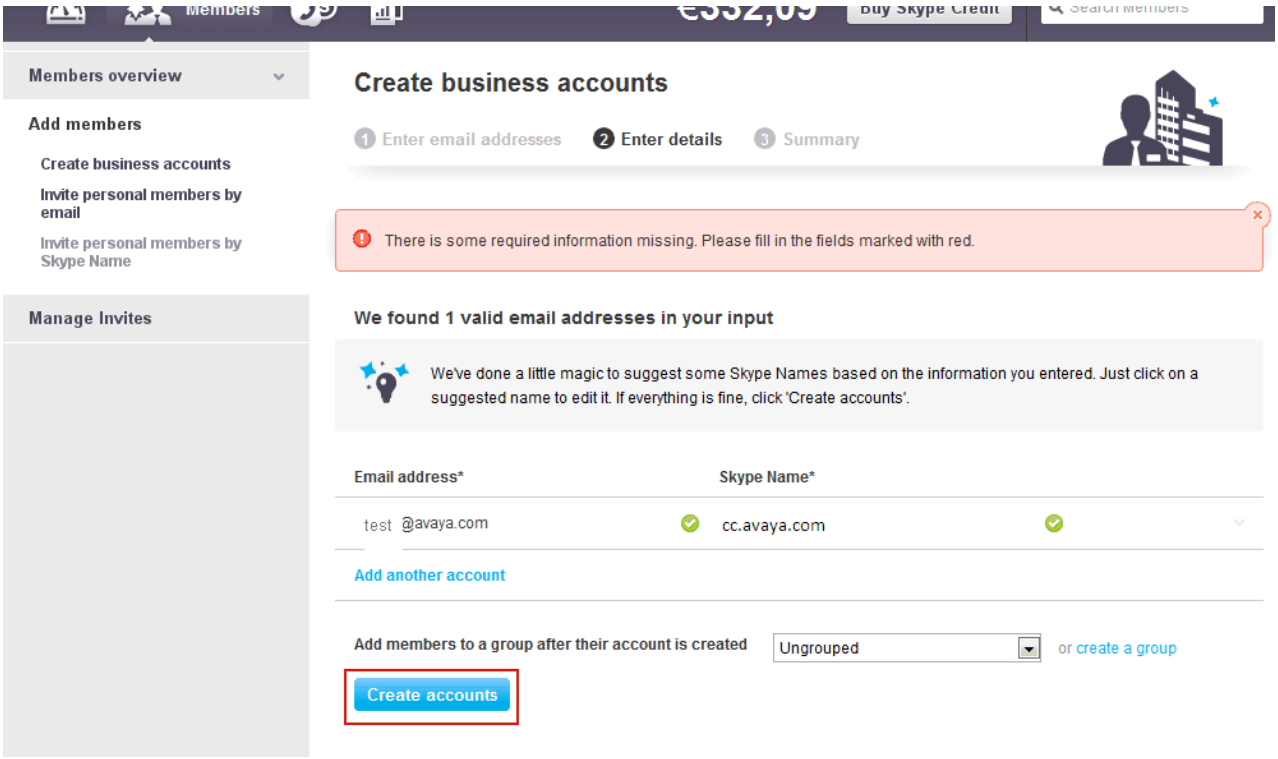
- **Skype Name:** Type a unique name. Skype Client users will click on this Skype Name to initiate a call to Voice Portal.
- **Password:** Type a password.
- **Repeat Password:** Repeat the password.

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



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

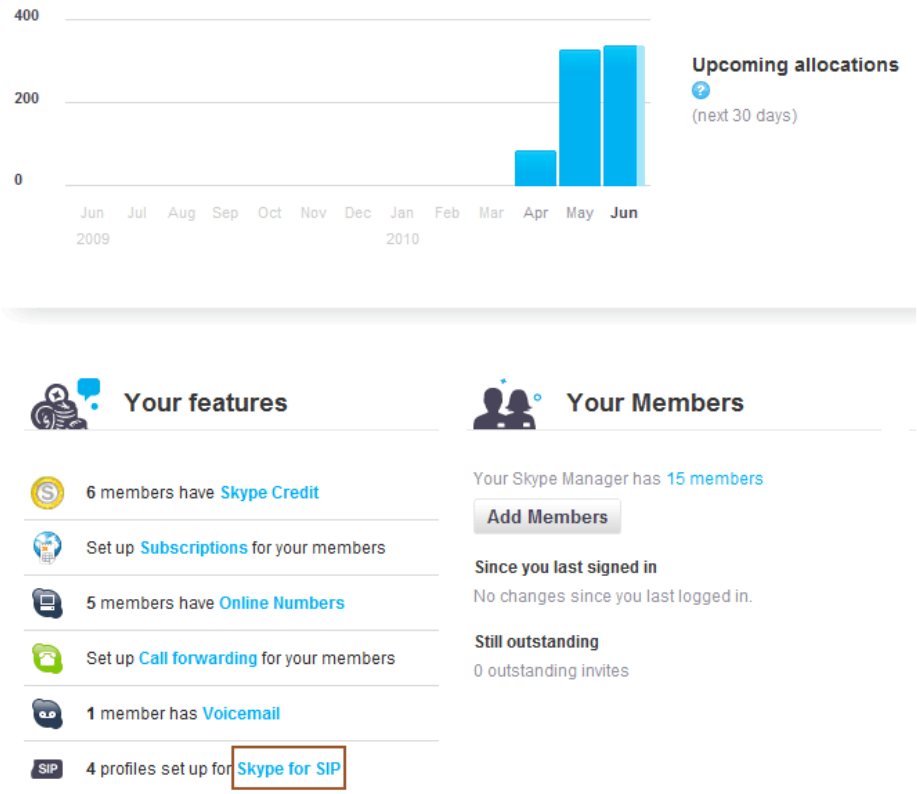


The **Create accounts - Summary** page is displayed (not shown). Click **Dashboard**.

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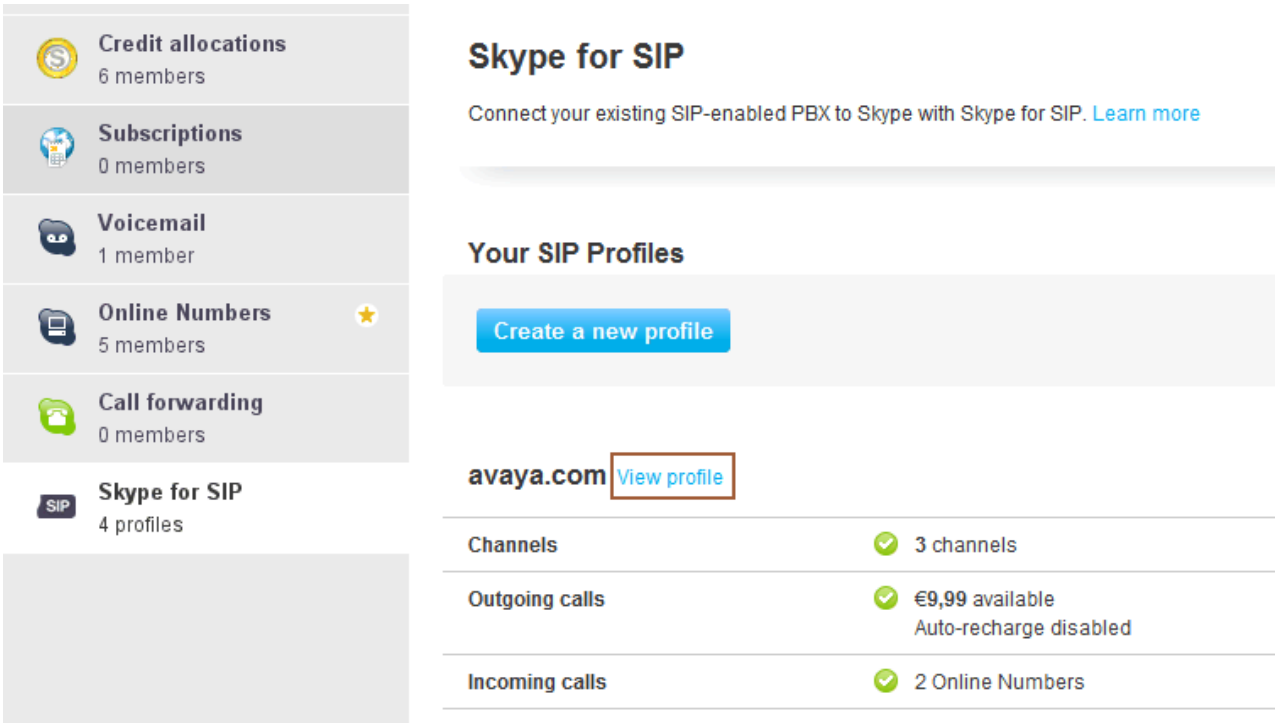
5.8. Add Business User to SIP Profile

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

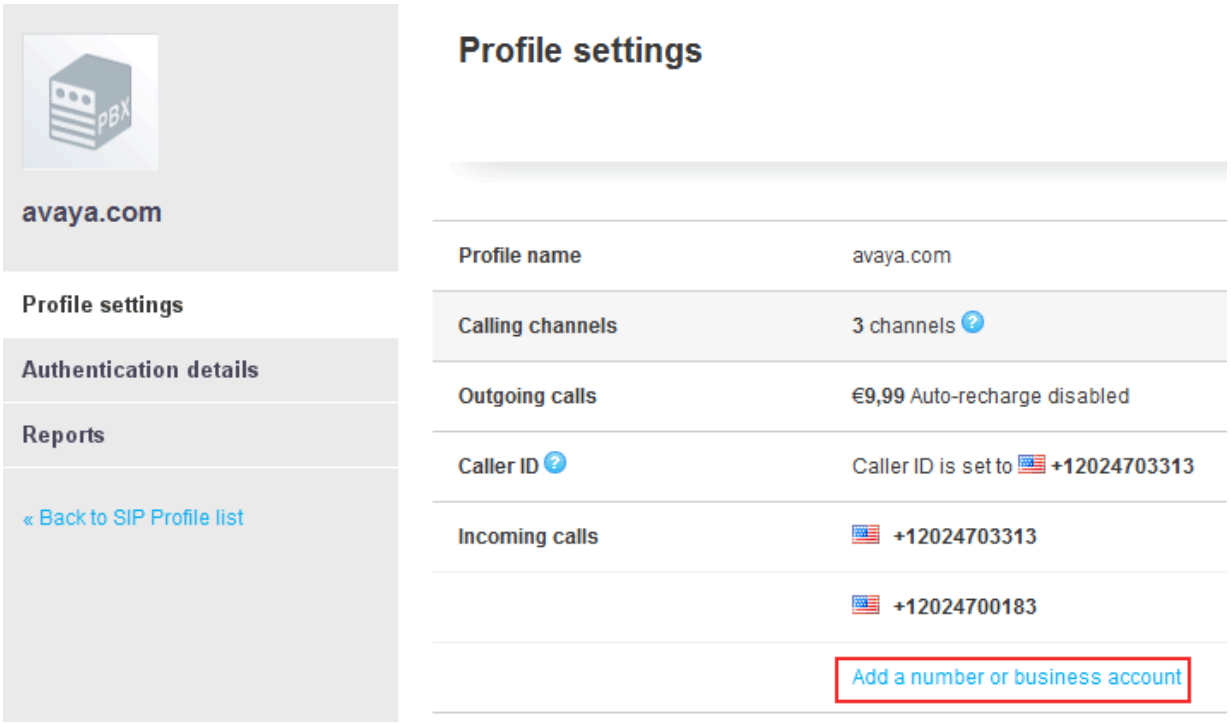


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The **Skype for SIP** page is displayed. Click **View profile**.

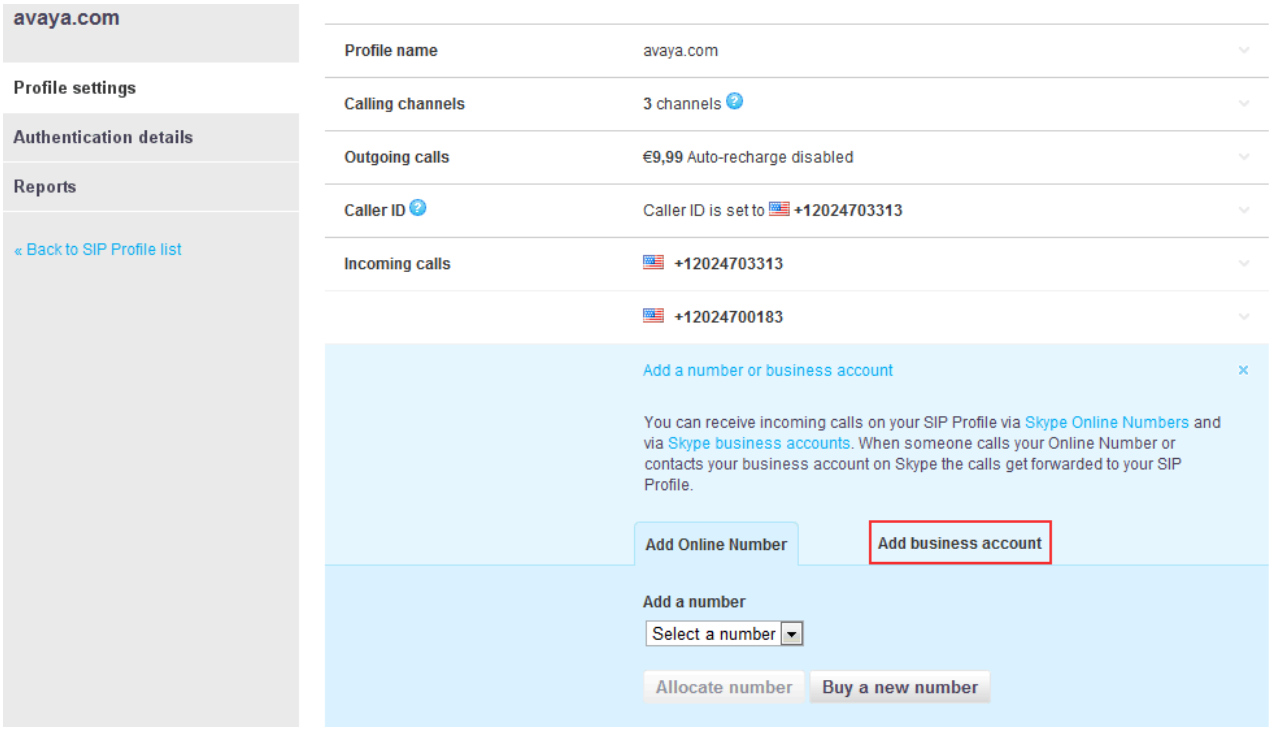


The **Profile settings** page is displayed. Click **Add a number or business account**.



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

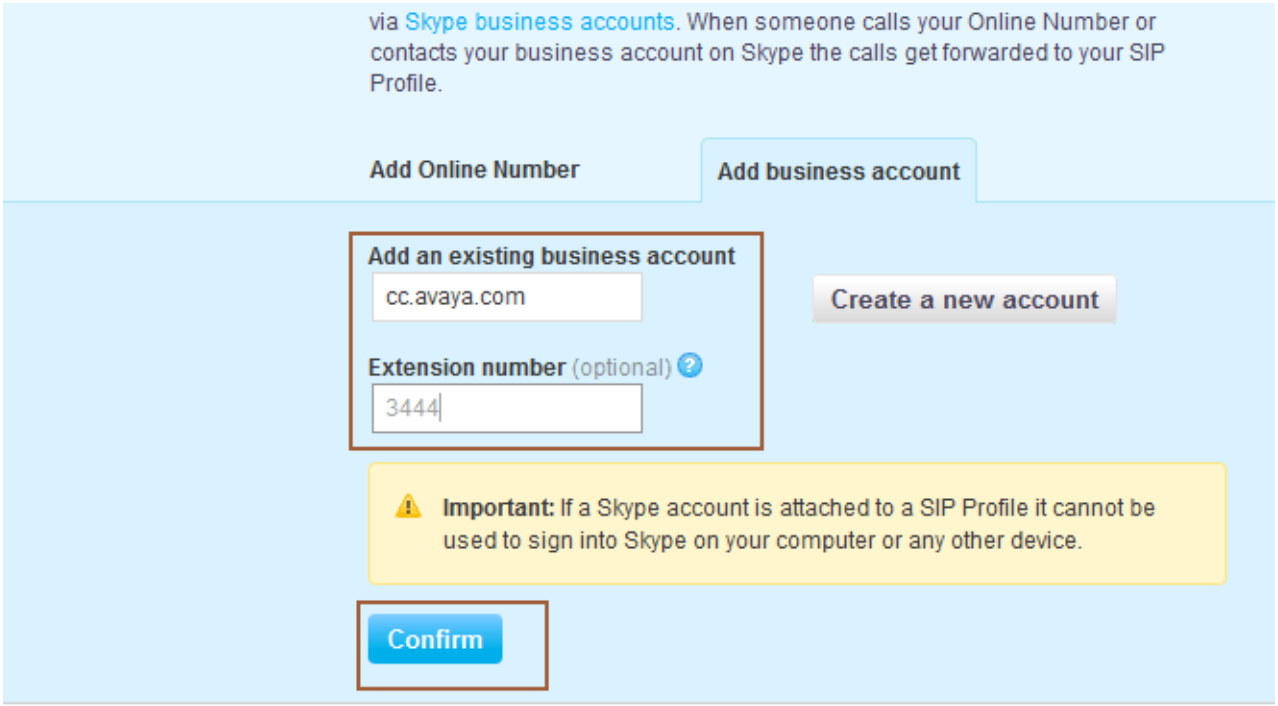


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Under the **Add business account** tab fill in the following fields:

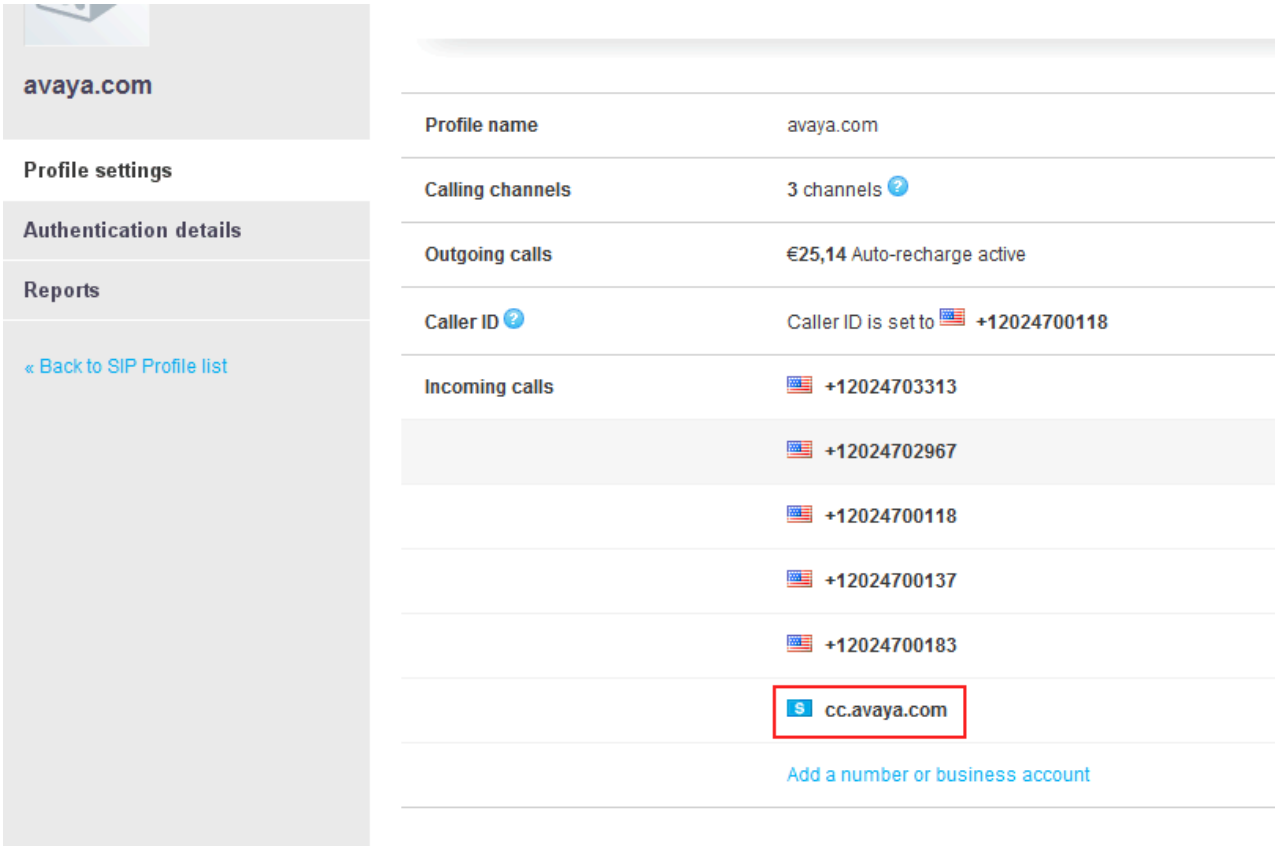
- **Add an existing business account:** Type a unique name. Skype Client users will click on the Skype Name to initiate a call to Voice Portal.
- **Extension number:** Type an extension.

Click **Confirm.**



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



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6. Configure Avaya Aura® Session Border Controller

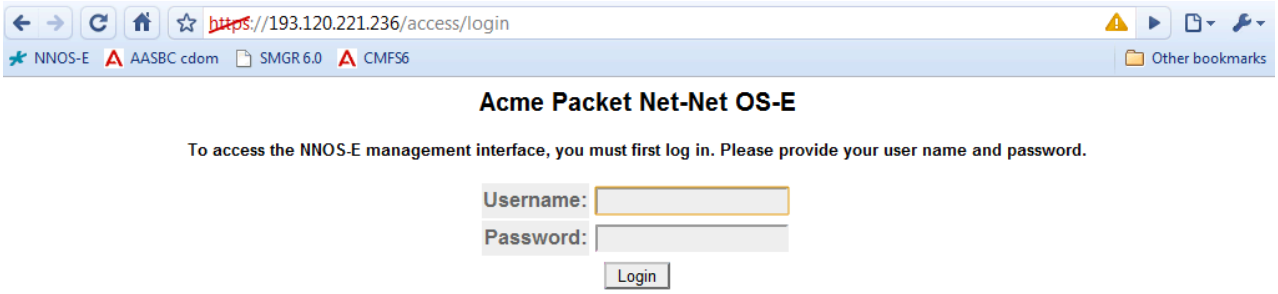
This section provides the procedures for configuring Session Border Controller and includes the following items:

- 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
- Administer Dial Plan
- Administer Third Party Call Control Settings
- Save the Configuration

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

6.1. Log in to Avaya Aura® Session Border Controller using the GUI

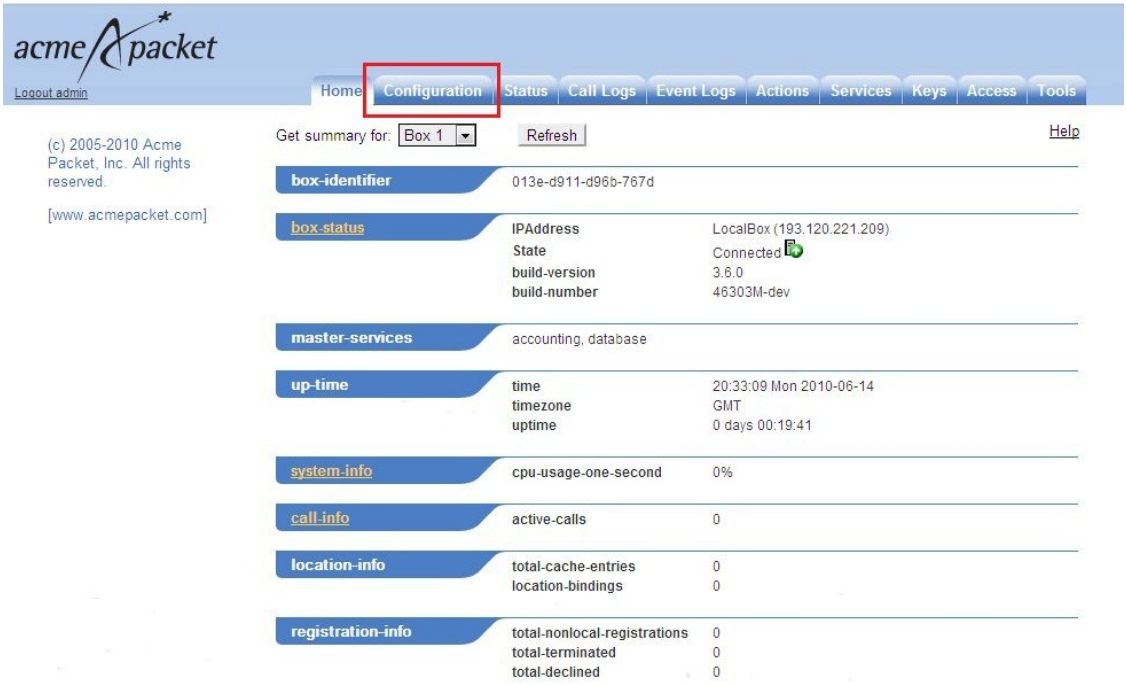
Configuration is accomplished by accessing the browser-based GUI of Session Border Controller, using the URL "**https://<ip-address>**", where "<ip-address>" is the IP address of the inside interface of the Session Border Controller. Log in with the appropriate credentials.



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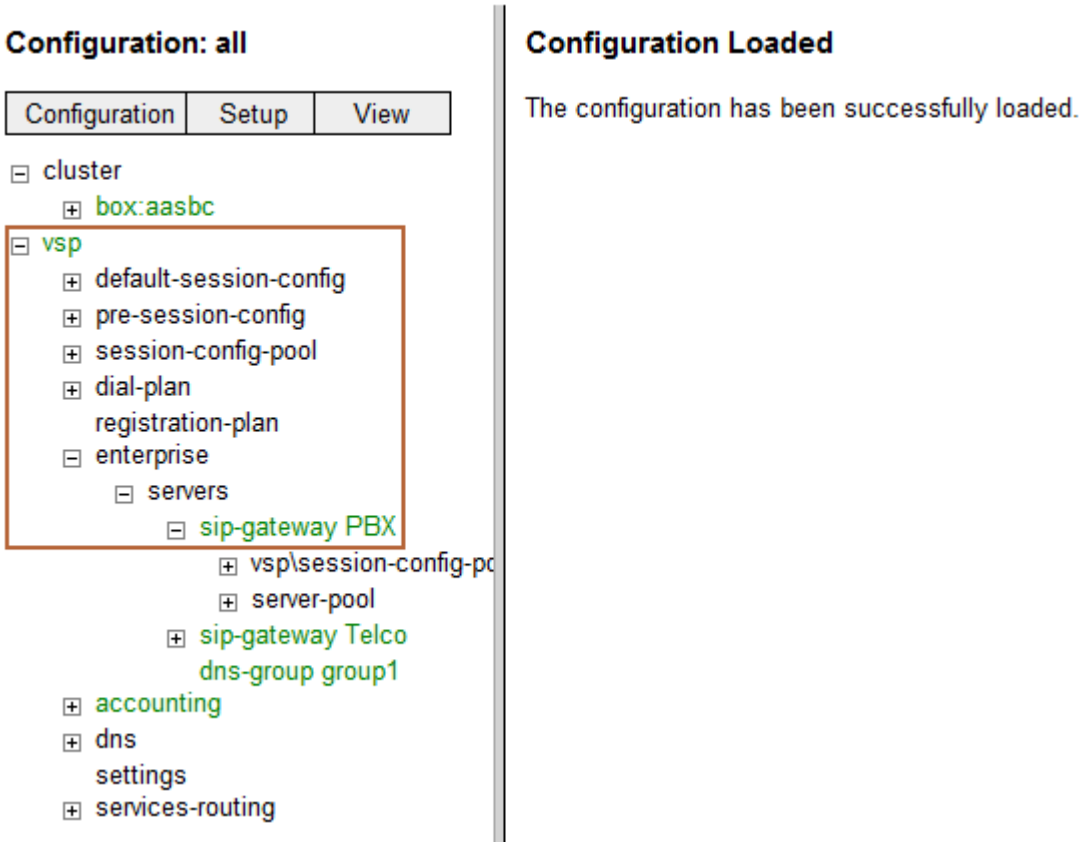
6.2. Administer SIP Domains

The 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.



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



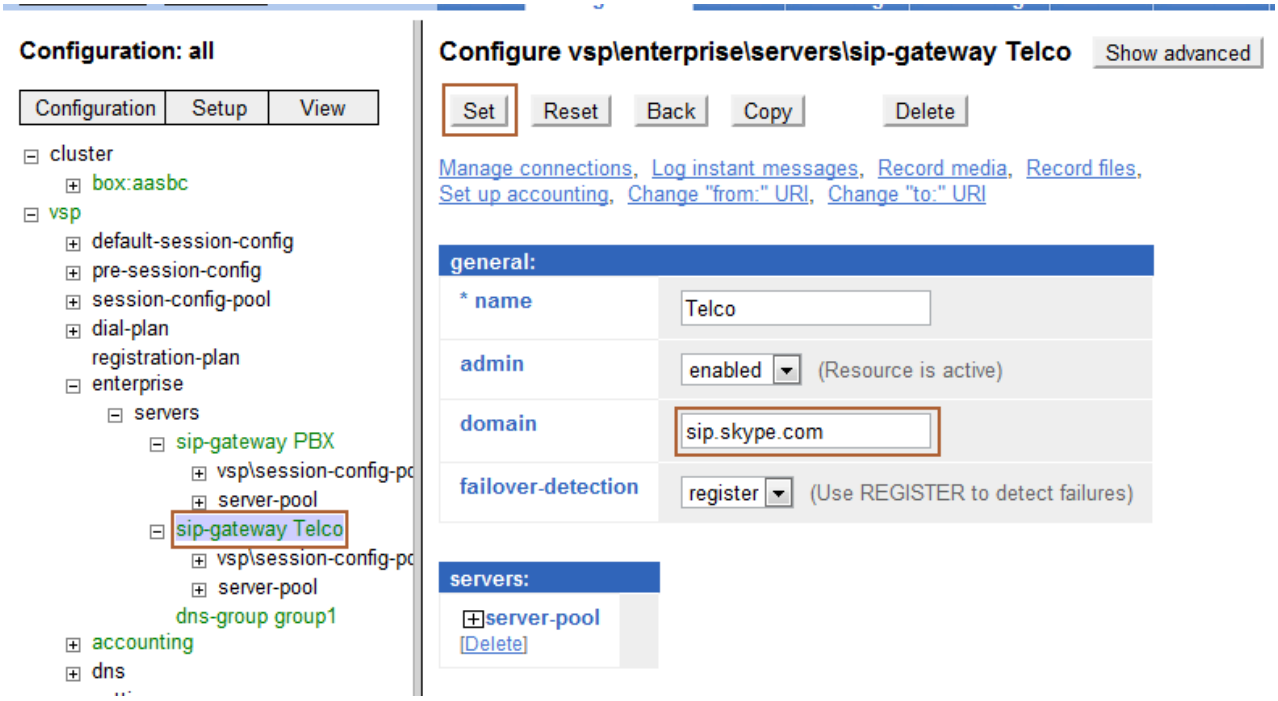
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The **Configure vsp\enterprise\servers\sip-gateway PBX** page is displayed. Type **avaya.com** in the domain field then click **Set**.



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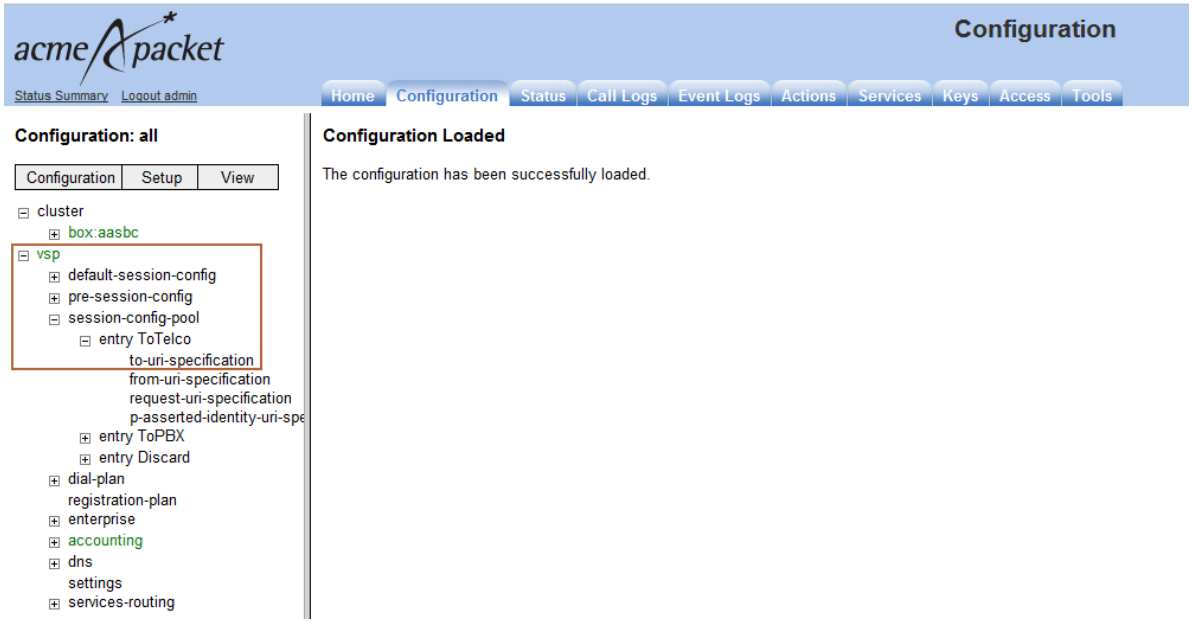
Select **sip-gateway Telco** on the left pane. The **Configure vsp\enterprise\servers\sip-gateway Telco** page is displayed. Type **sip.skype.com** in the **domain** field then click **Set**.



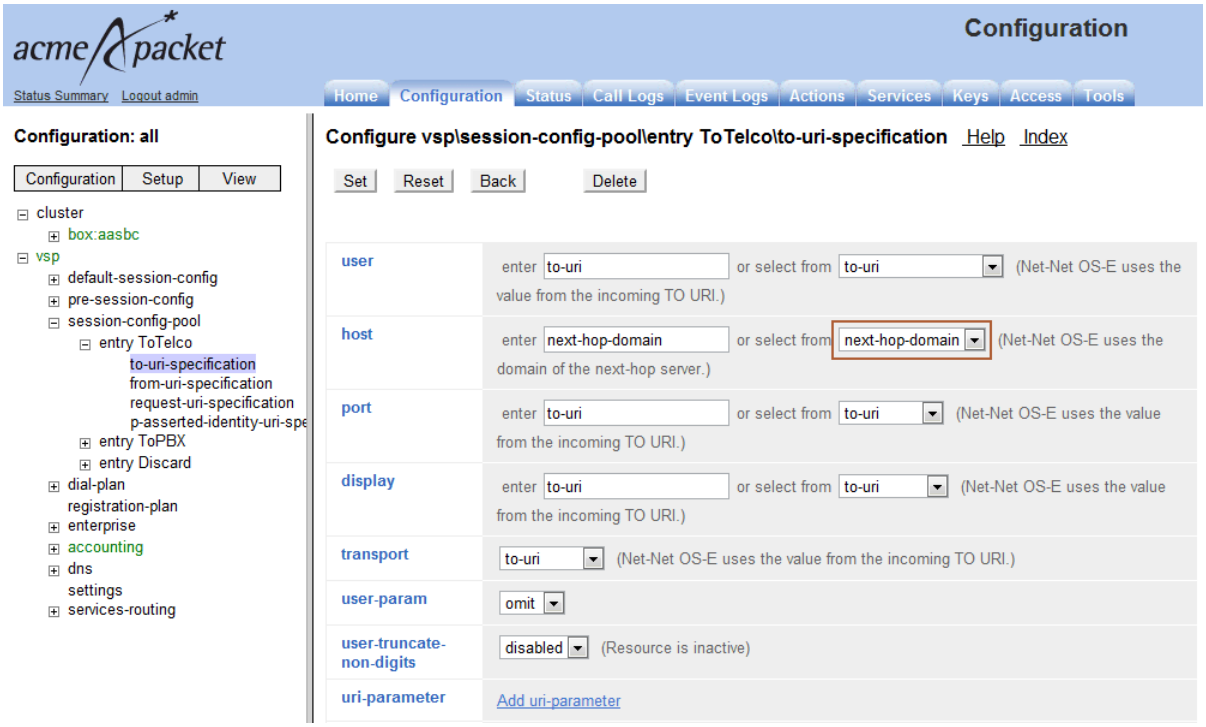
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6.3. Administer Outbound SIP Header Manipulation Rules

The following steps translate the SIP domain names created in **Section 4.2** 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.

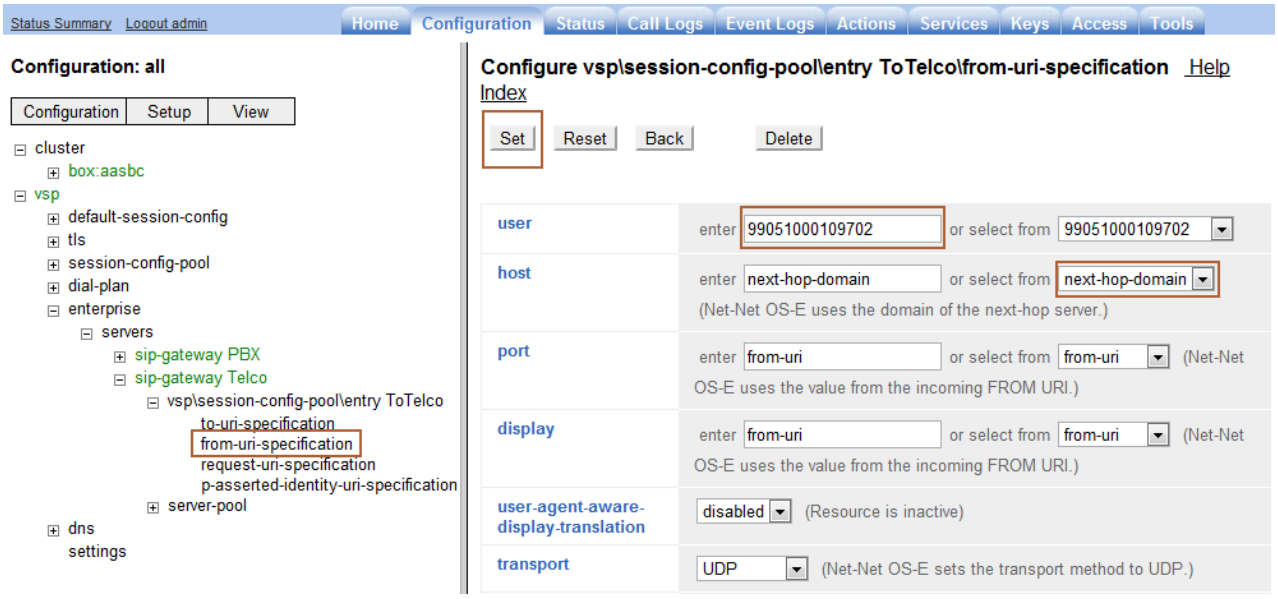


Under **host** select **next-hop-domain** from the drop-down list box then click **Set**.

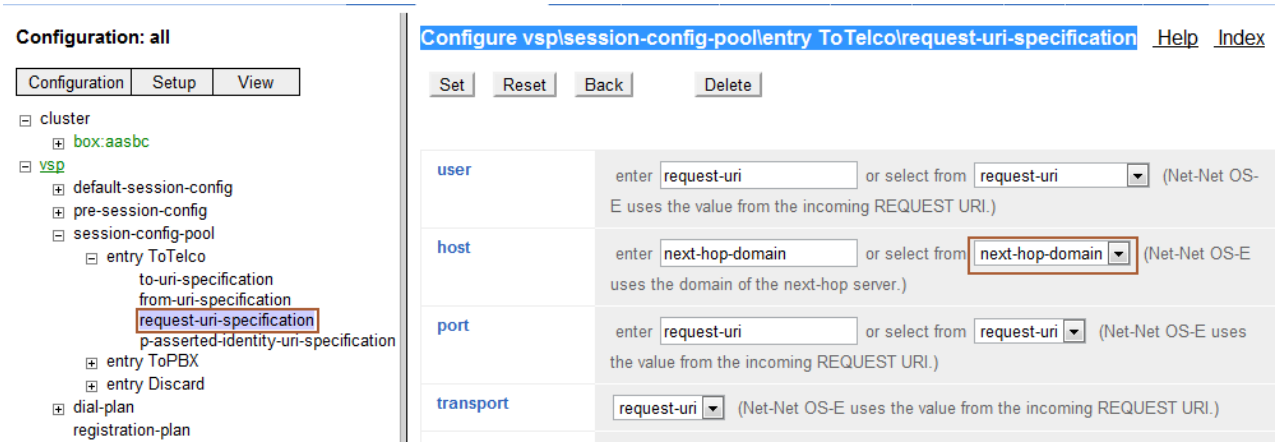


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Select **from-uri-specification** from the left pane. The **Configure vsp\session-config-pool\entry ToTelco\from-uri-specification** page is displayed. Type the Skype Manager assigned **Skype ID** in the **user** field. Under **host** select **next-hop-domain**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

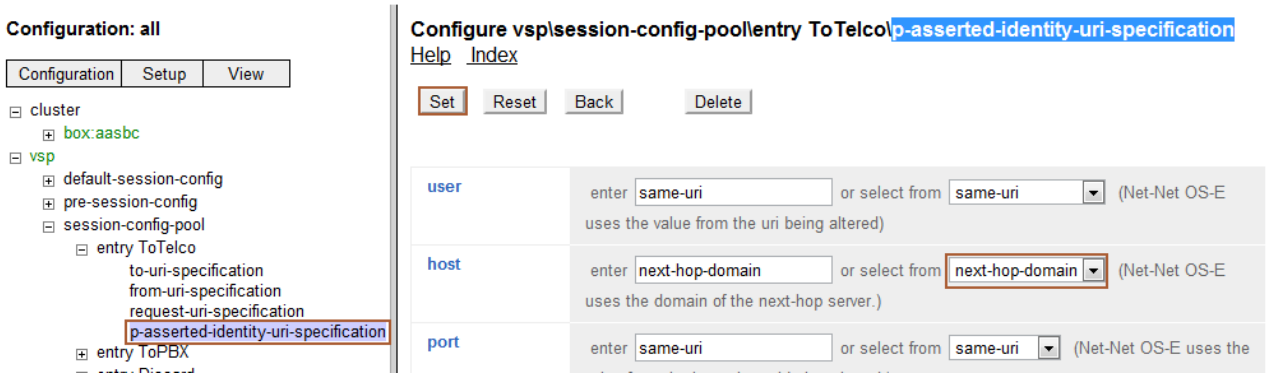


Select **request-uri-specification** from the left pane. The **Configure vsp\session-config-pool\entry ToTelco\request-uri-specification** page is displayed. Under **host** select **next-hop-domain** from the drop-down list box then click **Set**.



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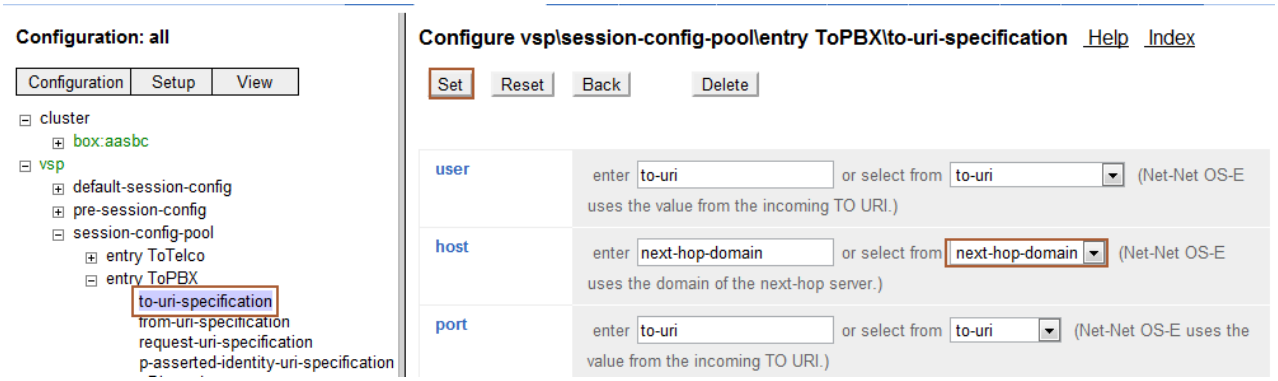
Select **p-asserted-identity-uri-specification** from the left pane. The **Configure vsp\session-config-pool\entry ToTelco\ p-asserted-identity-uri-specification** page is displayed. Under **host** select **next-hop-domain** from the drop-down list box then click **Set**.



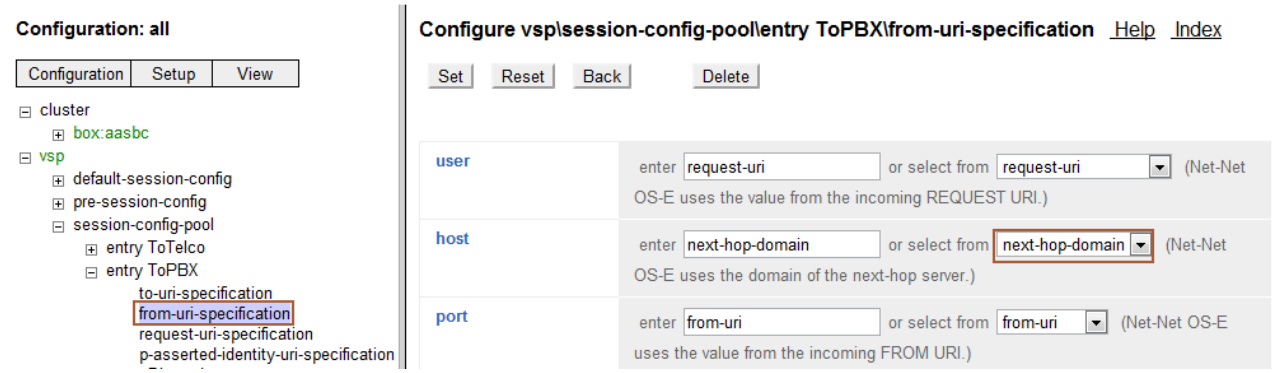
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6.4. Administer Inbound SIP Header Manipulation Rules

The following steps translate the SIP domain names created in **Section 4.2** 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.

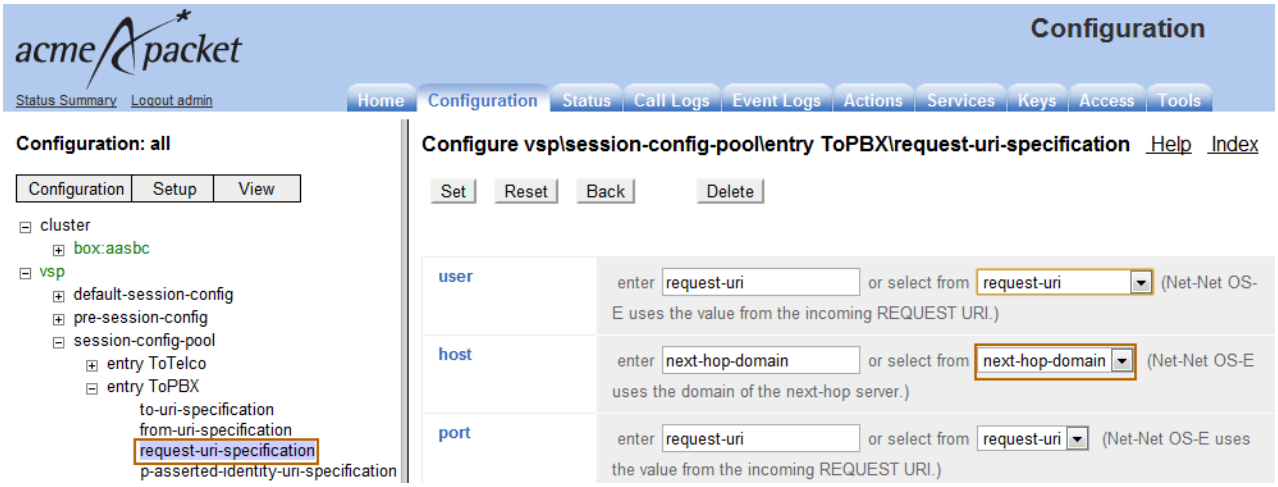


Select **from-uri-specification** from the left pane. The **Configure vsp\session-config-pool\entry ToPBX\from-uri-specification** page is displayed. Under **host** select **next-hop-domain** from the drop-down list box then click **Set**.

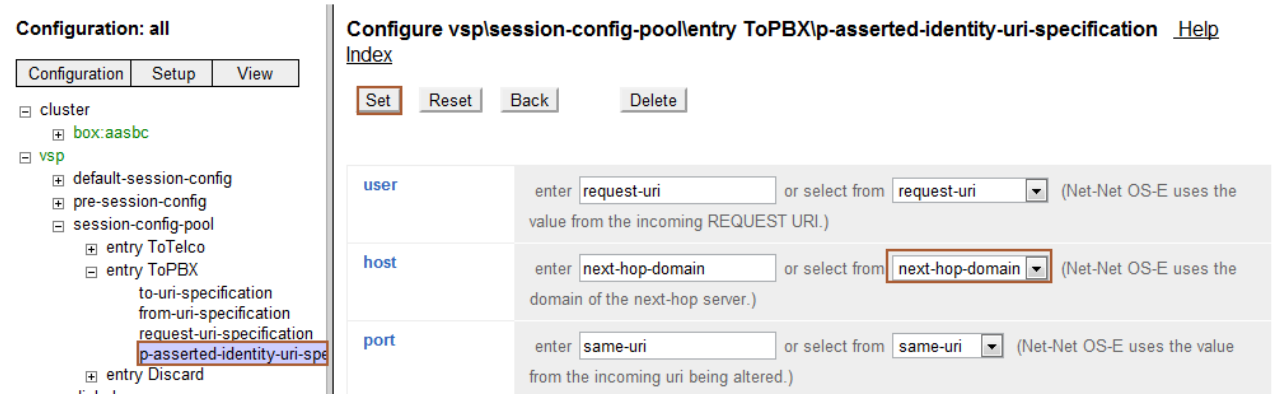


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Select **request-uri-specification** from the left pane. The **Configure vsp\session-config-pool\entry ToPBX\request-uri-specification** page is displayed. Under **host** select **next-hop-domain** from the drop-down list box then click **Set**.



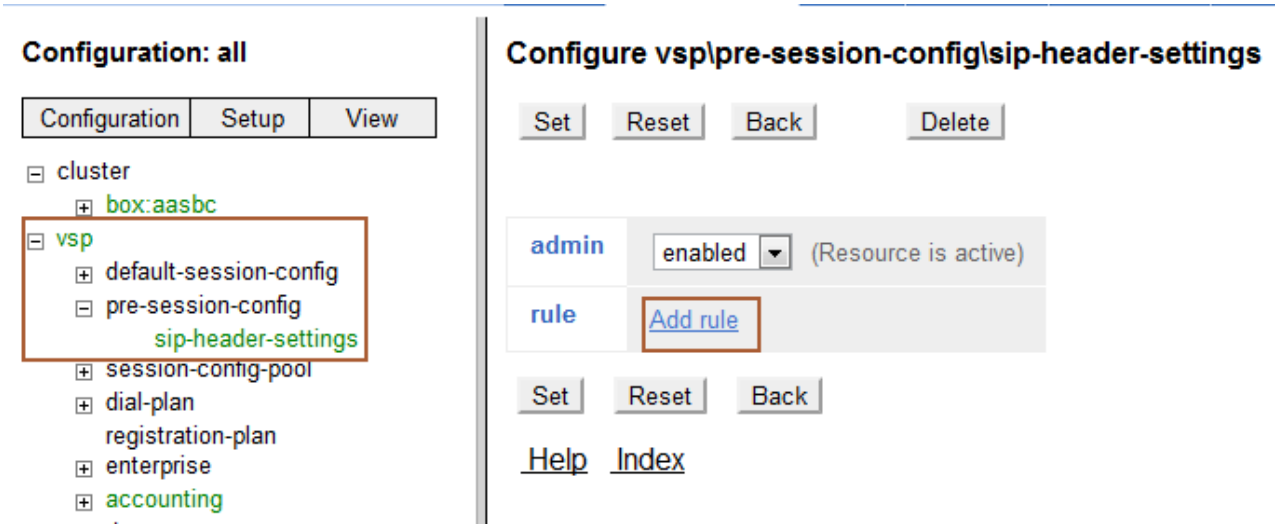
Select **p-asserted-identity-uri-specification** from the left pane. The **Configure vsp\session-config-pool\entry ToPBX\p-asserted-identity-uri-specification** page is displayed. Under **host** select **next-hop-domain** from the drop-down list box then click **Set**.



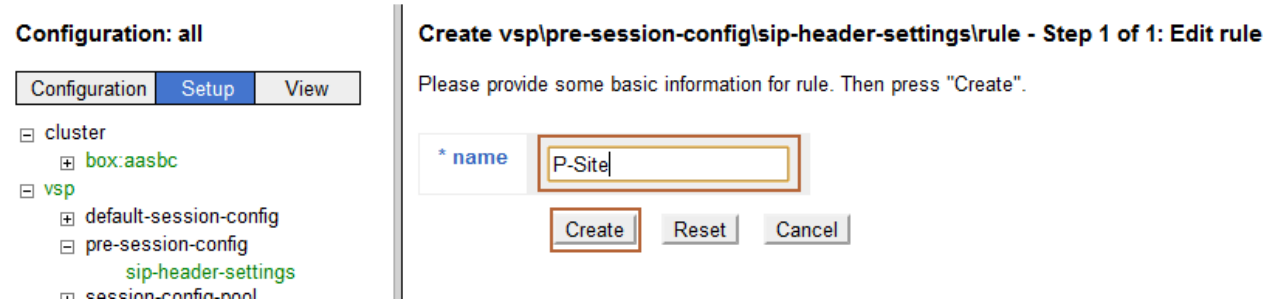
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6.5. Administer SIP Header Rules for Topology Hiding

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**.

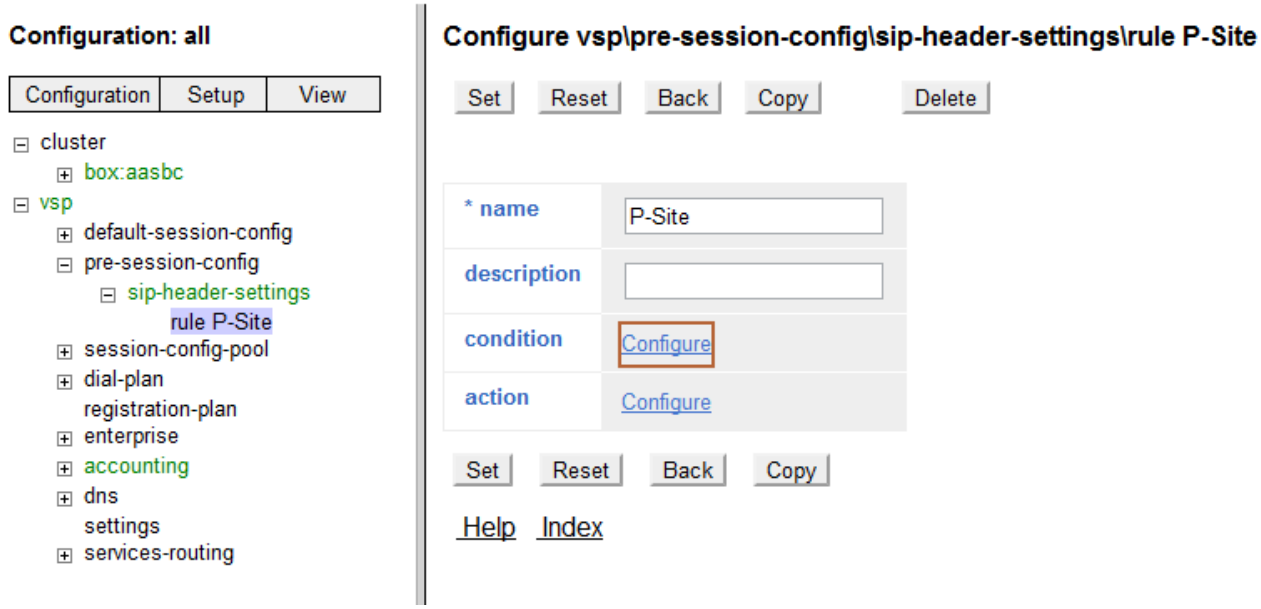


Type a descriptive name for the rule and click **Create**.

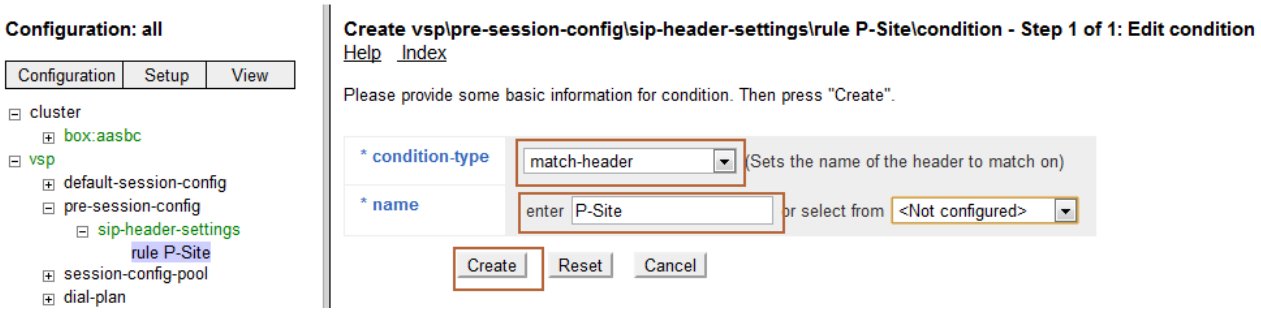


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The **Configure vsp\pre-session-config\sip-header-settings\rule P-Site** page is displayed. Click **Configure**.

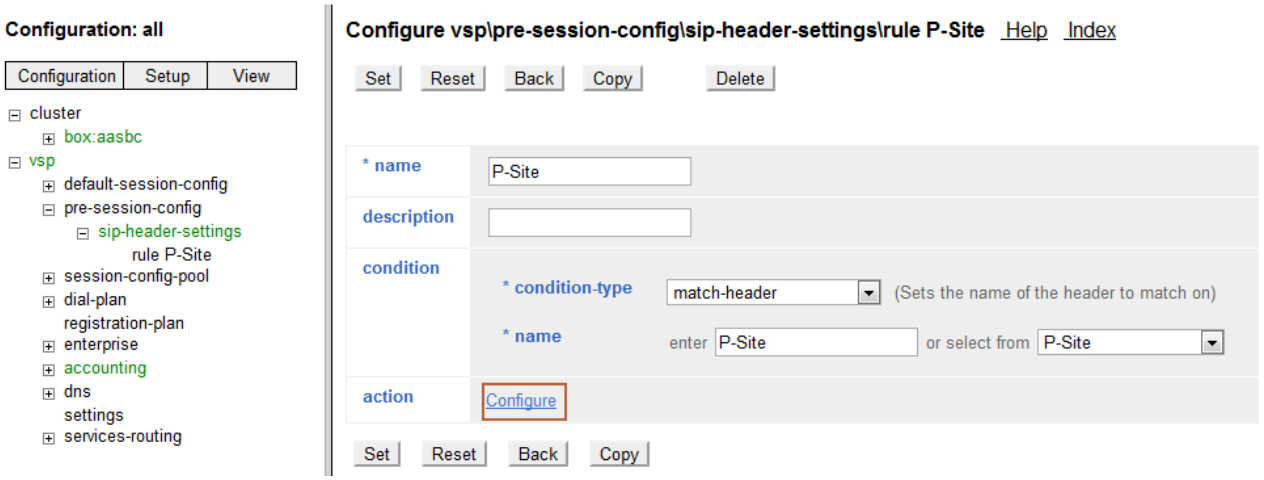


The **Create vsp\pre-session-config\sip-header-settings\rule P-Site\condition** page is displayed. Under **condition-type** select **match-header**. Under **name** type **P-Site** then click Create.

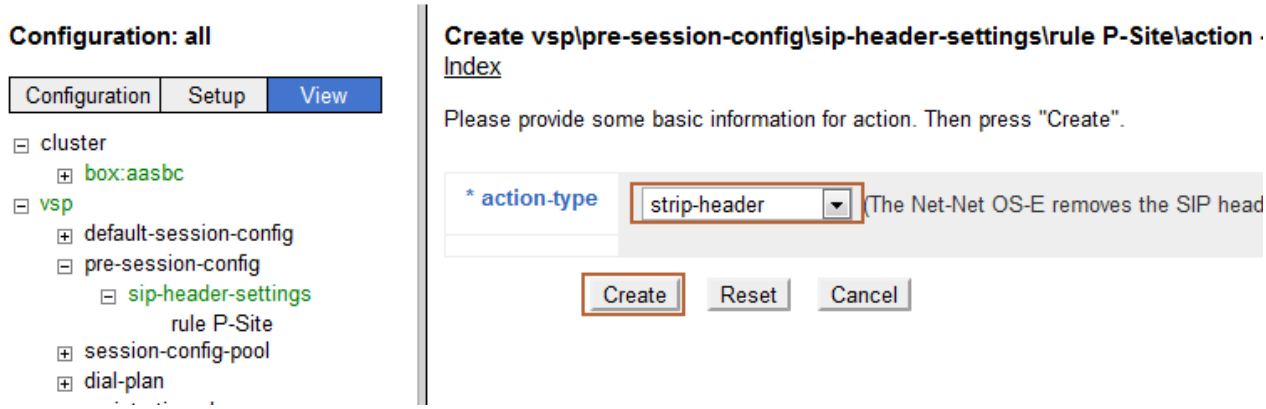


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The **Configure vsp\pre-session-config\sip-header-settings\rule P-Site** page is displayed. Click **Configure**.

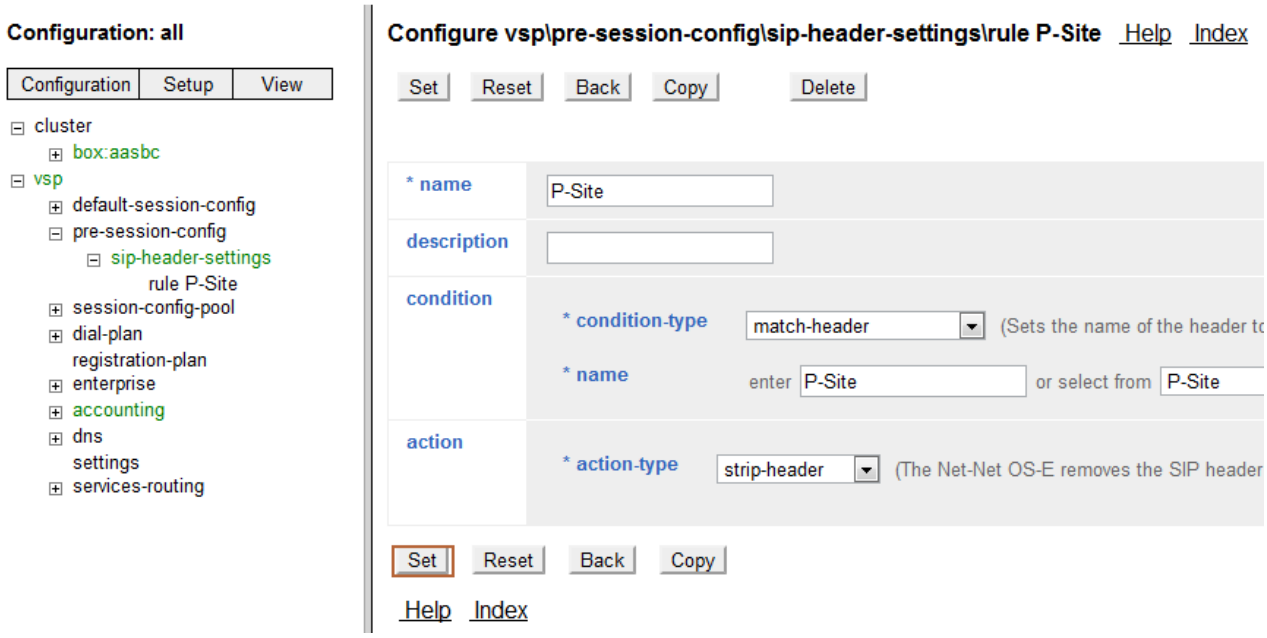


The **Create vsp\pre-session-config\sip-header-settings\rule P-Site\action** page is displayed. Under **action-type** select **strip-header** then click **Create**.



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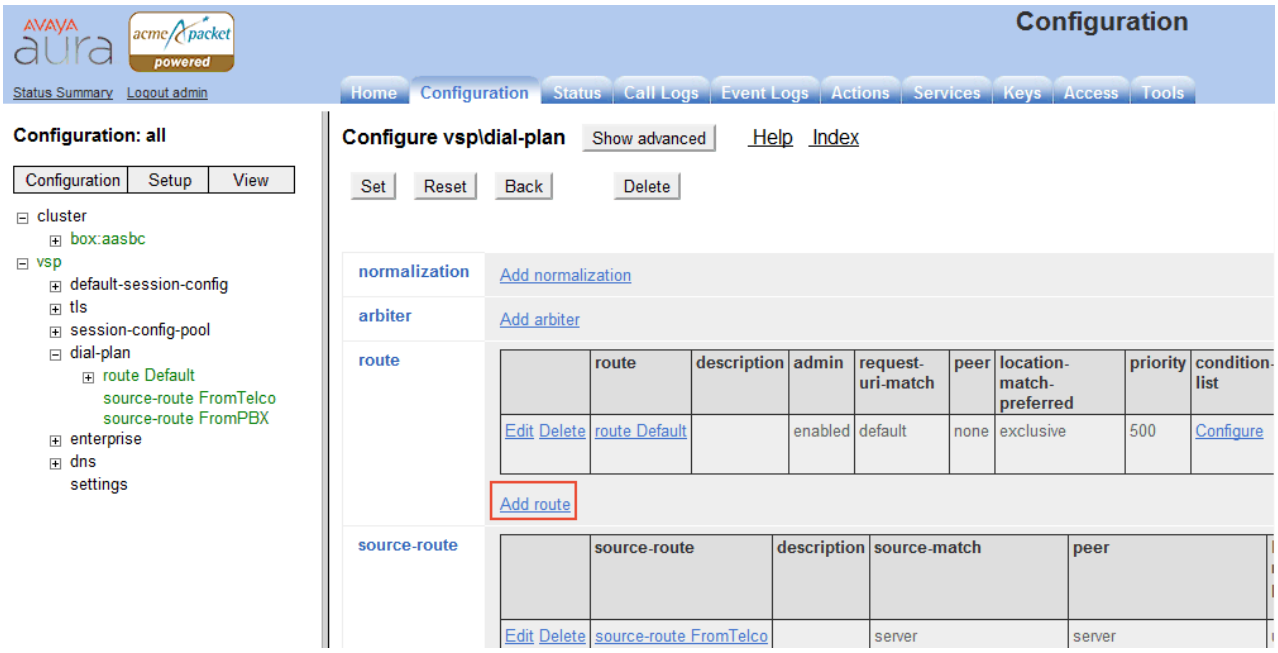
The **Configure vsp\pre-session-config\sip-header-settings\rule P-Site** page is displayed. Click **Set**.



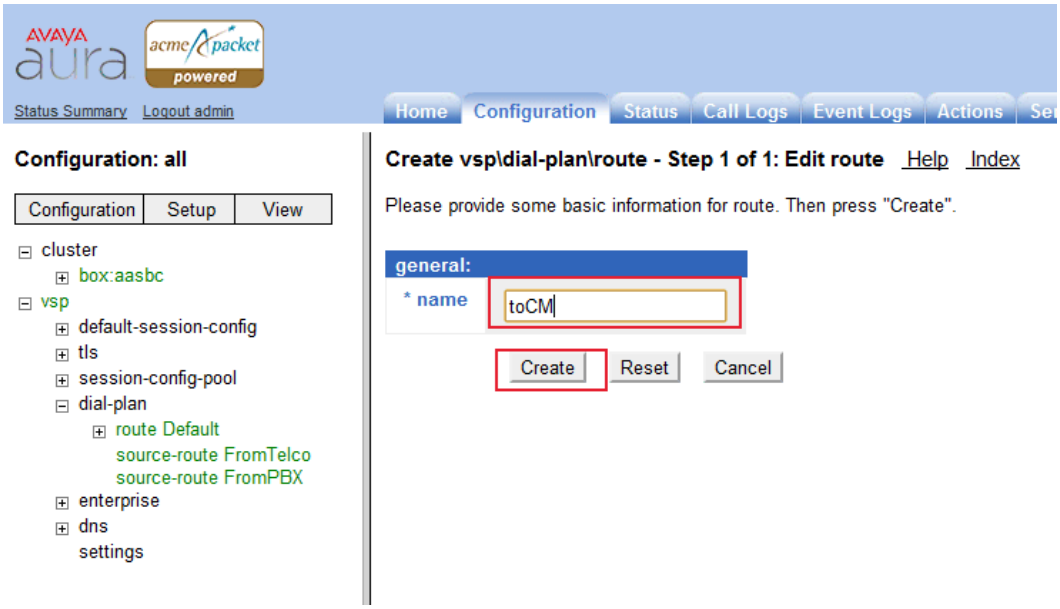
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6.6. Administer Dial Plan

A dial plan entry is required to route the INVITE request that Session Border Controller generates after intercepting the REFER request from Voice Portal as per **Step 9** in **Section 3.1**. Expand **vsp → dial plan → sip-header-settings**. The **Configure vsp\dial-plan** page is displayed. Click **Add route**.



The **Create vsp\dial-plan\route - Step 1 of 1: Edit route** screen is displayed. Type a descriptive name then click **Create**.



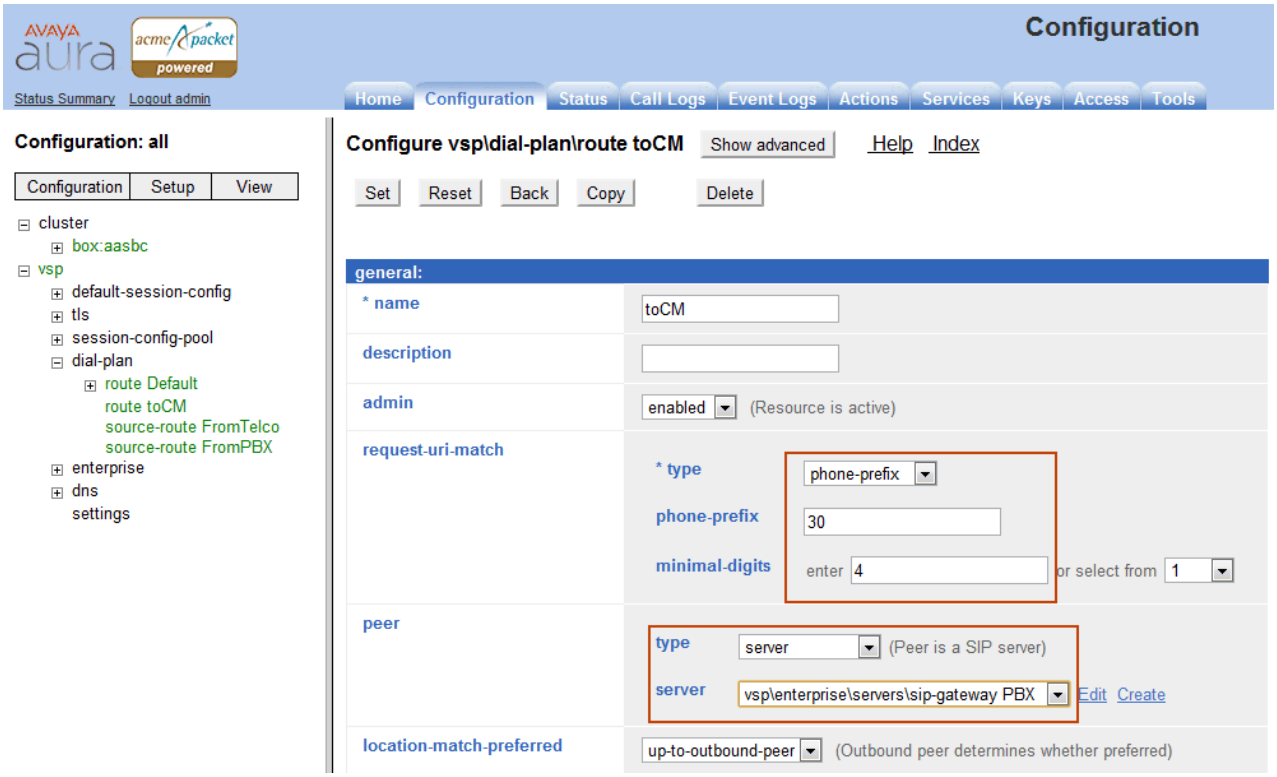
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The **Configure vsp\dial-plan\route toCM** screen is displayed. Under **request-uri-match** select the following options:

- **type:** From the drop-down list box select **phone-prefix.**
- **phone prefix:** Type the internal extension range administered on Communication Manager.
- **minimal-digits**: Type the internal extension length administered on Communication Manager.

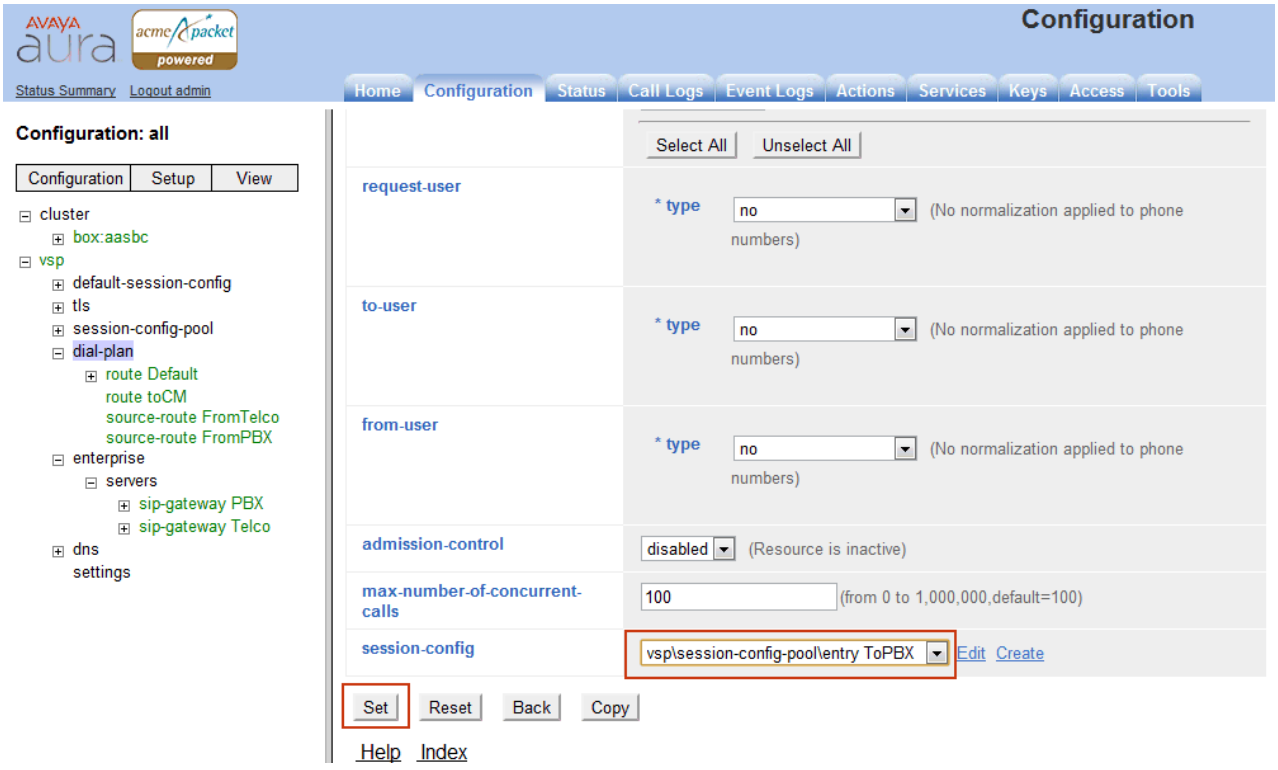
Under **peer** select the following options:

- **type:** From the drop-down list box select **server.**
- **server:** From the drop-down list box select **vsp\enterprise\servers\sip-gateway PBX**



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Scroll down to **session-config** and select **vsp\enterprise\servers\sip-gateway PBX.** Click **Set.**

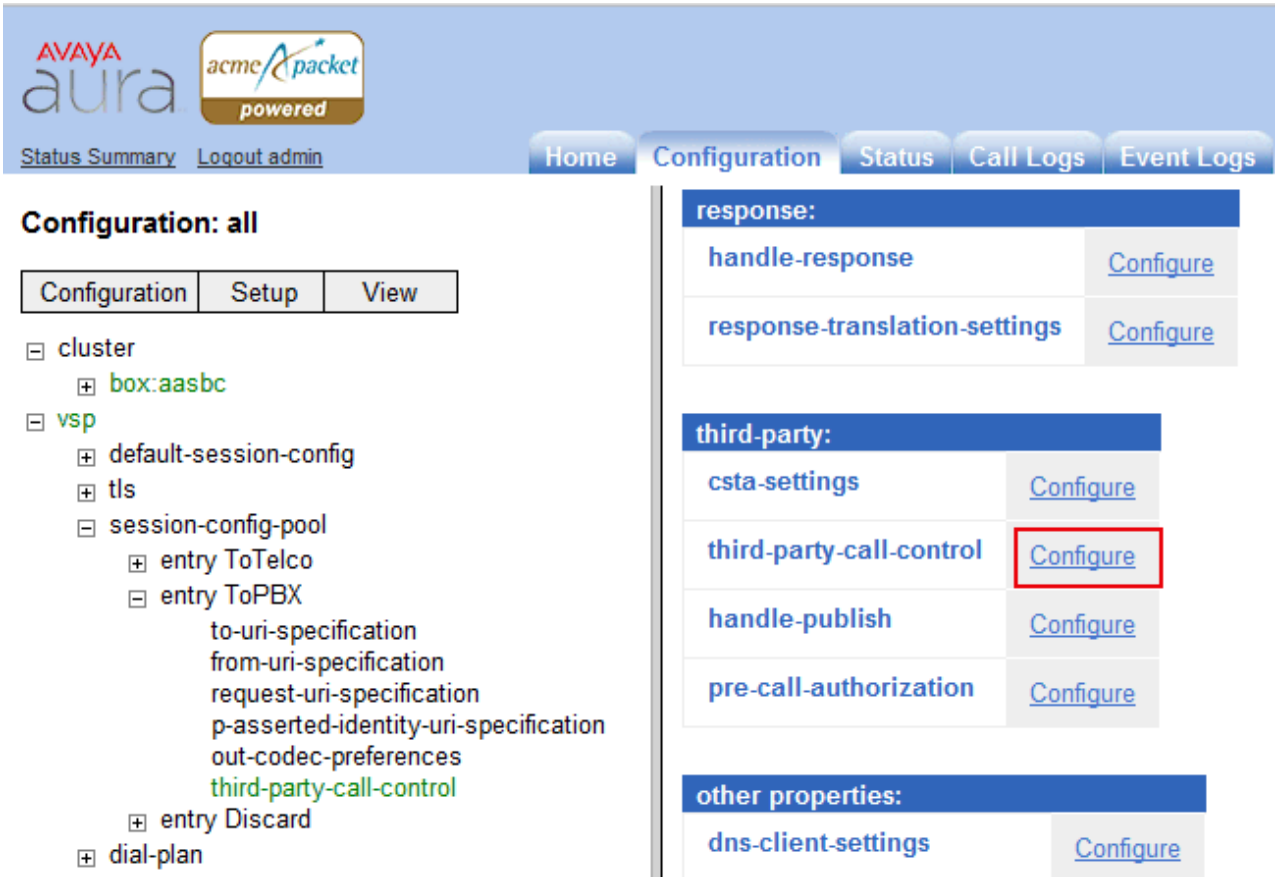


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6.7. Administer Third Party Call Control Settings

The Third Party Call Control settings determine whether a REFER request is handled locally by the Session Border Controller (INVITE with REPLACES is sent to Session Manager) or if it is proxied to Skype Connect. Until Skype Connect introduces support for the REFER method, third party call control needs to be enabled on the Session Border Controller.

Select **vsp\session-config-pool\entry ToPBX.** Scroll down to **third-party**. Click **Configure** next to **third-party-call-control**.

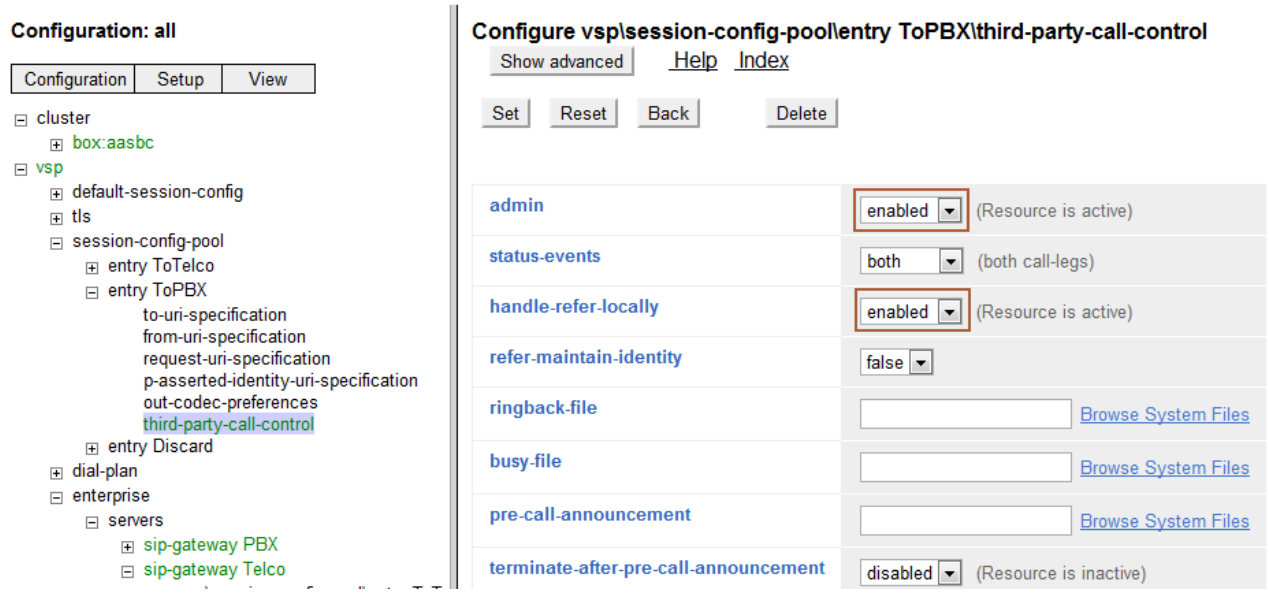


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The **Configure vsp\session-config-pool\entry ToTelco\third-party-call-control** screen is displayed.
Select the following options:

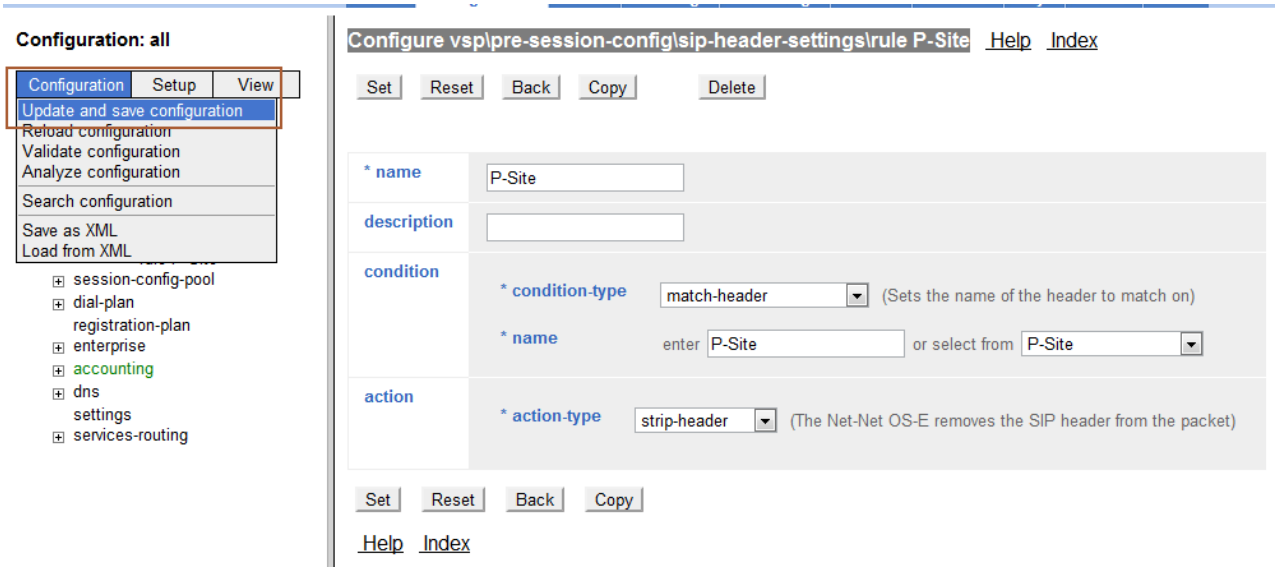
- **admin:** From the drop-down list box select **enabled.**
- **handle-refer-locally:** From the drop-down list box select **enabled.**

Click **Set.**



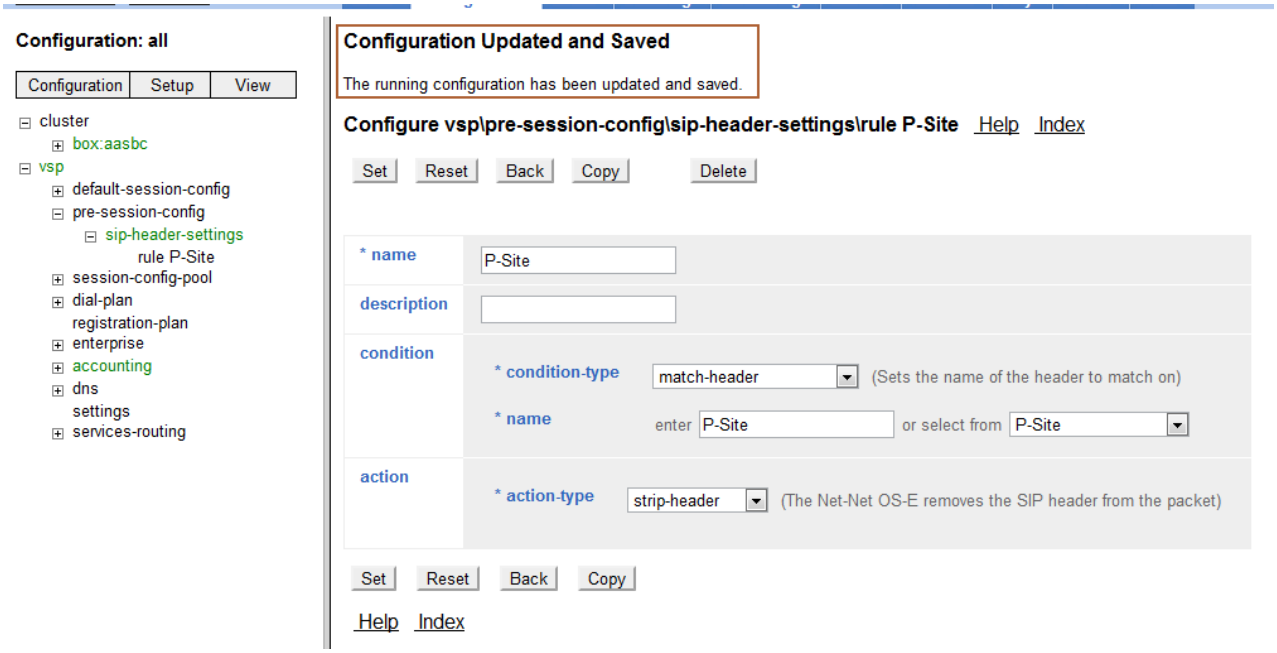
6.8. Save the Configuration

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



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



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7. Configure Avaya Voice Portal

This section describes the relevant configuration of the SIP Trunk between Voice Portal and Session Manager, and any administration necessary to configure Voice Portal to support calls from Skype Connect. In addition to the steps described in this section, other installation and initial configuration actions will be needed. For more information on these additional actions, see the appropriate documentation in **Section 11.**

Unless otherwise indicated, the administration steps described in this section were completed using the web administration page of the Voice Portal Management System (VPMS).

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7.1. Add SIP Connection to Avaya Aura® Session Manager

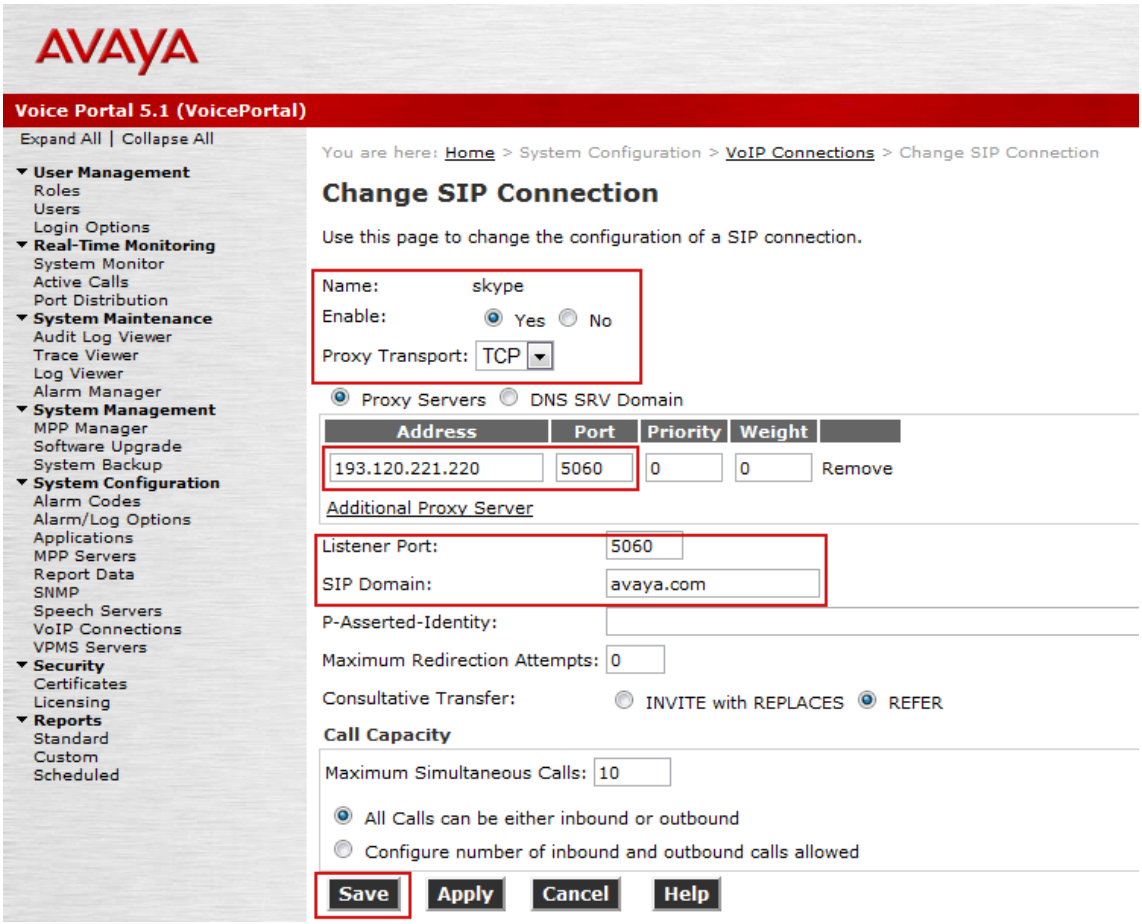
Navigate to the **System Configuration → VoIP Connections** task. Select **SIP** tab and click **Add** (not shown). Enter the following values and use defaults for remaining fields.

- **Name** Enter a descriptive name for Session Manager.
- **Enable** Select **Yes**
- **Proxy Transport** Select **TCP**

In the **Proxy Servers** table, enter the following values.

- **Address** Enter IP address of Session Manager (ASSET card).
- **Port** Enter **5060**
- **Listen Port** Enter **5060**
- **SIP Domain** Enter the same SIP Domain name as defined in **Section 4.2**

Click **Save** to save configuration.



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7.2. Configure Self Service Application

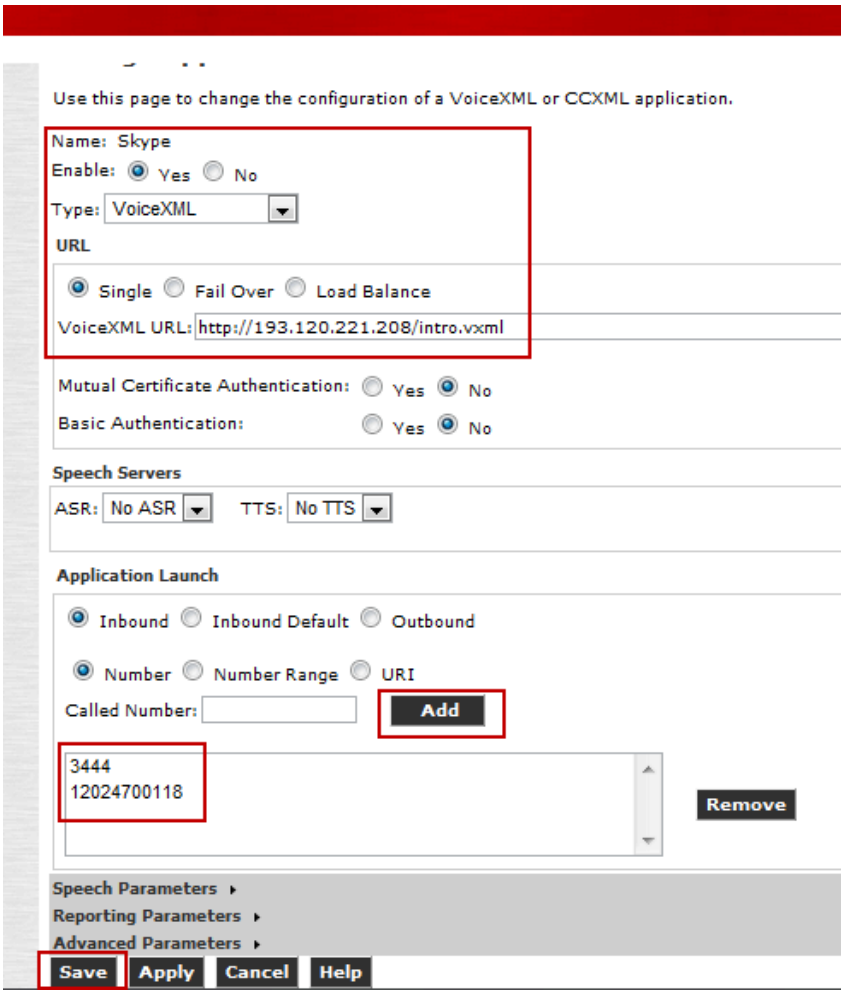
Navigate to the **System Configuration → Applications** task and click **New** (not shown). Enter the following values and use defaults for remaining fields.

- **Name** Enter a descriptive name for Session Manager
- **Enable** Select **Yes**
- **Type** Select **VoiceXML** from drop-down menu
- **VoiceXML URL** Enter the URL of self-service application hosted on an HTTP server

In the **Application Launch** section, enter the following values.

- **Type** Select **Inbound** and select **Number**
- **Called Number** Enter user part (phone number) of incoming SIP Request-URI header.

Click **Add** to save the new **Number**. Click **Save** to save the changes.

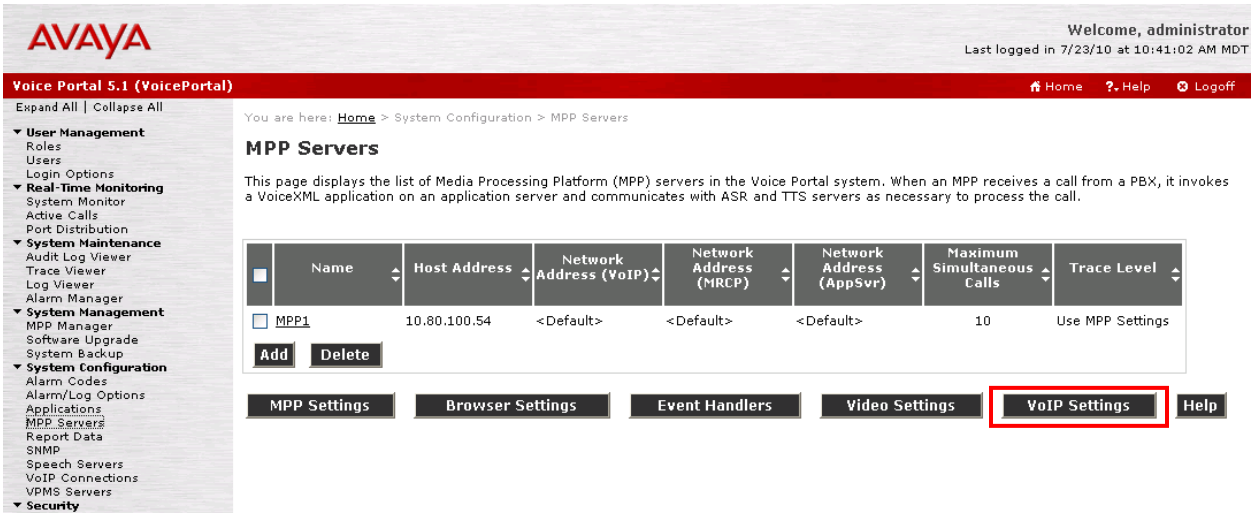


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7.3. Configuring Support for the G.729 Codec

By default Voice Portal supports the G.711Mu-law codec. Voice Portal 5.1 adds support for the G.729 codec as well. Please note that the following steps will only influence the codec selection/priority for outbound calls (calls generated by Voice Portal). The configuration steps will not reorder the codec priority of incoming SDP messages. To enable G.729 and to ensure it has a higher priority than G.711Mu-law take the following steps.

Navigate to **System Configuration →MPP Servers**. Select the **VoIP Settings** button as indicated below.

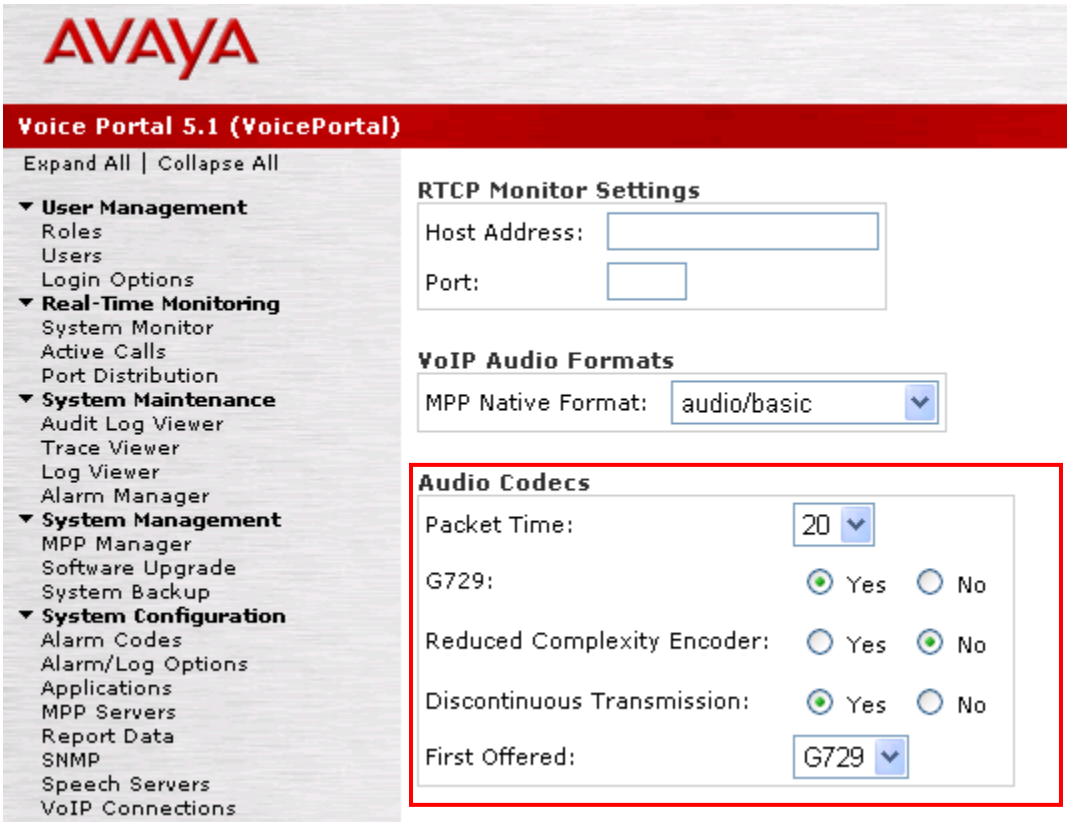


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In the screen that appears next, scroll down to the section titled **Audio Codecs** and configure the options as shown below.

- **Packet Time** Sampling rate for encoding audio. This value should match across systems
- **G729** Select the **Yes** radio button to enable G.729
- **Reduced Complexity Encoder**
  (*optional*) Enables support for the G.729A codec which further reduces bandwidth usage with little cost to voice quality.
- **Discontinuous Transmission**
  (*optional*) Enables support for the G.729B codec. The G.729B/Annex-B specification further reduces network bandwidth as it sends only the audio packets that contain speech data (packets that contain silence are not transmitted)
- **First Offered** Select G729 from the drop-down if using the G729 codec is preferred for calls rather than using G.711

Click **Save** to save the changes.



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8. General Test Approach and Test Results

The test environment consisted of:

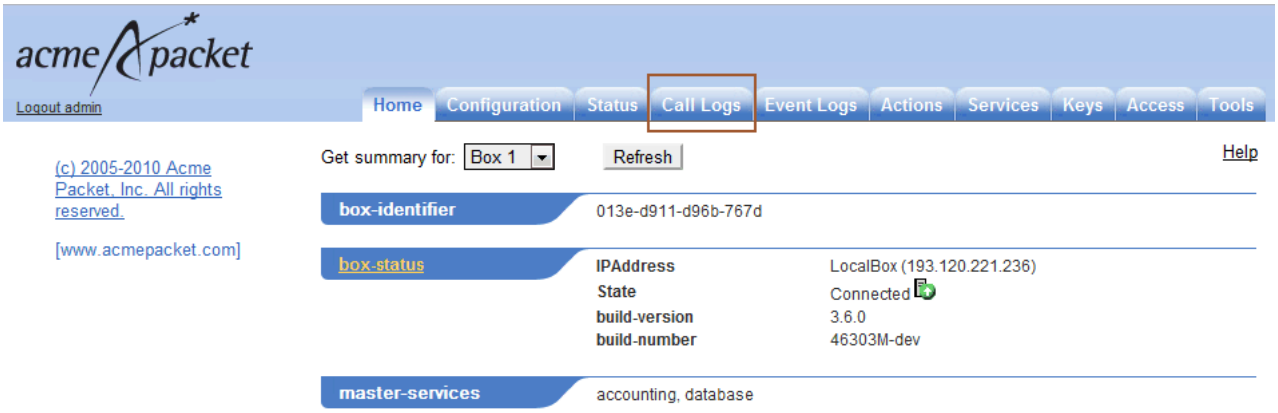
- A simulated enterprise with Voice Portal 5.1, System Manager R6, Session Manager R6, Communication Manager – Evolution Server R6, Session Border Controller R6 and Avaya phones.
- A production version of the Skype Connect 2.0 service, to which the simulated enterprise was connected.

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

- Voice Portal access from the PSTN and the Skype P2P Network
- Basic call scenarios using G.711 and G.729 codecs
- DTMF transmission
- Blind Transfer from Voice Portal to Communication Manager
- Consultative transfer from Voice Portal to Communication Manager using the following two SIP methods:
  - INVITE with REPLACES
  - REFER
- Bridged Transfer from Voice Portal to Communication Manager

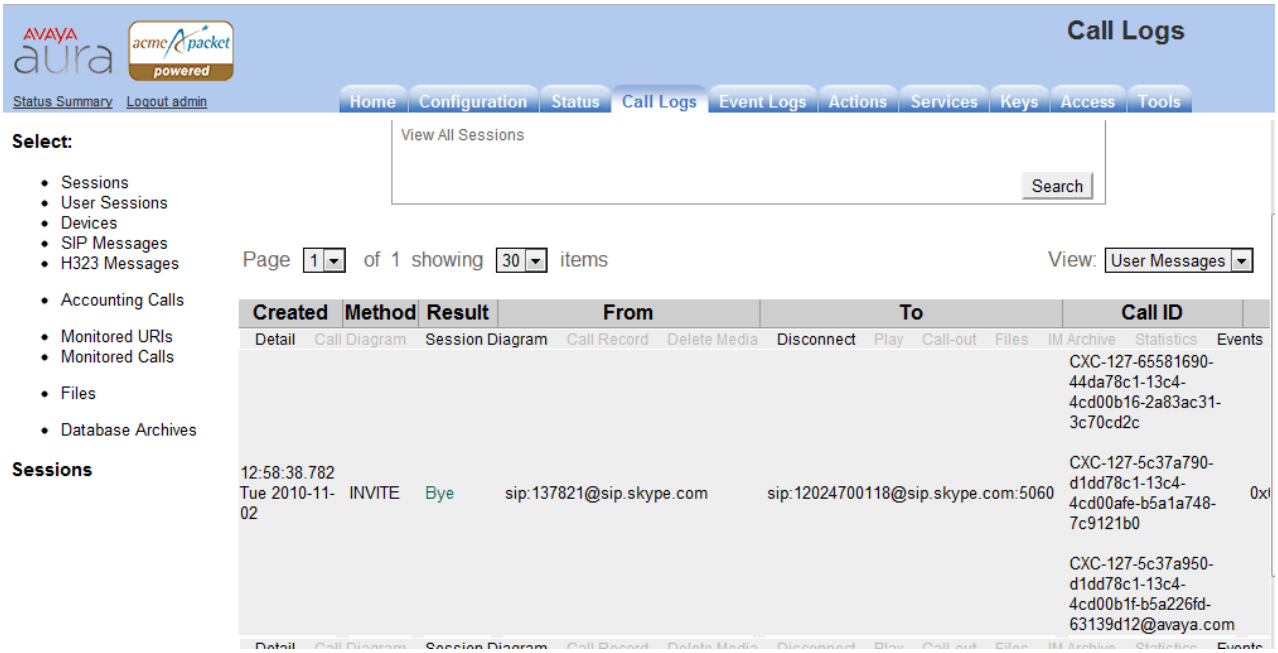
9. Verification Steps

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



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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 Voice Portal and Session Manager built-in SIP tracing tools or Wireshark can be used to capture SIP traces. SIP traces can be instrumental in understanding SIP protocol issues resulting from configuration problems.

10. Conclusion

As illustrated in these Application Notes Avaya Voice Portal R5.1, Avaya Aura® Session Manager R6, Avaya Aura® Session Border Controller R6 can be configured to interoperate successfully with Skype Connect™ 2.0. This solution provides PSTN and Skype Client users the ability to access the Avaya Voice Portal self-service platform hosted on a secure enterprise network through click-to-call applications over a public SIP trunk using the Skype Connect™ service.

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.

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11. References

This section references the product documentation relevant to these Application Notes.

Avaya Aura® Session Manager

1) *Avaya Aura® Session Manager Overview*, Doc ID 03-603323. Available at http://support.avaya.com.
2) *Installing and Upgrading Avaya Aura® Session Manager 6.0*, Doc ID 03-603324. Available at http://support.avaya.com.
3) *Installing and Upgrading Avaya Aura® System Manager 6.0*. Available at http://support.avaya.com.
4) *Maintaining and Troubleshooting Avaya Aura® Session Manager 6.0*, Doc ID 03-603325, available at http://support.avaya.com.
5) *Avaya Aura® Session Manager Case Studies*, dated January 2, 2010. Available at http://support.avaya.com

Avaya Voice Portal

6) *Administering Voice Portal*, available at http://support.avaya.com
7) *Configuring Avaya Voice Portal with Avaya Communication Manager and Designing a Sample Speech Application using Dialog Designer* – Issue 1.0, available at http://support.avaya.com

Avaya Aura® Communication Manager

8) *Hardware Description and Reference for Avaya Aura® Communication Manager* (COMCODE 555-245-207) http://support.avaya.com/elmodocs2/comm_mgr/r4_0/avayadoc/03_300151_6/245207_6/245207_6.pdf
9) *SIP Support in Avaya Aura® Communication Manager Running on Avaya S8xxx Servers,* Doc ID 555-245-206, May 2009. Available at http://support.avaya.com.
10) *Administering Avaya Aura® Communication Manager,* Doc ID 03-300509, June 2010. Available at http://support.avaya.com.

Skype Connect

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

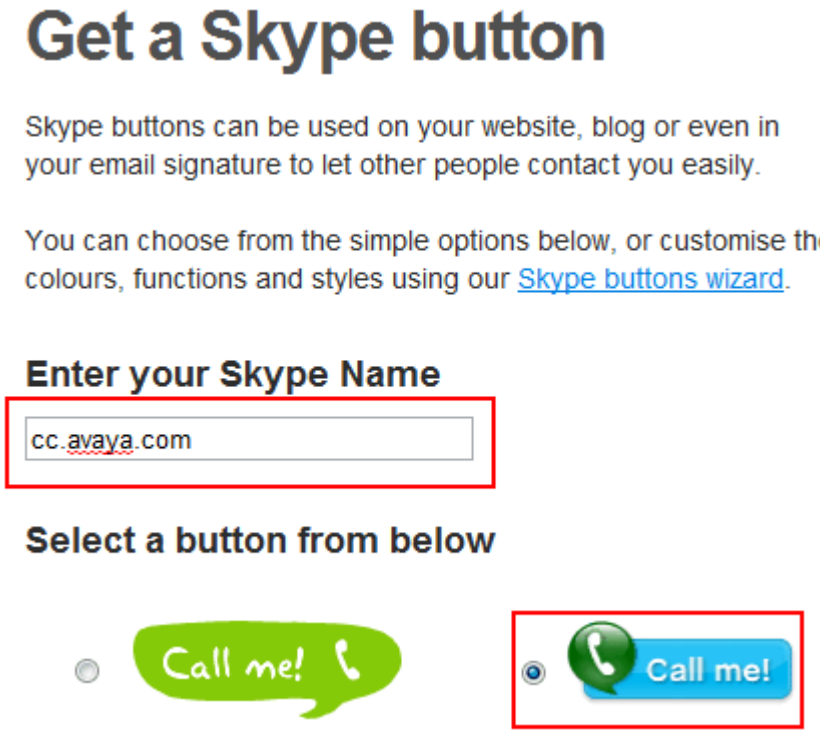
11) *Skype for SIP product datasheet, Version 2.0,* 2010.

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Appendix A

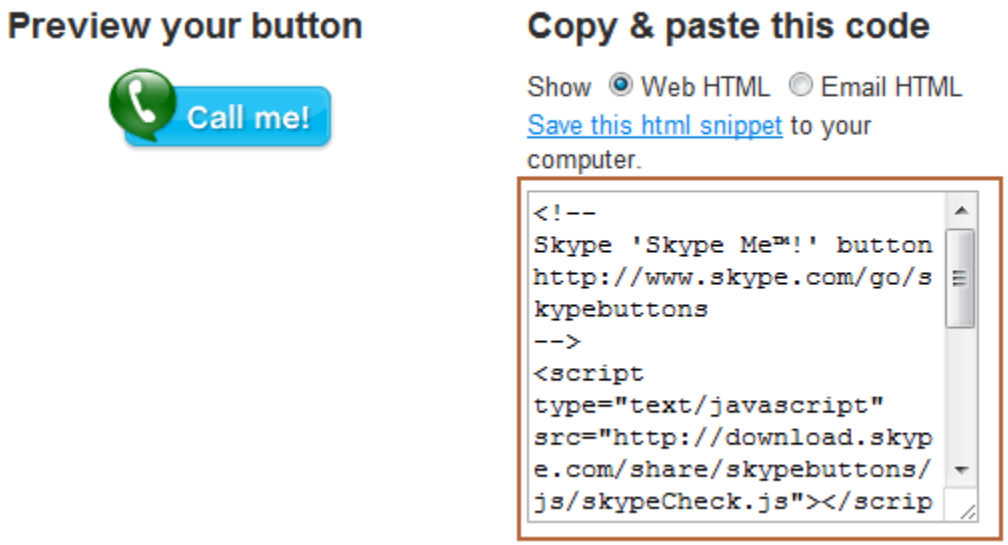
In **Section 5.7** the provisioning of a Business User is discussed. The Business User is associated with a unique Skype Name which can be dialed from the Skype Client application or can be used in click-to-call applications to access the Avaya Voice Portal self-service platform from an embedded Skype Button on a webpage or the from the Skype Client application. This section describes the steps for embedding a Skype Button in HTML code to allow Internet users access to Avaya Voice Portal.

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 5.7** and select a Skype Button.



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Scroll down to the bottom of the page. Copying and pasting the HTML snippet from the text box into an existing webpage will allow users to access Avaya Voice Portal.

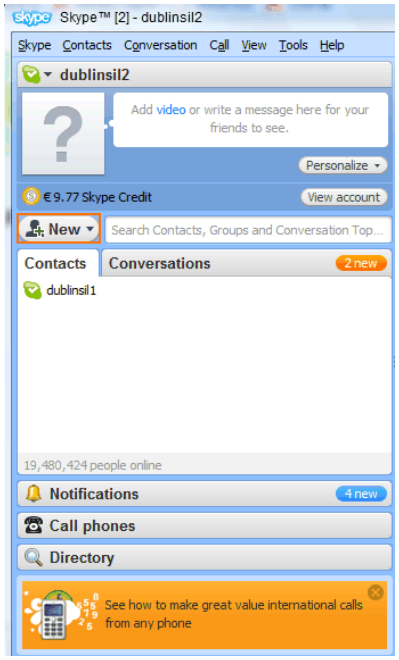


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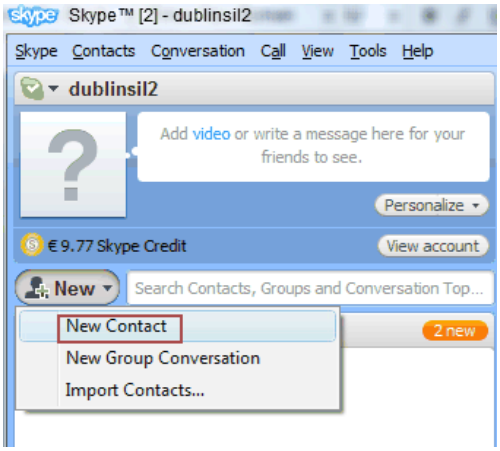
Appendix B

The steps for adding the Business User from **Section 5.7** as a new contact in the Skype Client application are discussed below.

Launch the Skype Client. Click on **New**.

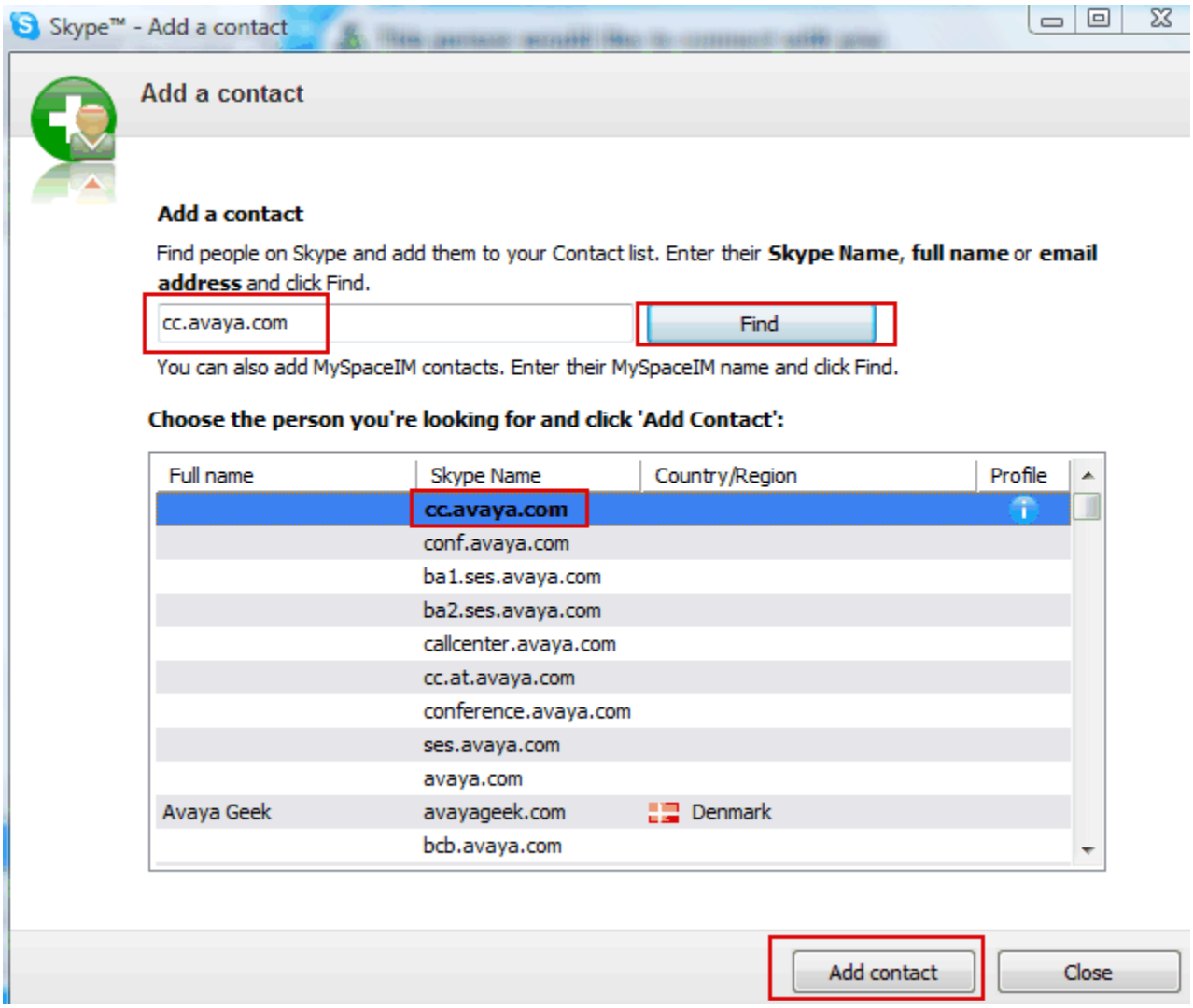


Select **New Contact** from the drop-down menu.



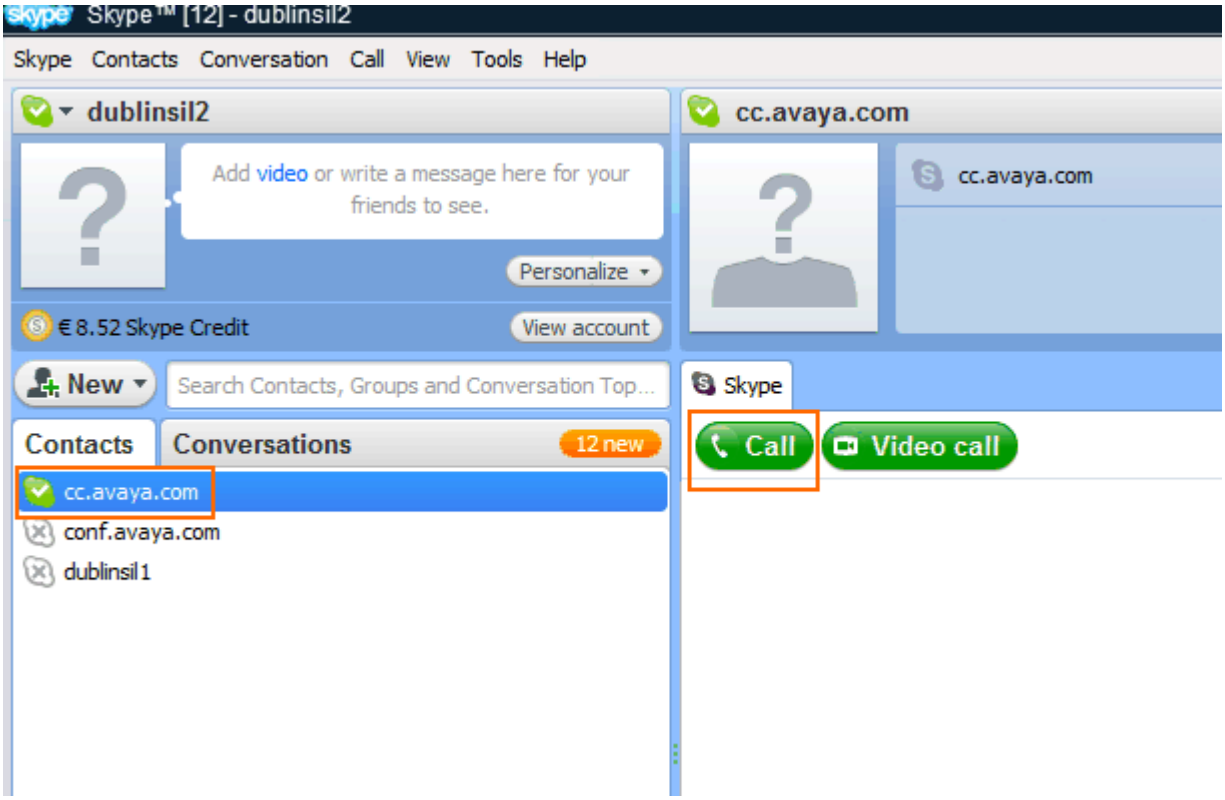
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The **Add a contact** screen is displayed. Type the Skype Name from **Section 5.7** and click **Find** then click **Add contact**.



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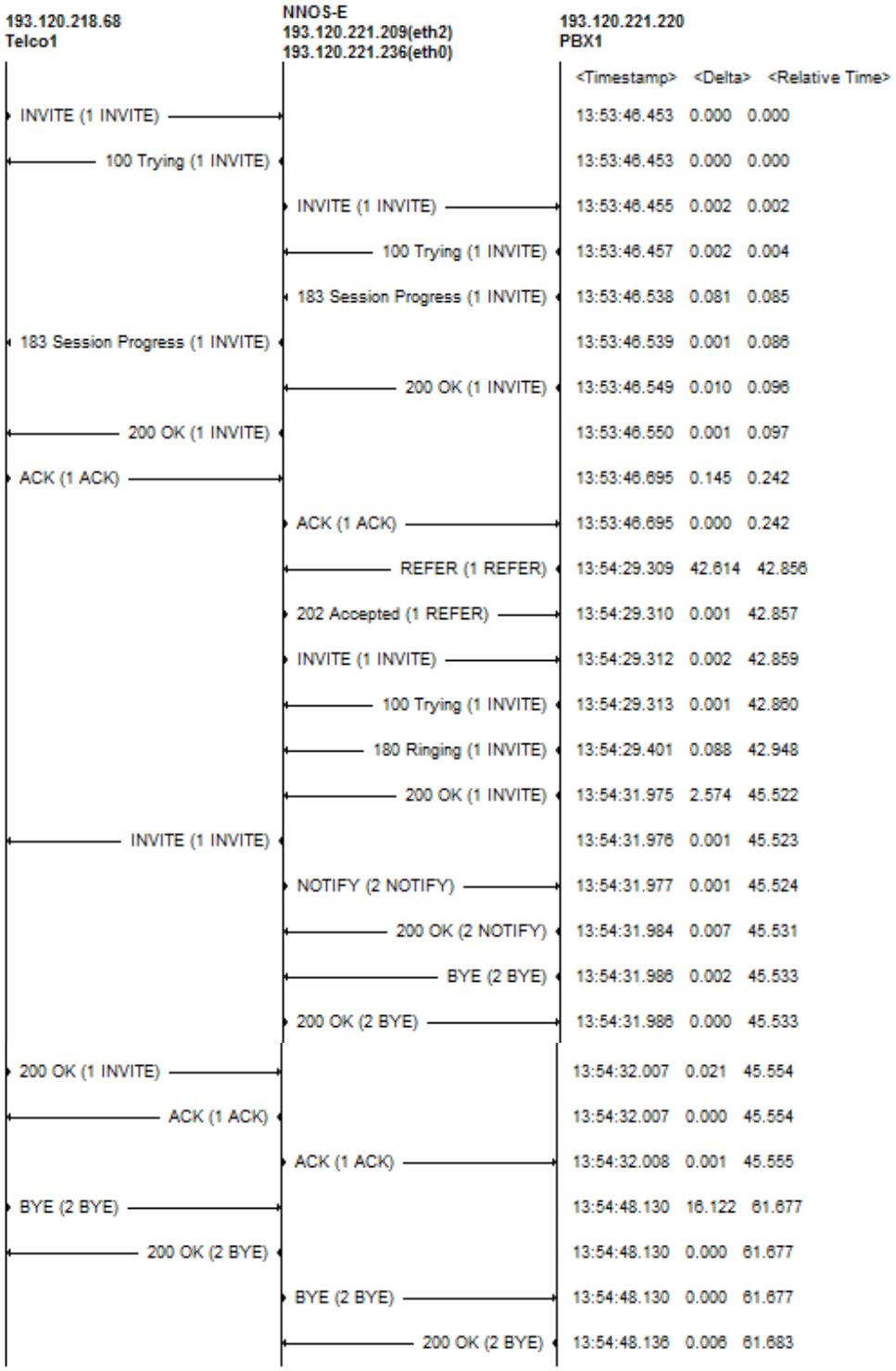
To access the Avaya Voice Portal self-service platform, under the **Contacts** tab click on the Skype Name then on the **Call** button in the right pane.



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Appendix C

Following is a ladder diagram taken from the Call Log of the Session Border Controller, displaying a call where Voice Portal uses blind transfer to connect the PSTN caller to an internal extension as per **Section 3.1**. Please note that SIP messages between Communication Manager and Voice Portal are not included.



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